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SCHEDULE OF COMMITTEEE MEETINGS AND OTHER EVENTS

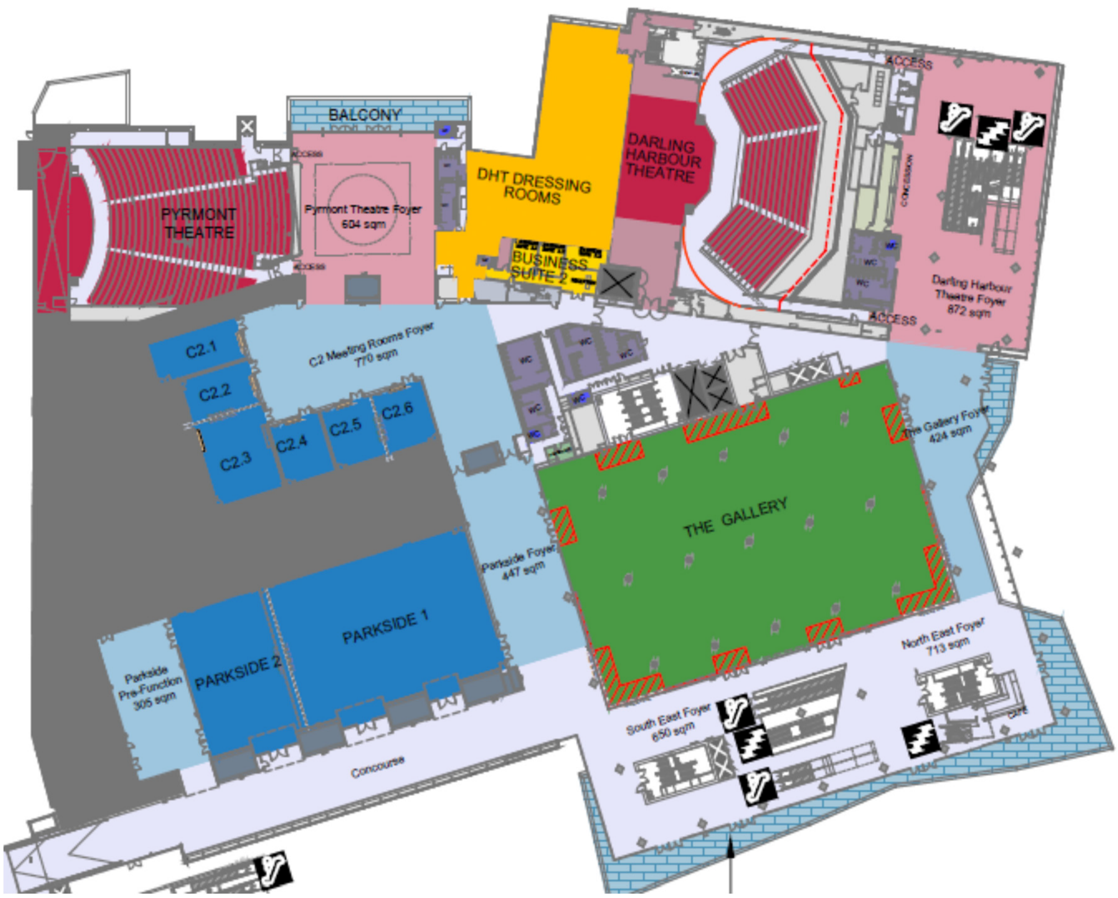
Please refer to the Meeting Mobile App for the up-to-date schedule of meeting services and events

Sydney International Convention Center

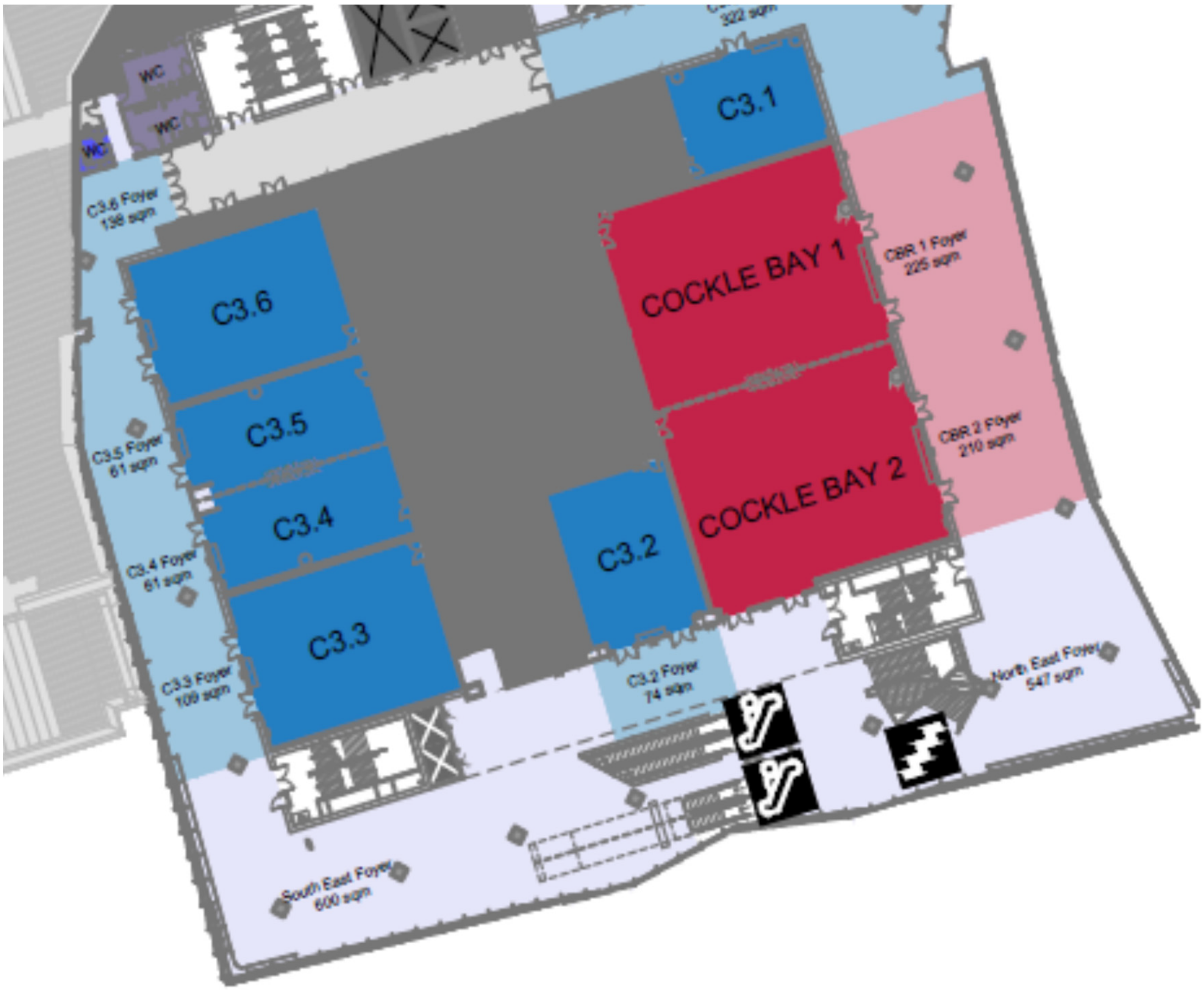
All Technical Sessions will be held on Levels 2 and 3

GROUND FLOOR LEVEL: Registration and Speaker Preparation Room

LEVEL 2: Prymont Theatre, The Gallery, C2 Meeting Rooms, Concourse to Exhibition Centre

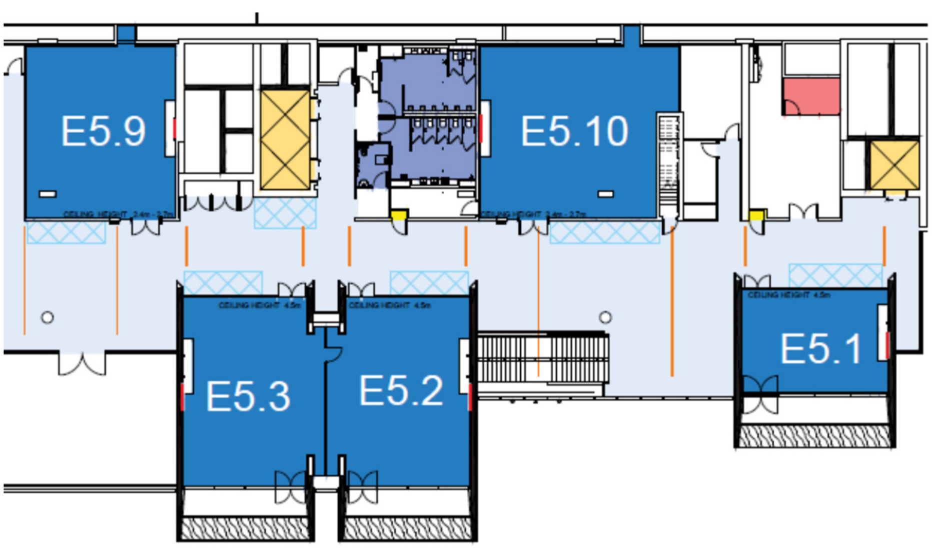


LEVEL 3: Cockle Bay Rooms and C3 Meeting Rooms



EXHIBITION CENTRE

LEVEL 5 – Women in Acoustics Luncheon – Rooms E5.2/E5.3



TECHNICAL PROGRAM CALENDAR
Acoustics 2023 Sydney
4-8 December 2023

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

Monday Morning

1aID 9:00 **Interdisciplinary:** Keynote: Exploring the Ocean with Sound: Telltale Acoustic Signatures of a Changing Ocean. Pyrmont Theatre

1aAA 10:15 **Architectural Acoustics:** Industry-Academia Collaboration on Architectural Acoustics I. Room C3.4

1aAB 10:20 **Animal Bioacoustics:** General Topics in Animal Bioacoustics (Poster Session). Prymont Foyer

1aBA 10:20 **Biomedical Acoustics:** General Topics in Biomedical Acoustics I: Imaging. Room C2.1

1aED 11:00 **Education in Acoustics and Musical Acoustics:** Connections Between Music and Math. Room C2.2

1aPP 10:20 **Psychological and Physiological Acoustics:** General Topics in Psychological and Physiological Acoustics (Poster Session). The Gallery

1aSA 11:00 **Structrual Acoustics and Vibration nad Physical Acoustics:** Non-Negative Acoustic Contribution Analysis. Room C2.5

1aSC 10:20 **Speech Communication:** Phonetics of Under-Documented Languages I (Poster Session). The Gallery

1aUW 10:20 **Underwater Acoustics:** Inversions of Underwater Sound. Room C3.6

Monday Afternoon

1pAA 12:55 **Architectural Acoustics:** Anomalous, Scattered and Steered Reflections. Room C3.4

1pAB 1:35 **Animal Bioacoustics:** Auditory Perception and Cognition in Animals. Room C2.3

1pAO 1:00 **Acoustical Oceanograpy and Underwater Acoustics:** Acoustic Sensing of the Indian/Southern Ocean. Room C3.5

1pBA 12:55 **Biomedical Acosutics and Physical Acoustics:** Ultrasound for Biomaterials and Bioprocessing. Room C2.1

1pCA 12:55 **Computational Acoustics:** Numerical Methods for Vibroacoustics of Underwater Structures. Room C2.6

1pEA 12:55 **Engineering Acoustics, Physical Acoustics, and Structural Acoustics and Vibration:** Acoustics in Spatiotemporally Varying Materials. Room C2.4

1pED 1:00 **Education in Acoustics:** General Topics in Acoustics Education. Room C2.2

1pNSa 12:55 **Noise, Engineering Acoustics, ASA Committee on Standards, and Physical Acoustics:** Measurement of Low-Frequency Sound and Standards. Room C3.3

1pNSb 1:00 **Noise:** Ground Transportation Noise. Room C3.2

1pPA 12:55 **Physical Acoustics, Architectural Acoustics, and Engineering Acoustics:** Acoustical Measurements and Sensors for Challenging Environments I. Room C3.1

1pPP 1:15 **Psychological and Physiological Acoustics:** Understanding Hearing in a Dynamic World. Cockle Bay 2

1pSA 1:00 **Structural Acoustics and Vibration:** General Topics in Structural Acoustics and Vibration. Room C2.5

1pSC 1:15 **Speech Communication:** Phonetics of Under-Documented Languages II. Cockle Bay 1

1pSP 1:00 **Signal Processing in Acoustics:** Signal Processing Poster Potpourri (Poster Session). The Gallery

1pUW 1:00 **Underwater Acoustics:** Underwater Acoustic Propagation and Modelling. Room C3.6

Tuesday Morning

2aAAa 8:00 **Architectural Acoustics and Noise:** Airborne and Impact Noise in Buildings I. Room C3.4

2aAAb 8:00 **Architectural Acoustics:** Student Design Competition. The Gallery

2aAB 7:55 **Animal Bioacoustics and Underwater Acoustics:** Session in Honor of Douglas H Cato I. Room C2.3

2aAO 8:55 **Acoustical Oceanography:** Bubbles from Seeps. Room C3.5

2aBA 9:00 **Biomedical Acoustics and Physical Acoustics:** Biomedical Acoustics in Ophthalmology I. Room C2.1

2aCA 7:55 **Computational Acoustics and Physical Acoustics:** Data-Driven Methods in Acoustics and Vibration I. Room C2.6

2aED 7:55 **Education in Acoustics and WESPAC:** Teaching Acoustics Across Trans-Disciplinary Areas. Room C2.4

2aNSa 7:55 **Noise:** Aeroacoustic Sources and Fields I. Room C3.3

2aNSb 10:35 **Noise and Physical Acoustics:** Sonic Boom I. Cockle Bay 1

2aPA 8:55 **Physical Acoustics and Biomedical Acoustics:** Multiphase Flow and Acoustics I. Room C3.1

2aPPa 8:00 **Psychological and Physiological Acoustics:** Understanding Hearing in a Dynamic World (Poster Session). The Gallery

2aPPb 8:00 **Psychological and Physiological Acoustics:** Auditory Cognition in Interactive Virtual Environments I. Cockle Bay 2

2aSA 9:55 **Structural Acoustics and Vibration, Engineering Acoustics, and Physical Acoustics:** Acoustic Metamaterials I. Room C2.5

2aSC 8:00 **Speech Communication:** Listening Challenges in Different Learning Environments. Cockle Bay 1

2aSP 8:35 **Signal Processing in Acoustics, Noise, and Physical Acoustics:** Signal Processing for Active Sound and Vibration Control I. Room C3.2

2aUW 8:15 **Underwater Acoustics and Acoustical Oceanography:** Effects of Shear Waves on Propagation and Scattering of Underwater Sound I. Room C3.6

Tuesday Afternoon

2pAA 1:00 **Architectural Acoustics and Noise:** Airborne and Impact Noise in Buildings II. Room C3.4

2pAB 12:55 **Animal Bioacoustics and Underwater Acoustics:** Session in Honor of Douglas H. Cato II. Room C2.3

2pAO 1:00 **Acoustical Oceanography:** Topics in Acoustical Oceanography. Room C3.5

2pBAa 1:00 **Biomedical Acoustics and Physical Acoustics:** Biomedical Acoustics in Ophthalmology II. Room C2.1

2pBAb 1:15 **Biomedical Acoustics and Physical Acoustics:** Novel Ultrasound Image Acquisition Technologies and Techniques. Room C2.2

2pCA 12:55 **Computational Acoustics and Physical Acoustics:** Data-Driven Methods in Acoustics and Vibration II. Room C2.6

2pNSa 12:55 **Noise:** Aeroacoustic Sources and Fields II. Room C3.3

2pNSb 2:00 **Noise and Physical Acoustics:** Sonic Boom II. Cockle Bay 1

2pPA 12:55 **Physical Acoustics, Architectural Acoustics, and Engineering Acoustics:** Acoustical Measurements and Sensors for Challenging Environments II. Room C3.1

2pPP 12:55 **Psychological and Physiological Acoustics:** Auditory Cognition in Interactive Virtual Environments II. Cockle Bay 2

2pSA 1:15 **Structural Acoustics and Vibration, Engineering Acoustics, and Physical Acoustics:** Acoustic Metamaterials II. Room C2.5

2pSC 1:00 **Speech Communication:** Speech Perception (Poster Session). The Gallery

2pSP 1:00 **Signal Processing in Acoustics, Noise, and Physical Acoustics:** Signal Processing for Active Sound and Vibration Control II. Room C3.2

2pUW 1:20 **Underwater Acoustics and Acoustical Oceanography:** Effects of Shear Waves on Propagation and Scattering of Underwater Sound II. Room C3.6

Wednesday Morning

3aAA 7:55 **Architectural Acoustics and Signal Processing in Acoustics:** Audio for Architectural Acoustics, Indoors and/or Outdoors I. Room C3.4

3aAB 8:40 **Animal Bioacoustics:** Acoustic Ecology and Biological Soundscapes I. Room C2.3

3aAO 8:00 **Acoustical Oceanography, Underwater Acoustics, and Physical Acoustics:** Observing the Ocean Acoustically using Submarine Cable Systems I. Room C3.5

3aBA 9:15 **Biomedical Acoustics and Physical Acoustics:** Bridging Preclinical and Clinical Acoustics I. Room C2.1

3aCA 8:00 **Computational Acoustics and Physical Acoustics:** Innovations in Computational Acoustics I. Room C2.6

3aEA 7:55 **Engineering Acoustics, Signal Processing in Acoustics, and Physical Acoustics:** Sound Field Manipulation for Personal Audio. Room C2.4

3aMU 7:55 **Musical Acoustics:** Indigenous Musical Instruments. Room C2.2

3aNSa 7:55 **Noise and Physical Acoustics:** Duct Noise Control- New Physical Mechanisms and Structures. Room C3.3

3aNSb 8:00 **Noise:** Community and Environmental Noise. Cockle Bay 1

3aPA 8:55 **Physical Acoustics and Biomedical Acoustics:** Multiphase Flow and Acoustics II. Room C3.1

3aPP 7:55 **Psychological and Physiological Acoustics:** Auditory Sensory Augmentation. Cockle Bay 2

3aSA 7:55 **Structural Acoustics and Vibration, Engineering Acoustics, and Physical Acoustics:** Acoustic Metamaterials III. Room C2.5

3aSC 8:00 **Speech Communication:** Speech Production I (Poster Session). The Gallery

3aSP 9:15 **Signal Processing in Acoustics:** Signal Processing Potpourri I. Room C3.2

3aUW 8:00 **Underwater Acoustics:** Underwater SONAR. Room C3.6

Wednesday Afternoon

3pAAa 1:00 **Architectural Acoustics and Signal Processing in Acoustics:** Audio for Architectural Acoustics, Indoors and/or Outdoors II. Room C3.4

3pAAb 2:40 **Architectural Acoustics:** Industry-Academia Collaboration on Architectural Acoustics II. Room C3.4

3pAB 1:00 **Animal Bioacoustics:** Acoustic Ecology and Biological Soundscapes II. Room C2.3

3pAO 1:00 **Acoustical Oceanography, Underwater Acoustics, and Physical Acoustics:** Observing the Ocean Acoustically using Submarine Cable Systems II. Room C3.5

3pBAa 1:35 **Biomedical Acoustics and Physical Acoustics:** Biomedical Acoustics in Pulmonology. Room C2.2

3pBAb 1:40 **Biomedical Acoustics and Physical Acoustics:** Bridging Preclinical and Clinical Acoustics II. Room C2.1

3pCA 1:00 **Computational Acoustics and Physical Acoustics:** Innovations in Computational Acoustics II. Room C2.6

3pEA 1:40 **Engineering Acoustics:** Transducer Design and Evaluation. Room C2.4

3pNSa 1:00 **Noise:** Soundscapes. Room C3.3

3pNSb 3:20 **Noise, Architectural Acoustics, and Physical Acoustics:** Forensic Acoustics: What's that Noise? Room C3.3

3pPA 12:55 **Physical Acoustics and Engineering Acoustics:** Artificial Intelligence for Metamaterials. Room C3.1

3pPPa 1:00 **Psychological and Physiological Acoustics:** Binaural Listening and Scene Analysis (Poster Session). The Gallery

3pPPb 2:20 **Psychological and Physiological Acoustics:** Cochlear Mechanics and Physiology (Poster Session). The Gallery

3pSC 2:20 **Speech Communication:** Speech Production II (Poster Session). The Gallery

3pSP 1:00 **Signal Processing in Acoustics:** Signal Processing Potpourri II. Room C3.2

3pUW 12:55 **Underwater Acoustics and Signal Processing in Acoustics:** Mobile Underwater Acoustic Sensor Networks: Communication, Localization, and Networking Challenges I. Room C3.6

Thursday Morning

4aID 11:00 **Interdisciplinary:** Keynote: The Listening Brain's Response to Adversity. Cockle Bay

4aAA 7:35 **Architectural Acoustics, Psychological and Physiological Acoustics, and Signal Processing in Acoustics:** Three-Dimensional Sound Display and Analysis for Virtual Auditory Immersive Environments I. Room C3.5

4aBA 8:55 **Biomedical Acoustics and Physical Acoustics:** Cavitation Therapies for Cancer Treatment I. Room C2.1

4aPA 8:00 **Physical Acoustics and Structural Acoustics and Vibration:** Nonlinear Acoustics in Solids. Room C3.1

4aPP 7:55 **Psychological and Physiological Acoustics:** Physiology and Psychophysics of Predictive Auditory Scene Analysis and Object Formation. Room C2.2/C2.3

4aSA 8:00 **Structural Acoustics and Vibration:** Acoustic Treatments and Vibration Isolation. Room C2.5

4aUW 8:55 **Underwater Acoustics, Acoustical Oceanography, and Physical Acoustics:** Acoustics of Extreme Weather Events. Room C3.6

Thursday Afternoon

4pAAa 1:00 **Architectural Acoustics, Psychological and Physiological Acoustics, and Signal Processing in Acoustics:** Three-Dimensional Sound Display and Analysis for Virtual Auditory Immersive Environments II. Room C3.4

4pAAb 1:00 **Architectural Acoustics and Noise:** Soundwalking in Sydney. Room C3.5

4pAB 12:55 **Animal Bioacoustics:** Marine Bioacoustics in the West Pacific. Room C2.3

4pBA 1:00 **Biomedical Acoustics and Physical Acoustics:** Cavitation Therapies for Cancer Treatment II. Room C2.1

4pCAa 12:55 **Computational Acoustics and Physical Acoustics:** Computational Aspects of Spatial Audio. Room C2.6

4pCAb 3:15 **Computational Acoustics and Physical Acoustics:** Computational Aeroacoustics. Room C2.6

4pEA 1:00 **Engineering Acoustics:** Topics in Engineering Acoustics. Room C2.4

4pMU 1:00 **Musical Acoustics:** General Topics in Musical Acoustics. Room C2.2

4pNSa 1:00 **Noise:** Assorted Topics on Noise. Room C3.3

4pNSb 1:55 **Noise:** Drone Noise. Cockle Bay 2

4pPA 1:00 **Physical Acoustics:** Acoustical Methods for Materials Analytics. Room C3.1

4pPP 1:35 **Psychological and Physiological Acoustics and Animal Bioacoustics:** Comparative Models of Hearing Loss. Cockle Bay 1

4pSA 1:00 **Structural Acoustics and Vibration and Physical Acoustics:** Structural Acoustics and Vibration in Buildings. Room C2.5

4pSP 12:55 **Signal Processing in Acoustics:** Signal Processing Potpourri III. Room C3.2

4pUWa 12:55 **Underwater Acoustics and Signal Processing in Acoustics:** Mobile Underwater Acoustic Sensor Networks: Communication, Localization and Networking Challenges II. Room C3.6

4pUWb 2:40 **Underwater Acoustics:** Underwater Noisescapes. Room C3.5

Friday Morning

5aAA 8:00 **Architectural Acoustics:** General Topics in Architectural Acoustics. Room C3.5

5aAB 7:55 **Animal Bioacoustics, Acoustical Oceanography, Signal Processing in Acoustics, and Underwater Acoustics:** Fisheries and Marine Park Management I. Room C2.3

5aBA 8:20 **Biomedical Acoustics:** General Topics in Biomedical Acoustics II: Bone and Soft Tissue Properties. Room C2.1

5aCA 8:00 **Computational Acoustics:** Topics in Computational Acoustics. Room C2.6

5aMU 7:55 **Musical Acoustics:** Player-Instrument Interaction I. Room C2.2

5aNSa 7:55 **Noise and Physical Acoustics:** Jet Noise. Room C3.3

5aNSb 8:00 **Noise:** Construction Noise. Room C3.2

5aPA 8:00 **Physical Acoustics:** General Topics in Physical Acoustics. Room C2.4

5aPP 7:55 **Psychological and Physiological Acoustics:** Top-Down and Bottom-Up Processing in Individuals with Normal Hearing and Hearing Difficulties. Cockle Bay 2

5aSCa 8:40 **Speech Communication:** Phonetics of Emerging Varieties of English. Cockle Bay 1

5aSCb 10:20 **Speech Communication:** Voice Therapy: Science and Clinical Efficacy I. Cockle Bay 1

5aSP 8:15 **Signal Processing in Acoustics and Physical Acoustics:** Passive Acoustic Sensing of the Underwater Environment Using Gliders and Uncrewed Renewable Energy Powered Surface Vessels. Room C3.1

5aUW 8:00 **Underwater Acoustics:** Computers in Underwater Acoustics. Room C3.6

Friday Afternoon

5pAB 12:55 **Animal Bioacoustics, Acoustical Oceanography, Signal Processing in Acoustics, and Underwater Acoustics:** Fisheries and Marine Park Management II. Room C2.3

5pMU 1:00 **Musical Acoustics:** Player-Instrument Interaction II. Room C2.2

5pNSa 12:55 **Noise and Physical Acoustics:** Rocket Noise. Room C3.3

5pNSb 12:55 **Noise and Physical Acoustics:** Benefits and Drawbacks of Non-Typical Hearing Protectors. Room C3.2

5pSC 1:00 **Speech Communication:** Voice Therapy: Science and Clinical Efficacy II. Cockle Bay 1

5pUW 1:00 **Underwater Acoustics:** Topics in Underwater Sound. Room C

Acoustics 2023 Sydney

Acoustics 2023 Sydney will be held Monday through Friday, 4-8 December 2023 at the International Convention Centre Sydney (ICC Sydney), Sydney, Australia

SECTION HEADINGS

1. REGISTRATION
2. TECHNICAL SESSIONS
3. TECHNICAL SESSION DESIGNATIONS
4. SPEAKER PREPARATION CENTRE AND SPEAKER MANAGEMENT SYSTEM
5. EXHIBIT AND EXHIBIT OPENING RECEPTION
6. TECHNICAL COMMITTEE OPEN MEETINGS
7. WIFI
8. PLENARY SESSION AND AWARDS CEREMONY
9. COFFEE BREAKS
10. PARENTS ROOM
11. SOCIAL
12. WOMEN IN ACOUSTICS LUNCHEON
13. STUDENT EVENTS: NEW STUDENTS/FIRST-TIME ATTENDEE ORIENTATION, MEET AND GREET, STUDENT RECEPTION
14. JAM
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16. PROCEEDINGS OF MEETINGS ON ACOUSTICS (POMA)
17. TECHNICAL PROGRAM ORGANIZING COMMITTEE
18. MEETING ORGANIZING COMMITTEE
19. PHOTOGRAPHING AND RECORDING
20. ABSTRACT ERRATA
21. GUIDELINES FOR ORAL PRESENTATIONS,
22. SUGGESTIONS FOR EFFECTIVE POSTER PRESENTATIONS
23. 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.

The Registration Desk will be located on the Ground Floor of the Convention Center which can be accessed from Tumbalong Boulevard, the Convention light rail crossing, the vehicle drop off point at Iron Wharf Place and Car Park 1 (P1). See the floor plans on pages A5-A6.

The registration desk will open on Sunday and will be open each day of the meeting.

Meeting attendees who have pre-registered may pick up their badges and registration materials at the pre-registration desk. On-site registrants may pay the registration fee using Visa, MasterCard and American Express credit cards. The preferred method is by credit card but cash payments in AUD will be accepted.

2. TECHNICAL SESSIONS

The technical program includes over 1150 abstracts. Floor plans of the International Convention Center appear on pages A5 and A6. 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, 4 December
2-Tuesday, 5 December
3-Wednesday, 6 December
4-Thursday, 7 December
5-Friday, 8 December

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

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

In sessions where the same group is the primary organizer of more than one session scheduled in the same morning or afternoon, a fifth character, either lower-case “a,” 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. SPEAKER PREPARATION CENTRE AND SPEAKER MANAGEMENT SYSTEM

The Speaker Preparation Centre is located on the Ground floor to the right of the registration desk. Hours are Sunday, 12:00 p.m. to 5:00 p.m. (12:00 to 17:00) and Monday through Friday, 7:30 a.m. to 5:00 p.m. (07:30 to 17:00).

This is the location where presenting authors must upload their slide presentations not later than 2 hours before the start of the sessions in which they are scheduled to present. There will be one ICC staff member present to assist you on Sunday and two on Monday through Friday.

5. EXHIBITION AND EXHIBITION OPENING RECEPTION

An instrument and equipment exhibition will be located in The Gallery on the 2nd floor and will open on Monday, 4 December, with an evening reception serving a complimentary drink. Exhibit hours are Monday, 4 December, 6:00 p.m. to 7:00 p.m. (18:00 to 19:00), Tuesday, 5 December, 9:00 a.m. to 5:00 p.m. (9:00 to 17:00), and Wednesday, 6 December, 9:00 a.m. to 12:00 noon.

The Exhibit will include manufacturers and suppliers to showcase the latest developments in acoustic instrumentation, software and noise and vibration control products.

6. TECHNICAL COMMITTEE OPEN MEETINGS

Technical Committees will hold open meetings on Tuesday, Wednesday, and Thursday. The schedule and rooms for each committee meeting are given on page A4. These are working, collegial meetings. All meeting participants are cordially invited to attend these meetings and to participate actively in the discussions.

7. WIFI

Complimentary access to WIFI is available throughout the Convention Center.

8. PLENARY SESSION AND AWARDS CEREMONY

A joint ASA/AAS plenary session will be held Wednesday, 6 December, at 4:30 p.m. in the Cockle Bay Room.

Certificates will be presented to ASA Fellows elected at the Chicago meeting and a presentation will be made in honor of Anne Cutler, the 2020 ASA Speech Communication Silver Medal awardee.

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

9. COFFEE BREAKS

Morning and afternoon coffee/tea breaks will be held daily. Monday to Wednesday breaks will be held in The Gallery. Breaks on Thursday and Friday will be held in the Cockle Bay foyer. Break schedule: Monday: 10:00–10:20 and 2:20–3:40; Tuesday and Wednesday: 9:20–10:40 and 2:20–3:40; Wednesday: 9:20–10:40 and 2:20–3:40; Thursday: 10:20–10:40 and 2:20–3:40; Friday: 9:20–10:40 and 2:20–3:40.

10. PARENTS ROOM

A Parents Room for meeting attendees will be available Monday to Friday, 8:00 a.m. to 5:00 p.m. (08:00 to 17:00) on the Ground floor and Level two of ICC Sydney. The accessible parenting rooms are equipped with a feeding area, change table and food preparation area.

11. SOCIAL

A Social will be held on Wednesday evening, 6:00 p.m. to 7:30 p.m. (18:00 to 19:30) in the Grand Ballroom on the 5th floor.

The Social will 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.

12. 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, 4 December, from 5:30 p.m. to 6:00 p.m. in Room C2.1. The Student Meet and Greet from will be held on Monday, 6:00 p.m. to 7:30 p.m. (18:00 to 19:30) in The Gallery where refreshments and a cash bar will be available.

The Students' Reception will be held on Wednesday, 6 December from 6:00 p.m. to 7:00 p.m. (18:00 to 19:00) in The Gallery on Convention Level 2. This reception will provide an opportunity for students to meet informally with fellow students and other members of the Acoustical Society of American and the Australian Acoustical Society. All students are encouraged to attend, especially students who are first time attendees or those from smaller universities.

13. WOMEN IN ACOUSTICS LUNCHEON

The Women in Acoustics luncheon will be held at 11:30 a.m. on Wednesday, 6 December, in rooms E5.2 and E5.3 on Level 3 of the Exhibition Mezzanine.

14. JAM SESSION

You are invited to the Foundry616 Jazz Club on Tuesday night, 5 December, from 7:00 p.m. to midnight (19:00 to 24:00). The Foundry616 Jazz Club is located at 616-620 Harris St. Ultimo, Sydney. 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, and all-around good times. Don't miss out.

15. ACCOMPANYING PERSONS PROGRAM

Accompanying persons who are attendees that will not participate in the technical sessions are welcome at the meeting. The on-site registration fee for accompanying persons is $535 AUD/$357 USD. This entitles you access to the accompanying persons program, social on Wednesday, the Jam Session on Tuesday, and the Plenary Session on Wednesday.

Accompanying persons will gather in Room E3.1/E32. in the Exhibition Building at the ICC on Monday, 4 December, for breakfast. On Tuesday and Wednesday, accompanying persons can meet in Business Center Room 2 to meet and plan daily outings with other attendees. No breakfast will be provided on Tuesday and Wednesday.

16. PROCEEDINGS OF MEETINGS ON ACOUSTICS (POMA)

Acoustics 2023 Sydney will have a published proceedings and submission is optional. The proceedings will be a separate volume of the online journal, "Proceedings of Meetings on Acoustics" (POMA). This is an open access journal, so that its articles are available in pdf format for downloading without charge to anyone in the world. Authors who are scheduled to present papers at the meeting are encouraged to prepare a suitable version in pdf format that will appear in POMA. It is not necessary to wait until after the meeting to submit one's paper to POMA. Further information regarding POMA can be found at the site http://acousticsauthors.org. Published papers from previous meeting can be seen at the site http://asadl/poma.

17. TECHNICAL PROGRAM ORGANIZING COMMITTEE

James H. Miller, ASA Technical Program Cochair, Benjamin Halkon, AAS Technical Program Cochair; Christopher Bassett, Acoustical Oceanography; Shane Guan, Animal Bioacoustics; Brandon Cudequest, David Manley, Architectural Acoustics; Libertario Demi, Kang Kim, Biomedical Acoustics; Amanda Hanford.

Computational Acoustics; Thomas Blanford, Engineering Acoustics; Daniel Russell, Education in Acoustics; Stephen Tufte, Kurt Hoffman, Musical Acoustics; James Phillips, Hales Swift, Aaron Vaughn, Marion Burgess, Noise; Raphael Hermann, Joel Lonzaga, Physical Acoustics; Gregory Ellis, Psychological and Physiological Acoustics; Kai Gemba, Trevor Jerome, Signal Processing in Acoustics; Pasquale Bottalico, Matthew Masapollo, Benjamin Tucker, Speech Communication; Anthony Bonomo, Stephanie Konarski, Benjamin Halkon, Structural Acoustics and Vibration; David Dall'Osto, Underwater Acoustics; Brijonnay Madrigal, Student Council; Nancy Blair-DeLeon, ASA Standards.

18. ACOUSTICS 2023 SYDNEY ORGANIZING COMMITTEE

Cochairs: Marcia Isakson, ASA; Jeffrey Parnell, AAS; Technical Program Cochairs: James Miller, ASA, Benjamin Halkon, AAS; ASA Headquarters Representative: Susan Fox; WESPAC Representative: Marion Burgess; ASA Co-Treasurer: Judy R. Dubno; AAS Co-Treasurer: John Wasserman; Exhibition Manager: Julie Sobolewski; Student Coordinators: Brijonnay Madrigal, ASA, Adrian Morris, AAS; Sarah Weatherby, Celina Bewick, Arinex, Professional Conference Organizers.

19. PHOTOGRAPHING AND RECORDING

Photographing and recording during regular sessions are not permitted without prior permission from the Organizing Committee.

20. ABSTRACT ERRATA

This meeting program is Part 2 of the October 2023 issue of The Journal of the Acoustical Society of America. Corrections, for printer's errors only, may be submitted for publication in the Errata section of the Journal.

21. GUIDELINES FOR ORAL PRESENTATIONS,

Preparation of Visual Aids

- 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, check 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.
- Each time you display a visual aid the audience needs time to interpret it. Describe the abscissa, ordinate, units, and the legend for each figure. If the shape of a curve or some other feature is important, tell the audience what they should observe to grasp the point. They won't have time to figure it out for themselves. A popular myth is that a technical audience requires a lot of technical details. Less can be more.
- Turn off your cell phone prior to your talk and put it away from your body. Cell phones can interfere with the speakers and the wireless microphone.

22. 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

- Poster boards (5.9 ft wide by 3.9 ft high; 1800 mm wide x 1200 mm 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.

23. DATES OF FUTURE ASA MEETINGS

For further information on any ASA meeting, contact the Acoustical Society of America, 1305

Walt Whitman Road, Suite 110, Melville, NY 11747-4300; E-mail: asa@acousticalsociety.org

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

187th Meeting – fall 2024, Virtual Meeting

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

PHYSICAL ACOUSTICS SUMMER SCHOOL

PASS
2024

JUNE 3-7, 2024

The Inn at Ole Miss Hotel & Conference Center
located on The University of Mississippi Oxford campus

Purpose	The purpose of the Summer School is to bring graduate students, distinguished lecturers, and discussion leaders together to discuss a wide variety of subjects in physical acoustics. PASS will give students opportunities to meet experts and discuss topics they would not ordinarily encounter in their own colleges and universities. Participation, including students, lecturers, and discussion leaders, is limited to fifty. Full-time participation is required.
Students	The focus of PASS is on intermediate and advanced graduate students.
Lecturers & Discussion Leaders	The PASS Organizing Committee selects approximately eight lecture topics and appropriate lecturers and discussion leaders.
Costs	Participants provide their own transportation. Oxford is about 80 miles from Memphis International Airport (MEM). Some free van transportation will be available from MEM to UM. There is a $350 Student Registration Fee.
Room & Board	Room and board, based on standard multiple-occupancy for students, will be supported by sponsors.
Program	Program information will be available on the web page (below) at the National Center for Physical Acoustics (NCPA) at The University of Mississippi.
Application Deadline	Complete applications for the 2024 Physical Acoustics Summer School must be received no later than **Thursday, February 8, 2024**.
Applications	All participants must have a completed Application Form on file and also provide an unofficial transcript and one professional reference letter. Copies of the Announcement, Application Form, and Preliminary Schedule will be available at https://ncpa.olemiss.edu/pass-2024/. The initial contact is Debra Bos (dperrier@olemiss.edu).

ASA School 2024

Living in the Acoustic Environment

11-12 May 2024

Ottawa, Ontario, Canada

ASA School 2024 is an Acoustical Society of America event for graduate students and early career acousticians in all areas of acoustics to learn about and discuss a wide variety of topics related to the interdisciplinary theme *Living in the Acoustic Environment*. ASA School 2024 follows on the success of five previous ASA Schools starting in 2012, and will provide opportunities for meeting instructors and fellow attendees, mentoring, discussing research topics, and developing collaborations and professional relationships within acoustics.

Program and Costs

ASA School 2024 will take place at the Westin Ottawa Hotel, which is the ASA meeting hotel. Lectures and demonstrations followed by discussions will be given by distinguished acousticians in a two-day program covering topics in *acoustical oceanography, animal bioacoustics, computational acoustics, musical acoustics, physical acoustics, signal processing in acoustics, structural acoustics and vibration, and underwater acoustics*. Although ASA School 2024 will focus primarily on these 8 technical areas, graduate students and early career professionals in all areas of acoustics are encouraged to attend to achieve a broader understanding of the diverse field of acoustics.

The registration fee is $50. Hotel rooms at the Westin Ottawa Hotel for two nights (double occupancy) and meals will be provided by ASA. Participants are responsible for their own travel costs and arrangements including transportation to the Westin Ottawa Hotel.

Participants and Requirements

ASA School 2024 is targeted to graduate students and early career acousticians (within 3 years of terminal degree) in all areas of acoustics. Attendance is limited to 60 participants who are expected to attend all School events and the ASA meeting immediately following on 13-17 May 2024. ASA School attendees are required to be an author or co-author on an abstract for presentation at the ASA Ottawa meeting.

Application and Deadlines

The application form and preliminary program will be available online in December 2023, at www.AcousticalSociety.org.

Session 1aAA

Architectural Acoustics: Industry-Academia Collaboration on Architectural Acoustics I

Lucky Tsaih, Cochair
Architecture, National Taiwan University of Science and Technology, 43 Keelung Rd. Sec. 4, Taipei 106, Taiwan

Wei-Hwa Chiang, Cochair
Architecture, National Taiwan University of Science and Technology, 43 Keelung Rd. Sec. 4, Taipei 106335, Taiwan

Chair's Introduction—10:15

Invited Papers

10:20

1aAA1. Integrating generative parametric design workflow in the terraced concert hall design process. Wei-Hwa Chiang (Architecture, National Taiwan Univ. of Sci. and Technol., 43 Keelung Rd. Sec. 4, Taipei 106335, Taiwan, whch@mail.ntust.edu.tw), Hung-Yi Lai, Wei-Lin Hwang, and Xin-You Lai (Architecture, National Taiwan Univ. of Sci. and Technol., Taipei, Taiwan)

Compared to shoe box halls, terraced concert halls are popular due to their proximity to the stage and expressive design. However, its complex geometry requires delicate adjustments during the design process to effectively deliver a design that satisfies both the needs of acoustics and creativity. The study aims to explore the integration of generative parametric design workflow into the design of terraced concert halls and apply the workflow to an actual competition project. This workflow consists of three steps: defining basic parameters and reference geometries, generating and composing geometric elements with synchronized acoustic and geometric verification, and conducting performance evaluation of the results. The workflow enables integration of diverse viewpoints for the competition project, covering both the acoustics of the concert hall and the spatial integration with the surroundings. The outcome is a multi-layered, terraced concert hall with acoustic performance comparable to a traditional shoebox hall, and a metaphorical image of the topography of the city in which the concert hall is located.

10:40

1aAA2. DLR Group collaboration with academia and the University of Nebraska Lincoln Durham School of Architectural Engineering and Construction. David Manley (DLR Group, 6457 Frances St., Omaha, NE 68106, dmanley@dlrgroup.com) and Lily Wang (Architectural Eng. and Construction, Univ. of Nebraska, Omaha, NE)

Recognizing that the future of the acoustical design profession lies within the walls of our academic institutions is foundational to DLR Group's commitment to collaboration, mentorship, and internship programs for students at the University of Nebraska Lincoln Durham School of Architectural Engineering and Construction. The 25-year industry-academic relationship has resulted in graduating students who are able to enter industry running, with minimal technical training on day one of employment. Benefits to the student experience include mentorship, paid summer internships, and safety of mind knowing students from the university are highly sought after. Perspectives from both DLR Group and the University of Nebraska will be presented.

11:00

1aAA3. Industry-academia collaboration to develop new building standards. Sunit Girdhar (Veneklasen Assoc., 1711 16th St., Santa Monica, CA 90404, sgirdhar@veneklasen.com), Andrew Barnard (Penn State Univ., University Park, PA), John LoVerde, and Wayland Dong (Veneklasen Assoc., Santa Monica, CA)

Industry and academia must go hand-in-hand to progress science and knowledge. Often, academic solutions may not be the most optimal solution for the field where there may be a time/ equipment/ training restriction. For our work, we focused on improving the building acoustic standards—specifically the impact performance of floor-ceiling assemblies–through industry and academic collaboration. The research student at Michigan Technological University worked closely with Veneklasen Associates through their (non-profit) funding arm, as well as through summer internships to learn the existing limitations for impact performance field tests. The student used this knowledge to develop a test method with improved reproducibility while ensuring that the test is fast and minimal new training is required. The student now works with Veneklasen Associates after graduating to continue developing this test method.

11:20

1aAA4. An expanding pipeline: 20 + years of Acentech internships. Benjamin E. Markham (Acentech, 33 Moulton St., Cambridge, MA 02138, bmarkham@acentech.com)

Acoustics interns at Acentech, a multidisciplinary acoustics, technology, and noise & vibration control consultancy based in Cambridge, Massachusetts, USA, have included undergraduate and graduate students as well as graduates in many fields of engineering, physics, and architecture. Acentech interns work on a deliberate mix of both project work and internal research and development, often inspired by research presented by academics in our field. For a time, interns were typically from one of approximately a dozen US-based graduate programs in acoustic, and more rarely, from one or two undergraduate programs with acoustics concentrations. In recent years, successful interns have hailed from an increasingly wide array of university programs, and the results have exceeded expectations: an increasingly diverse pipeline of skilled, intellectually curious individuals with a passion for music, buildings, and acoustical design.

MONDAY MORNING, 4 DECEMBER 2023 PYRMONT FOYER, 10:20 A.M. TO 12:20 P.M.

Session 1aAB

Animal Bioacoustics: General Topics in Animal Bioacoustics (Poster Session)

Laura Kloepper, Chair
Biological Sciences, University of New Hampshire, 38 Academic Way, Durham, NH 03824

All posters will be on display and all authors will be at their posters from 10:20 a.m. to 12:20 p.m.

Contributed Papers

1aAB1. Planning research of whale sound detection system. Bok Kyoung Choi (Korea Inst. of Ocean Sci. & Technol., 385, Haeyang-ro, Yeongdo-gu, Busan 49111, South Korea, bkchoi@kiost.ac.kr), Byoung-Nam Kim, Seong Hyeon Kim, and Eung Kim (Korea Inst. of Ocean Sci. & Technol., Busan, South Korea)

In maritime countries such as the United States and European countries, studies on cetacean resource distribution, vocalization characteristics, auditory characteristics, and habitat environment are actively being conducted in order to permanently preserve cetacean resources. It is necessary to establish a real-time buoy-type marine life observation system for the surrounding waters of the Korean Peninsula, and the components of the real-time buoy-type marine life observation system that can observe the underwater acoustic environment require a line-type underwater acoustic sensor, a large-capacity file real-time transmission system, and a high-capacity battery. Planning contents for the development of the whale sound detection system include realizing scientific whale tourism by combining marine science technology (MT + ICT), presenting a plan for constructing a real-time buoy-type observation system for whales based on ICT, and presenting an analysis algorithm for identifying whale ecology through observation and analysis of radioactive sound characteristics and acoustic characteristics of whales in real time. These research results can be developed as new tourism contents in the southeast sea of Korea, contribute to expanding the base of whale-watching tourism business, and provide information on migratory routes and resource distribution by identifying the sonic characteristics of cetaceans.

1aAB2. Bottlenose dolphin (*Tursiops truncatus*) temporary threshold shift in response to frequency-modulated and pure-tone exposures centered at 28 kHz. Madilyn R. Pardini (National Marine Mammal Foundation, 2240 Shelter Island Dr., Ste. 200, San Diego, CA 92106, madilyn.pardini@nmmpfoundation.org), Jason Mulsow (National Marine Mammal Foundation, San Diego, CA), Carolyn Schlundt (Peraton, San Diego, CA), Alyssa Accomando (National Marine Mammal Foundation, San Diego, CA), and James Finneran (Naval Information Warfare Ctr. Pacific, US Navy Marine Mammal Program, San Diego, CA)

There is a scarcity of marine mammal temporary threshold shift (TTS) data for noise exposures representative of naval continuous active sonar (CAS). In this study, bottlenose dolphin ($n = 2$) TTS was measured following exposure to frequency-modulated (FM) tones with temporal characteristics representative of CAS, but frequencies shifted to the dolphins' region of best hearing sensitivity (20–40 kHz). Exposures were conducted at multiple sound pressure levels and durations ranging from 2 to 60 min. Additional exposures to 28-kHz pure tones at matched cumulative sound exposure levels examined the effect of exposure bandwidth on TTS. Hearing thresholds were measured using behavioral methods and supplemented by electrophysiological auditory brainstem response and auditory steady-state response methods. Larger TTS magnitudes were observed for the FM versus the pure-tone exposures in both dolphins. A consistent pattern of TTS was not observed using either electrophysiological method, contrary to what has been noted in some previous studies. Future testing will evaluate TTS following exposure to simulated CAS centered near 3 kHz, at the lower-frequency end of the bottlenose dolphin's hearing range but representative of the actual frequencies of naval CAS. [Work funded by US Navy Living Marine Resources.]

1aAB3. An expanding mechanical role for the tiny mammalian endolymphatic duct: 2. An explanation for the beaching of whales–failure of osmotic pressure counterbalancing mechanism in the cochlea causing acute vertigo attack. Eric L. LePage (Director, innerearmechanisms.org, PO Box 2564, Mount Claremont, Western Australia 6010, Australia, ericlepage@innerearmechanisms.org)

Endolymphatic Hydrops (EH) describes a swelling of the endolymphatic duct (ED). It is a key correlate of human Ménière's disease featuring incapacitating giddy attacks. Direct measurements by micropuncture to be presented support the hypothesis that swelling occurs because water flows through aquaporins into the ED under an osmotic gradient to stabilize basilar membrane position against deep dive pressures. Patients having vestibular attacks often display nystagmus if not also perilymphatic fistulae or dehiscences. As shown by Ketten *et al*. (2004), stranded whales also exhibit nystagmus and also intracochlear bleeding. Here, EH is modelled as a normal rate-limited osmotic mechanism which likely evolved to cope with synoptic variation. It has been extensively adapted by beaked whales to allow diving to feed on squid at great depths—where they still need to hear. Their more-robust EDs normally cope with very slow equalization against megapascal pressures. An unexpected undersea pressure transient may trigger perilymph-endolymph membrane rupture and consequent vertigo attack, well documented in humans. Due to acoustic spread, whole pods may be variously affected. Humans need to *remain stationary* while waiting hours for their attack to subside. Marine mammals with limited oxygen stores may panic and surface too quickly, accounting for life-threatening decompression sickness.

1aAB4. Baleen whale sound perception and anthropogenic ocean noise: Analyzing the behavioral response of migrating humpback whales to tones and airguns. Kelsey Stone (The Univ. of Queensland, School of Biological Sci., Brisbane, Queensland 4072, Australia, kelsey.stone@uq.edu.au), Riona McNamara, Michael Noad, and Rebecca Dunlop (The Univ. of Queensland, Brisbane, Queensland, Australia)

Baleen whale sound perception is an important factor to consider when predicting and mitigating the impacts of anthropogenic ocean noise. Some sound types, for example predator calls, may elicit greater responses, meaning whale behavior is not only driven by proximity and received level, but other factors. Here, we compared the response of migrating eastern Australian humpback whales (*Megaptera novaeangliae*) to tones and airguns. We tested the hypothesis that groups would have a greater response magnitude to higher frequency tones given they sound like killer whale whistles, a known predator. An airgun shot, however, has a similar sound profile to a breaching whale. Whale groups were exposed to either a 20 cubic inch air gun or one of four tonal frequencies (250 Hz, 1 kHz, 4 kHz, and 16 kHz), and their behavior was compared before and during the sound exposure. Results show that the 16 kHz tone elicited the largest response as measured by alterations in group movement and dive behavior. Their behavioral changes to lower frequencies (250 Hz and 1 kHz) were similar in magnitude to their responses to airguns. Results suggest humpback whales may perceive certain sound types as threatening, eliciting more dramatic behavioral changes than conspecific-like sounds.

1aAB5. The phase shift of the envelope-following response as an indicator of the traveling wave propagation in the beluga cochlea. Dmitry Nechaev (Lab. of Vertebrate Sensory Systems, Inst. of Ecology and Evolution RAS, Moscow, Russian Federation, dm.nechaev@yandex.ru) and Alexander Supin (Lab. of Vertebrate Sensory Systems, Inst. of Ecology and Evolution RAS, Moscow, Russian Federation)

Auditory system of the toothed whales features a number of modifications that allow to adapt to aquatic life and echolocation. One of unanswered questions is how the characteristics of the traveling wave changed due to extension of frequency range. We hypothesize that the phase shift of the envelope-following response (EFR) reflects the traveling wave propagation along the basilar membrane in the cochlea. EFRs were non-invasively recorded in the beluga whale. The sound stimuli were tone burst of carrier frequencies of 16 to 128 kHz and envelope frequency of 1 kHz. Thus, all EFR contained the fluctuations with 1 kHz frequency. The EFR phase shift depended on the carrier frequency and the sound intensity: the higher intensity and frequency, the greater the phase shift. If it is accepted that at the EFR threshold the phase depends only on the carrier frequency, the phase shift allows to calculate the time of the traveling wave propagation along the basilar membrane. The travelling wave time between the regions corresponding to 16 and 128 kHz was approximately 0.8 ms.

1aAB6. Forward masking of the auditory evoked potentials in a bottlenose dolphin at monaural and dichotic auditory stimulation. Evgeniya Sysueva (Lab. of Vertebrate Sensory Systems, A.N. Severtsov Inst. of Ecology and Evolution of the Russian Acad. of Sci., Leninsky Prospect, 33, Moscow 119071, Russian Federation, evgeniasysueva@gmail.com) and Vladimir Popov (Lab. of Vertebrate Sensory Systems, A.N. Severtsov Inst. of Ecology and Evolution of the Russian Acad. of Sci., Moscow, Russian Federation)

Short-latency auditory evoked potentials (AEPs) to paired sound pulses (the leading stimulus (masker) and the delayed stimulus (test)) were recorded non-invasively in a bottlenose dolphin Tursiops truncatus. The stimuli were played through transducers contacting the left and right acoustic windows at the lower jaw. Two types of stimulation were used: monaurally (the both stimuli played through one and the same transducer) and dichotically (the stimuli played through different transducers, contacting the left and right acoustic window). The masker and test stimuli were equal in level and duration characteristics. The inter-stimulus delay varied from 0.15 to 10 ms. At the monaural stimulation, the suppression of the test stimulus was constant at interstimulus intervals from 0.15 to 0.5 ms; at longer intervals, the test response recovered. At the dichotic stimulation, the deepest suppression of the test response appeared at an interval of 0.5 ms; the test response recovered at both shorter and longer intervals. The complete recovery appeared at intervals as short as 0.15 ms and as long as 2 ms. Implications of the found regularities for the preceding effect and biosonar is discussed. [Work supported by RSF (project No. 22-25-00025).]

1aAB7. Acoustical and visual monitoring of the discharge process of the last captive Indo-Pacific Bottlenose Dolphin in South Korea. Changsoo Kim (Res. Inst. for Basic Sci., Jeju National Univ., Dept. of Ocean System Eng., Jeju-si, Jeju-do 63243, Korea (the Republic of), yustchang@gamil.com), Byung Yeop Kim (Dept. of Marine Industrial and Maritime Police, Jeju National Univ., Jeju-si, Jeju-do, Korea (the Republic of)), Seung Uk I. M., and Dong-Guk Paeng (Ocean System Eng., Jeju National Univ., Jeju-si, Jeju-do, South Korea)

The discharge of Bibong, a 23-year-old Indo-Pacific Bottlenose Dolphin, marked the end of its 17-year aquatic show life in an aquarium on November 16, 2022. The three-month adaptation training included a temporary retreat due to typhoon HINNAMNOR. To continuously capture underwater sound, two hydrophones were alternately used due to limited operation time. Additionally, two CCTV cameras and two drones were employed to record events around the discharge cage, with one observer per drone. Using a CNN-based whistle profile extractor, whistle sounds were automatically detected and exported as time-stamped data. The corresponding CCTV and drone clips were examined for each whistle encounter. Our study presents the whistle sound encounter pattern and the synchronized acoustical and video data. These findings offer valuable insights into understanding the discharge process, serving as a reference for future discharge initiatives and welfare assessments of captive marine mammals.

1aAB8. Vocalization behavior response of captive bottlenose dolphins (*Tursiops truncatus*) to playbacks of pile driving noise. Chuan-Kuan Chao (Inst. of Ecology and Evolutionary Biology, National Taiwan Univ., No.1, Sec. 4, Roosevelt Rd., Taipei City 106319, Taiwan, r11b44008@ntu.edu.tw), Pei-Ying Wu, and Wei-Cheng Yang (School of Veterinary Medicine, National Taiwan Univ., Taipei, Taiwan)

Anthropogenic noises in the ocean may cause significant impacts on the behavior of marine mammals, including stress, hearing impairment, and disruptions in social interactions. To investigate the potential effects of offshore wind farm pile driving noise on dolphins, we conducted controlled experiments to 3 captive bottlenose dolphins (*Tursiops truncatus*) exposed to pile driving noise playbacks at four different sound pressure levels (Mean

Lpeak 0,127,147,160 dB re 1 μPa) with a Lubell LL1424HP underwater transducer. Several vocalizational parameters of the underwater video recordings during the exposure were analyzed to assess the quantity of vocalizations and frequency modulations. Our findings revealed a notable increase in frequency modulation within whistle calls as the sound levels of pile driving noises increased. Changes in the number of calls at different noise levels can also be noticed. Additionally, we observed time-matching jaw claps to the pile driving sounds. These results suggest that pile driving activity several kilometres away from the habitat may still cause frequency masking, leading to altered behaviors of dolphins and potentially inducing population-level stress. Understanding the impact of anthropogenic noise including behavioral and acoustic responses, auditory masking, and stress on marine species is crucial for effective conservation and mitigation strategies.

1aAB9. A first look at sound transmission through the elephant middle ear—Not what we expected. Caitlin O'Connell-Rodwell (Otolaryngol., Head & Neck Surgery, Harvard Med. School, Eaton Peabody Lab/OHNS Harvard Med. School, Mass Eye & Ear, 243 Charles St., Boston, MA 02114, Caitlin_OConnell@meei.harvard.edu), Jodie Berezin (Eaton Peabody Lab, Mass Eye & Ear, Boston, MA), Mike Ravicz, and Sunil Puria (Otolaryngol., Head & Neck Surgery, Harvard Med. School, Boston, MA)

Elephants, the largest terrestrial mammals, have the lowest frequency auditory system, even lower than previously thought, which likely facilitates their long-distance communication. We used a 3D laser Doppler vibrometer to quantify ossicular motions of cadaveric middle ears of both African (*Loxodonta africana*) and Asian (*Elephas maximus*) elephants as well as humans to understand the mechanics of sound transmission in animals with larger ears than humans. Velocity in the x, y, and z directions from the umbo and stapes over the 7–13,000 Hz range were converted to their anatomical piston directions. The magnitude of umbo velocity was an order of magnitude greater in elephants than humans below the elephant middle ear resonance of about 300 Hz, whereas the stapes velocity was 5 times greater. These higher magnitude velocities are most likely due to an eardrum that is 7 times larger. Despite elephant ossicles being 10 times heavier, the magnitude of stapes velocity above 1 kHz in elephants and humans was similar, but even more surprising, elephants showed much more group delay than anticipated. We present our results in the context of eardrum size, ossicle mass, the malleus-incus lever ratio that characterize elephant and human middle ear sound transmission. [Work supported by the Amelia Peabody Fund and K01 DC017812.]

1aAB10. Spectral and temporal cues used by echolocating bottlenose dolphins to discriminate changes in inter-highlight intervals. Katie A. Christman (Psych., Univ. of California, San Diego, Muir Ln., La Jolla, CA 92093, katie.christman@nmmf.org), James Finneran (U.S. Navy Marine Mammal Program, Naval Information Warfare Ctr. Pacific, San Diego, CA), Jason Mulsow, Katelin Lally (Biologic and Bioacoustic Res., National Marine Mammal Foundation, San Diego, CA), Austin O'Kelley (Psych., Univ. of California, San Diego, San Diego, CA), Matthew Bannon (Univ. of Miami, Miami, FL), Dorian Houser (Conservation Biology, National Marine Mammal Foundation, San Deigo, CA), and Timothy Gentner (Psych., Univ. of California, San Diego, La Jolla, CA)

Echolocating dolphins use spectral cues to discriminate complex echoes from targets with multiple reflective surfaces; however, the specific cues involved are not clear. This study investigated the role of spectral interference patterns on the dolphin's ability to discriminate two-highlight echoes. In task one of the study, two dolphins were trained to echolocate, listen to returning electronic ("phantom") echoes with two identical highlights, and produce a conditioned acoustic response if the inter-highlight interval (IHI) increased. The amount that the IHI increased varied across trials. Task two replicated task one but applied a random phase shift to all highlights. This varied the specific locations of notches along the frequency axis in the complex echo spectrum but preserved the notch spacing (equal to the inverse of the IHI). For IHIs less than 250 μs, discrimination thresholds increased with IHI and were larger with random phase shifts. For IHIs greater than 250 μs, thresholds plateaued and were similar for both tasks. Results agree with those from a previous study with passively listening dolphins. Namely, echolocating dolphins can use the spacing of notches in spectral interference patterns to detect IHI changes within the ~250-μs temporal window, but temporal cues are used at longer IHIs. [Work supported by ONR]

1aAB11. Finite-element modeling on noise impacts and hearing in marine fauna. Chong Wei (Ctr. for Marine Sci. and Technol. (CMST), Curtin Univ., Ctr. for Marine Sci. & Technol. Curtin University GPO Box U1987, Perth, Western Australia 6845, Australia, chong.wei@curtin.edu.au), Christine Erbe (Australia), and Robert D. McCauley (Ctr. for Marine Sci. and Technol. (CMST), Curtin Univ., Perth, Western Australia, Australia)

Despite increasing interest in the effects of anthropogenic noise on marine fauna, relevant research is limited, particularly in those inaccessible species. Unfortunately, some of these inaccessible species are endangered, such as the little penguins (*Eudyptula minor*) and Australian sealions (*Neophoca cinerea*). In this study, we collected freshly deceased bight redfish (*Centroberyx gerrardi*), little penguins, and Australian sealion for medical computed tomography (CT)/microCT scans. Ear structures were reconstructed based on high-resolution imaging data for the three species. Moreover, 3D finite-element models were built to simulate the sound reception processes and ear responses to the incident planar waves from various directions toward the heads. The received sound pressure levels and motion (e.g., displacement and acceleration) of the internal ear related structures were computed. The data was used to predict the absolute hearing curves of the little penguins and Australian sealions. Our results also indicate that the motion difference plus the geometry of the otoliths plays an important role in the fish otoliths' response to incident sounds from different directions, resulting in different levels of damage to the hair cells. This study may potentially aid researchers in investigating the hearing of some other marine animals.

1aAB12. How the structures of the Risso's Dolphin (*Grampus griseus*) affect sound propagation and reception. Chong Wei (Ctr. for Marine Sci. and Technol. (CMST), Curtin Univ., GPO Box U1987, Perth, Western Australia 6845, Australia, chong.wei@curtin.edu.au), Christine Erbe (Ctr. for Marine Sci. and Technol. (CMST), Curtin Univ., Perth, Western Australia, Australia), Adam B. Smith (Marine Res. Ctr., Univ. of Southern Denmark, Kerteminde, Denmark), Wei-Cheng Yang (School of Veterinary Medicine, National Taiwan Univ., Taipei, Taiwan), and Leander Erbe King (Ctr. for Marine Sci. and Technol. (CMST), Curtin Univ., Perth, Western Australia, Australia)

Like other odontocetes, Risso's dolphins actively emit clicks and passively listen to the echoes during echolocation. However, Risso's dolphin head anatomy differs from other odontocetes by a unique vertical cleft along the anterior surface of the forehead and a differently shaped lower jaw. In this study, 3D finite-element sound propagation and reception models were constructed based on CT data of a deceased Risso's dolphin. Model results of sound propagation and reception were verified by finding good agreement with practical recording data from previous biosonar beam measurements and hearing sensitivity measurements. The cleft was filled with neighboring soft tissues, creating a hypothetical "cleftness" head. Comparison between sound travelling through a "cleftness" head versus an original head indicate that the distinctive cleft plays a limited role in biosonar sound propagation. Additionally, we digitally simulated the acoustic pathway for sounds to travel from the water into dolphin's tympanoperiotic complexes (TPCs). The gular reception mechanism, previously discovered in *Delphinus delphis* and *Ziphius cavirostris* was also found in this species. Sound pressure levels and displacement (motion) of the TPCs were compared between the cases with/without the mandibular fats or mandible. The results demonstrate a wave-guiding role of the mandibular fats and a bone conductor role of the mandible to transfer acoustic energy to the TPCs.

1aAB13. Investigating the impact of biomimetic pinna shape variations on clutter echoes received from natural environments. Ibrahim Eshera (Elec. and Comput. Eng., Virginia Tech, 1075 Life Sci. Cir, (Mail Code 0917), Blacksburg, VA 24061, ieshera@vt.edu), Sanmeel V. Lagad, and Rolf Müller (Mech. Eng., Virginia Tech, Blacksburg, VA)

Bat species navigating dense vegetation based on biosonar have to obtain the necessary sensory information from "clutter echoes", i.e., echoes that are superpositions of contributions from many reflecting facets (e.g., leaves) and hence have highly unpredictable waveforms. Prior results have suggested that pinna motions could aid in direction-finding tasks based on deterministic echo patterns. This raises the question whether varying pinna shapes could also have a function significance for challenging biosonar tasks performed on clutter echoes. As a first, task-independent step to test this hypothesis it has been investigated whether different pinna shapes have a consistent effect on clutter echoes despite the random nature of these signals. This was accomplished using a dedicated laboratory setup that produced large amounts of uncorrelated clutter echo data by agitating an artificial foliage with fans between echoes. Deep learning methods were then applied to identify the pinna shape that received a given clutter echo. A two-dimensional convolutional neural network operating on spectrograms achieved 90% validation accuracy in this task. This finding demonstrates that even small pinna deformations can impart consistent effects on the clutter echoes. Ongoing research is directed at analyzing the nature of the signal properties that the successful classifications were based on.

1aAB14. Ultrasonic hearing abilities of the domestic cat assessed with auditory brainstem responses. M Charlotte Kruger (Physiol., McGill Univ., 3655 Promenade Sir William Osler, Rm. 1246A, Montreal, QC H3G 1Y6, Canada, charlotte.kruger@mail.mcgill.ca) and Stephen G. Lomber (Physiol., McGill Univ., Montreal, QC, Canada)

Domestic cats (*Felis catus*) have sharp sensory abilities which they use in various circumstances. For example, they have acute hearing for detecting and localizing relevant auditory information, such as the presence of potential threats or prey items. However, there are notable discrepancies in the literature regarding the full extent of the cat's hearing abilities. Here, we hypothesize that domestic cats can hear ultrasonic frequencies above 60kHz, since they might utilize their hearing abilities to detect the ultrasonic vocalizations emitted by rodent prey. We used auditory brainstem responses (ABRs), a more time efficient method compared to behavioral psychoacoustic techniques, to evaluate the sensitivity of the cat's auditory system to ultrasonic frequencies. We presented artificial and behaviorally relevant stimuli containing ultrasonic frequencies to each cat ($n = 6$). We then recorded the resulting ABRs, measured the wave amplitudes and latencies, and determined the ABR thresholds to these stimuli. The ABR data presented here will be useful in conjunction with psychoacoustic experiments to provide insight into the neural mechanisms that might be involved when cats perceive high frequency signals. This work will ultimately contribute to a better understanding of the cat's hearing abilities. [Work supported by the Natural Sciences and Engineering Research Council of Canada.]

1aAB15. Influence of ocean dynamics on Cuvier's beaked whale *(Ziphius cavirostris)*: Presence and their prey. Shannon M. Dolan (Scripps Inst. of Oceanogr., Univ. of California, San Diego, 8622 Kennel Way, La Jolla, CA 92037, shdolan@ucsd.edu), Simone Baumman-Pickering, Ashlyn Giddings, Eric R. Snyder (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA), David A. Demer (NOAA Fisheries-Southwest Fisheries Sci. Ctr., La Jolla, CA), Jennifer S. Trickey, and Peter JS Franks (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA)

Cuvier's beaked whales *(Ziphius cavirostris; Zc)* dive below 1000 m to forage predominantly on deep-sea squids. The impact of oceanographic conditions on Zc prey and the effects of anthropogenic sounds on Zc foraging remain unclear. Zc prey distribution is patchy and potentially influenced by mesoscale oceanographic features interacting with local bathymetry. Favorable environmental conditions lead to aggregations of lower trophic level species that attract squid, and potentially increase Zc presence. We collected simultaneous active and passive acoustic data in Tanner Basin, a southern California submarine canyon, intermittently between 2017 and 2022. Passive acoustic data revealed Zc were present throughout all years, with lower presence during late summer and fall. Subsurface Zc 3D track reconstructions indicated a preference for one side of the canyon, where Zc foraged within a layer of potential prey, detected from active acoustic data. Prey concentrations, quantified as volume backscatter strength below ~1150 m depth, exhibited an increase following the passage of mesoscale features, which were measured using Finite-size Lyapunov Exponents (FSLE). These prey increases drove immediate increases in Zc presence. Understanding potential physical drivers of Zc predator-prey dynamics is a critical first step for assessing natural variability and potential impacts of sonar use on Zc foraging behavior.

1aAB16. Potential passive acoustic tool for reef fish monitoring: A learning-based method. Aléxia A. Lessa (Ecology, Universidade Federal do Rio de Janeiro, Rua Alfeu Ferreira - Travessa F, 508, Iguaba Grande, Rio de Janeiro 28960000, Brazil, alexiaalc@id.uff.br), Viviane Barroso (Marine Biotechnology, Instituto de Estudos do Mar Almirante Paulo Moreira, Arraial do Cabo, Brazil), Carlos Ferreira (Marine Biology, Universidade Federal Fluminense, Niterói, Brazil), and Fábio Xavier (Marine Biotechnology, Instituto de Estudos do Mar Almirante Paulo Moreira, Arraial do Cabo, Brazil)

Visual non destructive methods are widely used for assessing marine biodiversity, but a logistically simpler approach is combining passive acoustic techniques with artificial intelligence (AI). Here, we classified sounds of fish on subtropical rocky reefs in Arraial do Cabo (22°57'S, 41°01'W), RJ, Brazil, using supervised machine learning algorithms. Sound detection and feature extraction were performed manually by inspecting spectrograms using Raven Pro 1.6 software. Five supervised algorithms were implemented for classification: naive Bayes, support vector machine, random forest, decision tree, and multilayer perceptron. A representative subset of samples for each sound class was used to train the supervised algorithms. The accuracy of the algorithms was between 67% and 97%. Four classes of sounds were recognized, consisting of sequences of pulses. In general, temporal features were more important, however high frequency was the most important feature of all. Understanding the contribution of each feature is crucial for sound classification, however it is at an early stage for fishes. The classifier showed promising results, highlighting the effectiveness of applying AI to passive acoustics as a tool for monitoring of fish assemblages.

1aAB17. Where are the whales? Continued acoustic monitoring efforts of California's Central Coast. Adelle Wilkin (Statistics, California Polytechnic State Univ. (Cal Poly San Luis Obispo), 1 Grande Ave., San Luis Obispo, CA 93410-1722, awilki07@calpoly.edu), Sophie M. Short (Statistics, California Polytechnic State Univ. (Cal Poly San Luis Obispo), San Luis Obispo, CA), Isaiah Orlando, Lucy Nosbisch (California Polytechnic State Univ. (Cal Poly) San Luis Obispo, San Luis Obispo, CA), and Maddie Schroth-Glanz (Statistics, California Polytechnic State Univ. (Cal Poly San Luis Obispo), San Luis Obispo, CA)

Understanding and conserving ocean ecosystems is a difficult, but important task made easier with bioacoustic technology. Although marine mammal monitoring efforts can be done with visual surveys, they often lack temporal information on a finer scale. Long-term passive acoustic monitoring efforts of critical migratory species can provide detailed information about the region's marine soundscape that is key to conservation efforts. The aim of our project is to establish a long-term bioacoustic monitoring station off the coast of central California and help close the gap in our understanding of marine mammal presence along the Pacific coastline. The goals put forth by the bioacoustic lab for this academic year include (1) determining probable species presence, (2) ascertaining temporal (day/night) variations of vocalizations, and (3) examining key differences in species and call type between two deployments from Fall 2022 and Winter 2023. These comparisons will be crucial to help develop a baseline understanding of seasonal and temporal variations of the soundscape in our region. We aim to implement various long-term deployments in order to gather more complete and consistent bioacoustic data in our local waters. These deployments and the subsequent analysis will contribute to large scale marine mammal monitoring and conservation efforts. [This project was

made possible by a grant from the Santa Rosa Creek Foundation and the COAST undergraduate research fund.]

1aAB18. Who's knocking? Gray whale acoustic detection along California's central coast. Sophie M. Short (Statistics, California Polytechnic State Univ. (Cal Poly) San Luis Obispo, Grand Ave. San Luis Obispo, CA 93407, smshort@calpoly.edu), Adelle Wilkin, Isaiah Orlando, Lucy Nosbisch, and Maddie Schroth-Glanz (Statistics, California Polytechnic State Univ. (Cal Poly) San Luis Obispo, San Luis Obispo, CA)

Passive acoustic monitoring (PAM) efforts are essential to marine mammal conservation. Bioacoustic techniques can be preferred as visual surveys are time intensive, costly, and limiting. California's coastline is a biodiversity hotspot and a known migration corridor of multiple cetacean species, including gray whales, according to stock assessments provided by NOAA. Certain studies have indicated that gray whales vocalize during migration which can help confirm cetacean presence throughout the year. Off the coast of San Luis Obispo, California in an area that overlaps with the gray whale's southern migration, our team detected an unidentified knocking noise. Although gray whale vocalizations can be difficult to confirm, we believe these knocks are produced by this transient species. In order to determine the source of these knocking sounds, the goals of this project are to (1) classify the repetitive noise as biotic or abiotic, (2) perform a comparative analysis of various species to narrow down possible biotic candidates, (3) utilize records from local whale watching companies for more accurate evidence of species presence at the time of our deployment, and (4) examine temporal variation in the occurrence of this knocking sound. Acoustic monitoring provides information about migration of gray whales necessary for conservation efforts.

1aAB19. Distribution and seasonality of the Omura's whale (*Balaenoptera omurai*) in Australia based on passive acoustic recordings. Ciara E. Browne (Ctr. for Marine Sci. and Technol., Curtin Univ., 16 Baal St., Palmyra, Western Australia 6157, Australia, ciara.browne@postgrad.curtin.edu.au), Christine Erbe, and Robert McCauley (Ctr. for Marine Sci. and Technol., Curtin Univ., Perth, Western Australia, Australia)

The Omura's whale is one of the most recently described species of baleen whale. Initially known only from stranding and whaling specimens, it has now been identified in all ocean basins excluding the central and eastern Pacific. Unlike most baleen whales that migrate between the poles and the equator seasonally, the Omura's whale is known to inhabit tropical to sub-tropical waters year-round. In Australian waters, there remain fewer than 30 confirmed visual sightings over the past decade. This study utilizes passive acoustic recordings from 41 locations around Australia from 2005 to 2023 to assess the distribution and seasonality of the Omura's whale. The seasonal presence of Omura's whale vocalizations varied by location, with higher presence at lower latitudes. Vocalizations were detected year-round in the Timor Sea and the Kimberley region. The most southerly occurrence of Omura's whale vocalizations was recorded off the North West Cape, WA. In Australia, the vocalizations detected differ from those of other Indian Ocean Omura's whale populations. Geographic variation of the vocalization was also observed within Australia, including vocalizations similar but not identical to those of the Omura's whale detected in the Great Barrier Reef for the first time. The identified seasonal distribution and possible acoustic populations of the species in Australia provides valuable information to assess environmental and anthropogenic pressures on the Omura's whale and to aid in creating management policies for the species.

1aAB20. Using a deep neural network to classify echolocation clicks and identify biogeographic patterns of Pacific white-sided dolphins. Michaela Alksne (Scripps Inst. of Oceanogr., 115 Reno Way, Santa Cruz, CA 95060, m1alksne@ucsd.edu), Annebelle KoK, Annika Agarwal, Kaitlin Frasier (Scripps Inst. of Oceanogr., San Diego, CA), and Simone Baumman-Pickering (Scripps Inst. of Oceanogr., La Jolla, CA)

Pacific white-sided dolphins are endemic to the Pacific Ocean; and two genetically distinct populations overlap along the west coast of North America. However, they are visually indistinguishable and the degree of spatial overlap remains unknown. Here, we use a deep neural network to show that the populations are acoustically distinguishable. Previous studies described two distinct echolocation click types associated with pacific white-sided dolphins and hypothesized that they were population-specific. Our neural network was trained to classify the click types based on spectral and temporal properties described previously. The neural network enabled us to analyze passive acoustic recordings from sites between the Gulf of California and the Gulf of Alaska over multiple years to investigate possible population-specific trends. The latitudinal occurrence pattern of the two click types supports the population-specific hypothesis: type A clicks continue to associate with the northern population distribution, and type B clicks with the southern population distribution. At long-term monitoring sites in Southern California, type B clicks were dominant during periods of warm water anomalies. This pattern may be an early indicator of future biogeographic shifts in the distribution of pacific white-sided dolphins and demonstrates the utility of long-term passive acoustic monitoring.

1aAB21. Acoustic enrichment trials using autonomous cameras on a Hawaiian coral reef. Océane Boulais (Scripps Inst. of Oceanogr., Univ. of California San Diego, 1043 Chalcedony St., San Diego, CA 92109, oceane@ucsd.edu), Daniel Schar (Hawai'i Inst. of Marine Biology, Univ. of Hawai'i, Honolulu, HI), Josh Levy (Appl. Res. Lab., Univ. of Hawai'i, Honolulu, HI), Katherine Kim (Greeneridge Sci., Inc, San Diego, CA), Natalie Levy (Dept. of Nanoengineering, Univ. of California San Diego, San Diego, CA), Jessica Reichert, Nina Schiettekatte (Hawai'i Inst. of Marine Biology, Univ. of Hawai'i, Honolulu, HI), Daniel Wangpraseurt (Dept. of Nanoeng. & Scripps Inst. of Oceanogr., Univ. of California San Diego, San Diego, CA), Josh Madin (Hawai'i Inst. of Marine Biology, Univ. of Hawai'i, Honolulu, HI), and Aaron M. Thode (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA)

Playbacks of ambient sound from healthy shallow coral reefs, known as acoustic enrichment (AE), can attract fish larvae and potentially coral larvae to degraded or artificial reefs. Large-scale studies face power supply challenges and require less intensive in-person surveys of recruited larvae ($\leq$500 μm) and fish. Here, we present a preliminary assessment of artificial structure deployments with AE, conducted between July and September 2023 in Kane'ohe Bay (Oahu, Hawai'i). The artificial structures were designed to attract herbivorous juvenile reef fish and subsequently suppress algal growth. The work is part of a future artificial reef construction project in Hawai'i where in-person surveys and frequent battery replacements are logistically challenging. The deployments consisted of one control and one treatment site 65 m apart, in 5 m water depth and 10 m from the natural reef. To document ecosystem responses, both sites use open-source burst-mode autonomous cameras, a directional vector sensor seafloor acoustic recorder, and hydrophones. A 3.8 kW-h NiMh battery supply powers sound output for three weeks during evening hours. We discuss preliminary evaluations of AE and the validity of using autonomous cameras in evaluating performance in terms of fish recruitment and algal growth. [Work sponsored by DARPA.]

1aAB22. Deep learning for large scale conservation bioacoustics—A demonstration on the Malabar whistling thrush and the dhole. Namitha Suresh (Phys., Cornell Univ., 617 Space Sci. Bldg., Ithaca, NY 14850, ns873@cornell.edu), Benjamin Thomas, Sha Wang (K. Lisa Yang Ctr. for Conservation Bioacoustics, Cornell Lab of Ornithology, Ithaca, NY), Shyam Madhusudhana (Ctr. for Marine Sci. and Technol., Curtin Univ., Perth, Western Australia, Australia), Vijay Ramesh, and Holger Klinck (K. Lisa Yang Ctr. for Conservation Bioacoustics, Cornell Lab of Ornithology, Ithaca, NY)

Over the past several years, deep learning-based algorithms have become the go-to solution in conservation bioacoustics for detecting and classifying animal calls in soundscape recordings. While these algorithms often perform very well, they frequently require specialized computer setups and expertise for training and running, thereby hindering the scalability of conservation bioacoustics solutions. Here we evaluate a new framework to overcome this challenge. We train a ResNet model (ResNet50V2 pre-trained with the ImageNet dataset) using the open-source Python package Koogu (https://github.com/shyamblast/Koogu). We focus on two species of

interest: the Malabar whistling thrush (Myophonus horsfieldii), a resident bird of Peninsular India, and the dhole (Cuon alpinus), an endangered canid species native to Asia. Preliminary results for the whistling thrush showed that the model achieved a precision of 0.95 and recall of 0.71 on the test data for a detection threshold of 0.7. The work on the dholes is still ongoing. The trained models are then incorporated into the Raven Pro 1.6.5 sound analysis software (https://ravensoundsoftware.com/), equipped with a Deep Learning Detector functionality, which allows Deep Learning models to be run on acoustic data. The models integrated into the software are made publicly available to other researchers and practitioners enabling them to run the models over their datasets as well as integrate their own custom models into Raven Pro.

1aAB23. Using the soundscape to monitor vessel visitation at an Australian fur seal (*Arctocephalus pusillus doriferus*) breeding colony. Jessalyn Taylor (Sydney School of Veterinary Sci., The Univ. of Sydney, Regimental Dr., Camperdown, New South Wales 2050, Australia, jtay0903@uni.sydney.edu.au), Rebecca McIntosh (Phillip Island Nature Parks, Cowes, Victoria, Australia), Chloé Huetz (Institut des Neurosciences Paris-Saclay, Université Paris-Saclay, CNRS, Saclay, France), Rachael Gray (Sydney School of Veterinary Sci., The Univ. of Sydney, Camperdown, New South Wales, Australia), and Isabelle Charrier (Institut des Neurosciences Paris-Saclay, Université Paris-Saclay, CNRS, Saclay, France)

Marine vessel traffic is a significant source of anthropogenic noise pollution and a major source of disturbance for many marine species. Despite Seal Rocks (Victoria, Australia) being the largest Australian fur seal breeding colony, and subject to a high volume of visitation, particularly during the breeding season, there has been no long-term effort to monitor visitation and its potential impact on seals. Autonomous passive acoustic recorders were deployed at Seal Rocks to monitor vessel visitation during both low visitation (off-peak) and the breeding (peak) season. Proportion of time vessels present in the area was calculated from visual inspection of long-term spectral averages (LTSA). Soundscape metrics were calculated to determine broadband sound pressure level (BL) and percentiles of power spectral density (PSD). Visitation was higher in the peak season, as was BL, particularly during daylight hours. While there were periods of high winds and heavy swell during both deployment seasons, vessel noise was still a regular and loud contributor to the soundscape. From these results, the airborne noise received by seals can also be estimated. These results will contribute towards the development of an impact model to assess noise impacts at Seal Rocks and other sensitive marine sites.

1aAB24. Sub-unit classification of Pacific and Atlantic humpback high-complexity song units—Sub-unit diversity drives unit diversity. Howard S. Pines (The Whale Song Translation Project, Working From Home, El Cerrito, CA 94530, howardpines@ieee.org)

Humpback whale communication studies have primarily focused on holistic unit classification methods to identify thematic patterns of song structure. These various whole-unit, spectral-feature-extraction and classification procedures have resulted in low information entropy estimates of songs, akin to bird song. A recent study of high complexity, frequency-modulated song units shows that unit diversity and information entropy of humpback vocalizations is much higher and variable across regions of the Pacific and the Atlantic. Sub-unit-level feature-extraction and classification of these high-complexity units indicates that humpbacks have precision control of the generation of target frequencies, a characteristic of human vowel production. The number of distinct sub-units varies by geographic region and is comparable to English and Asian language phoneme sets.

Session 1aBA

Biomedical Acoustics: General Topics in Biomedical Acoustics I: Imaging

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Contributed Papers

10:20

1aBA1. Optimizing transducer aperture subdivision for aberration correction during transabdominal histotripsy. Ellen Yeats (Univ. of Michigan, 2293 Stone Rd., Ann Arbor, MI 48105, yeatsem@umich.edu), Hadrien Padilla, Jonathan Sukovich, and Timothy L. Hall (Univ. of Michigan, Ann Arbor, MI)

This study investigated the effect of transducer aperture subdivision on aberration correction performance for transabdominal histotripsy. A transducer aperture and geometric curvature similar to clinical histotripsy devices (750 kHz, 20 × 16 cm width, 14 cm radius of curvature) was subdivided into 64, 256, and 1024 elements (10.7λ, 5.3λ, 2.6λ). For each of the three designs, optimal space filling (100%) equal area polygons with a pseudo-random arrangement of elements was used following the technique described by Rosnitskiy. 3D acoustic path maps were constructed from abdominal CTs of 3 de-identified human subjects and assigned acoustic properties from literature. For each transducer, acoustic transmission was simulated using k-Wave at 3 target locations in the liver of each subject with and without implementing ideal aberration correction. Across all subjects and targets ($n = 9$), AC increased the maximum focal pressure amplitude by 26% (0%-59%) on average for 64 elements, 38% (5%-84%) for 256 elements, and 42% (7%–92%) for 1024 elements. These results suggest 5.3λ elements may near an optimal subdivision for transabdominal histotripsy aberration correction when taking into account the cost of transducer and driving system fabrication. Acoustic simulations can save fabrication time and costs when optimizing device design.

10:40

1aBA2. Verification of relationship between transmission/reception conditions of high-frequency plane wave compounding and evaluation accuracy of extended amplitude envelope statistics. Taisei Higa (Graduate School of Sci. and Eng., Chiba Univ., 1-33 Yayoi-cho Inage-ku, Ctr. for Frontier Medical Eng., Chiba, Chiba 263-8522, Japan, thiga@chiba-u.jp), Jeffrey A. Ketterling, Jonathan Mamou, Cameron Hoerig, Daniel H. Gross (Radiology, Weill Cornell Medicine, New York, NY), Shinnosuke Hirata, Kenji Yoshida, and Tadashi Yamaguchi (Ctr. for Frontier Medical Eng., Chiba Univ., Chiba, Japan)

Amplitude envelopment statistics is one of the effective analysis method for quantitative ultrasound. In the studies using a single transducer of around 15–25 MHz and the Double-Nakagami (DN) model, it was possible to evaluate fat mass in rat livers. In order to realize clinical applications, it is essential to study with array probes, especially to understand the influence of the sound field characteristics on the amplitude envelope statistics. In this study, fatty liver simulated phantoms which were containing two types of scatterers was observed with five types of high-frequency (15–40 MHz) linear array probes, and the amplitude enveloping characteristics were analyzed using the DN model to evaluate the fat constitution. As a result, it was possible to evaluate fatty components with high accuracy under conditions in which fat is distributed in low density, such as in mild fatty liver. No significant differences in evaluation systems were observed among the probes, probably due to the effect of parallel plane wave transmission and reception.

11:00

1aBA3. Investigation of spatio-temporal inertial cavitation activity for optimization of needle-free ultrasound-enhanced vaccine delivery. Darcy Dunn-Lawless (Inst. of Biomedical Eng., Univ. of Oxford, Botnar Inst. for Musculoskeletal Sci., Windmill Rd., Oxford OX37LD, United Kingdom, darcy.dunn-lawless@magd.ox.ac.uk), Joel Balkaran, Brian Lyons, Robert Carlisle, Constantin Coussios, and Michael Gray (Inst. of Biomedical Eng., Univ. of Oxford, Oxford, United Kingdom)

Ultrasound-induced cavitation is a promising mechanism for pain-free delivery of vaccines without needles. However, the relationships between cavitation energy and the various bioeffects involved in transdermal vaccination, including skin permeabilization, convective drug transport, reversible and irreversible sonoporation, and immune-stimulation, are not well understood. Previous transdermal ultrasound experiments have demonstrated inhomogeneous delivery across the exposed surface, which remains poorly understood. In this work, we first seek to fully characterize and quantify spatio-temporal cavitation activity across the skin surface, to identify a local cavitation dose that is optimal for all four key bioeffects. We have designed a new *in vitro* experimental setup that exposes several potential skin models to 265 kHz focused ultrasound, a new generation of protein-based cavitation nuclei (pCaNs), and a fluorescently labelled vaccine analogue, while simultaneously imaging cavitation activity parallel to the skin surface by Passive Acoustic Mapping (PAM). Subsequent staining and multi-photon microscopy of the skin models allows a direct comparison between the PAM-derived spatiotemporal inertial cavitation dose and locations where particular bioeffects were maximized. We will discuss the results of this investigation, how they relate to our recent *in vivo* study, and whether these findings enable enhancements of the spatial homogeneity and reproducibility of needle-free ultrasound vaccination.

11:20

1aBA4. Investigation of the effect of a multi-frequency ultrasound signal on the inertial cavitation threshold is various viscoelastic mediums. Tatiana Filonets (National Taiwan Univ., National Health Res. Institutes, Taipei, Taiwan) and Maxim Solovchuk (Inst. of Biomedical Eng. and nanomedicine, National Health Res. Institutes, No. 35, Keyan Rd., Zhunan, Miaoli County 35053, Taiwan, solovchuk@gmail.com)

The objective of the present study is the numerical investigation of the inertial cavitation threshold in soft tissue under different multi-frequency signals of high intensity focused ultrasound (HIFU). For the modeling of bubble dynamics, the Gilmore-Akulichev-Zener model was used. The inertial cavitation threshold was calculated for different criteria, for different signal modes (single-, dual-, and triple-frequency), and for different viscoelastic mediums. Effect of nonlinear wave propagation and mechanical effects of acoustic caviation have been also studied. The obtaining results demonstrated that a criterion, based on bubble size, gives lower threshold values than a criterion using bubble collapse velocity. An increase in the values of viscosity and elasticity leads to a rise in the threshold amplitude. Analyzing threshold values for dual- and triple-frequency signal modes, it can be seen that the inertial cavitation threshold can be sufficiently reduced when a multi-frequency signal with the correct frequency combination is

applied. However, the inertial cavitation threshold also depends on the initial bubble size. This dependency was also investigated in the current study.

11:40

1aBA5. Thermal control of engineered induced pluripotent stem cells via focused ultrasound. Kama Bell (Acoust., Penn State, 201 Appl. Sci. Bldg., University Park, PA 16802, krb5915@psu.edu), Alessandro Howells, Lance Lian (Bioengineering, Penn State, University Park, PA), and Jing Yun (Acoust., Penn State, University Park, PA)

Sonogenetics has drawn significant interest from the neuromodulation community since its inception, due to its non-invasive and targeted modulation of neurons. Simultaneously, stem cell therapy presents a promising avenue for addressing neurodegenerative disorders like Parkinson's disease. Central to the field of stem cell therapy is cell fate engineering, which leverages induced pluripotent stem cells (iPSCs) to generate therapeutically relevant cell types capable of combating neurodegeneration. Although the inducible overexpression of transcription factors (TFs) through drug administration can convert neural progenitors into mature neurons, its application in the brain is hindered by systemic side effects and the challenges posed by the blood-brain barrier, impeding efficient drug delivery. In this context, we present our recent *in vitro* work on a sonogenetics-enabled system that allows for spatial and temporal control of TFs, enabling the direct programming of human stem cells. By incorporating heat shock promotors, we have developed a method to thermally regulate engineered iPSCs using focused ultrasound, thereby directing their differentiation into dopaminergic neurons. Our findings suggest that this approach holds promise as a potential therapeutic strategy for treating Parkinson's disease.

Session 1aED

Education in Acoustics and Musical Acoustics: Connections Between Music and Math

Gordon P. Ramsey, Cochair
Physics Dept., Loyola University Chicago, Chicago, IL 60660

Andrea Calilhanna, Cochair
Faculty of Arts, Elder Conservatorium of Music, University of Adelaide, North Terrace, Adelaide, 5005, Australia

Invited Papers

11:00

1aED1. Teaching the connections between music and mathematics. Gordon P. Ramsey (Phys., Loyola Univ. Chicago, Chicago, IL 60660, gramsey@luc.edu)

Mathematics is often perceived by students as being difficult. However, when they see how math and music are intimately related, their attitude often changes. Some of the most fundamental aspects of music are described by math. For example, the Circle of Fifths closely reflects musical structure and is used to create scales and chord progressions. The Circle is structured around the mathematical relations of frequency ratios of the corresponding notes in a 12-tone scale. The reactions to music of consonance and dissonance are related to the relative ratios of the notes in a chord or note sequence in a piece. The connection between music and math forms the basis for creating instruments that play the music. I will suggest ways to present these concepts to students for their greater appreciation of music and math.

11:20

1aED2. The ripples of consonant intervals and chords arising in different tuning systems. Luis Nuño (Communications, Polytechnic Univ. of Valencia, ETSI Telecomunicación, UPV, Camino de Vera, S/N, Valencia, Valencia 46022, Spain, harmonicwheel@gmail.com)

Pure consonances are combinations of exact multiples of a fundamental frequency, particularly with the factors 2, 3, and 5. They give rise to the consonant intervals (octave, perfect fifth, and major and minor thirds) and consonant chords (major and minor). However, these kinds of consonances are only achieved in just intonation, which does not produce beats or ripples, that is, their envelopes are constant. On the contrary, other tuning systems, such as Pythagorean, meantone, or N-tone equal temperaments, which slightly differ from just intonation, actually produce beats or ripples, that is, the corresponding envelopes oscillate and, acoustically, resemble a vibrato in amplitude. In this paper, the envelopes of consonant intervals and chords are modelled mathematically by approximate formulas. They contain one variable for the intervals and two variables for the chords. The formulas include deviations of one full trigonometric period, thus being valid for any tuning system. In the case of intervals, the corresponding ripples are obtained by following a line (1D) at a specific velocity and, in the case of chords, by following a line on a graph (2D) at specific velocity and direction. The corresponding audio files considering pure tones are also obtained.

Contributed Paper

11:40

1aED3. Ski-hill Graph Pedagogy meter fundamentals acoustics education. Andrea Calilhanna (Faculty of Arts, Business, Law, and Economics; Elder Conservatorium of Music, Univ. of Adelaide, The University of Adelaide, Elder Conservatorium of Music, North Terrace, Adelaide, South Australia 5005, Australia, a.calilhanna@gmail.com)

Acoustics education in school music classrooms is generally approached as a study of sound as pitch frequencies, spectral analysis, and other elements of music such as rhythm, volume, and timbre. While the psychoacoustic aspects of these acoustic properties continue to inspire and educate, the musical meter is yet to be introduced as an ordinary aspect of acoustics discussions. The meter is taught as notation, time signatures and a beat grouped in a measure, even though the listener may experience a disconnect between what is experienced and what is taught and represented as meter. Combined, the meter's psychoacoustic role during music experiences is less critical or misunderstood due to the previous lack of meter theory than tonality. Meter theorists and music educators are responding to the call for solutions (Cohn, 2020). To illustrate how meter can be understood and taught through listening to music, as psychoacoustical and mathematical from the earliest lessons, the paper provides a solution to introduce contemporary meter theory (Cohn, 2020) to beginners based on listening experiences. The report discusses the ski-hill graph as applied to meter fundamentals and provides inclusive music education; cross-cultural, multi-sensory applications demonstrate plausible benefits through an interdisciplinary approach to the meter.

Session 1aID

Interdisciplinary: Keynote: Exploring the Ocean with Sound: Telltale Acoustic Signatures of a Changing Ocean

James H. Miller, Chair
Ocean Engineering, University of Rhode Island, URI Bay Campus, Sheets Lab 111, Narragansett, RI 02882

Chair's Introduction—9:00

Invited Paper

9:05

1aID1. Exploring the ocean with sound: Telltale acoustic signatures of a changing ocean. Andone C. Lavery (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., 98 Water St., Woods Hole, MA 02540, alavery@whoi.edu)

Our ocean plays a major role in climate regulation, hosts vast biodiversity, sustains economies, ensures food security, and supports international shipping, tourism, recreation, and human health and security. Yet, the ocean is experiencing unprecedented changes: sea level rise, coral bleaching, ocean acidification, anoxia, accelerated melting of the major ice sheets, shifts in ocean circulation, pollution, overfishing, and increased industrialization. Sound plays a vital role in exploring, understanding, and monitoring these changes over a range of spatial and temporal scales. In this talk, I will focus on the multitude of approaches taken to using sound to understand our changing ocean, the advances in platforms used to enable these studies, the expansion of global ocean observing systems that include measurements of sound as an essential ocean variable, and the enormous complexities in interpreting acoustic signals in a dynamic and evolving ocean landscape. I will end by discussing the opportunities and obstacles associated with globalizing approaches for using sound to improve ocean stewardship.

Session 1aPP

Psychological and Physiological Acoustics: General Topics in Psychological and Physiological Acoustics (Poster Session)

Yi Shen, Chair
Speech and Hearing Sciences, University of Washington, 1417 NE 42nd Street, Seattle, WA 98105-6246

All posters will be on display from 10:20 a.m. to 12:20 p.m. Authors of odd-numbered abstracts will be at their posters from 10:20 a.m. to 11:20 p.m. and authors of even-numbered abstracts will be at their posters from 11:20 a.m. to 12:20 p.m.

Contributed Papers

1aPP1. Comparison of independent, participant-led, high-resolution studies of pitch-matching in post-lingual single-sided cochlear implant recipients. David Lovell (School of Comput. Sci., Ctr. for Data Sci., Queensland Univ. of Technol., GPO Box 2434, Brisbane, Queensland 4001, Australia, David.Lovell@qut.edu.au), William Martens (School of Eng., Macquarie Univ., Sydney, New South Wales, Australia), Jake Bradford (School of Comput. Sci., Ctr. for Data Sci., Queensland Univ. of Technol., Brisbane, Queensland, Australia), Andrew Bradley (School of Sci., Technol. and Eng., Univ. of the Sunshine Coast, Sippy Downs, Queensland, Australia), Anne-Marie Crowe (Hear Life Audiol., Canberra, Australian Capital Territory, Australia), Matthew McKague, and Dimitri Perrin (School of Comput. Sci., Ctr. for Data Sci., Queensland Univ. of Technol., Brisbane, Queensland, Australia)

Understanding "what the world sounds like to someone else" is fundamental to developing technologies and therapies to improve auditory perception for people with hearing impairment. Post-lingual single-sided deaf (SSD) cochlear implant (CI) recipients can play a special role here through their ability to compare perceptions of sounds heard acoustically and "electrically." Pitch is, ostensibly, a sonic attribute that can be quantitatively compared, and pitch perception has been studied by asking SSD CI recipients to rank or discriminate limited sets of pitches, mindful of the burden on experimental participants. Through unusual circumstances, we have independently conceived and conducted two sets of participant-driven pitch-matching experiments on authors WM and DL gaining more data and richer insights than reported elsewhere. While our findings are consistent with existing lower-resolution studies (e.g., sounds perceived via CI are at higher pitch than via normal hearing), we have gained much more detailed quantitative and qualitative descriptions of those perceptual differences. Results can be meaningfully compared after accounting for individual differences and suggest that the CI transformation of pitch information for these two recipients could be individually adjusted to improve music appreciation, particularly with respect to restoring perception of musical intervals between successive notes.

1aPP2. Deep neural network-based speech enhancement for cochlear implants. Tim J. Brochier (Res. and Development, Cochlear, 119 Fyffe St., Thornbury, Victoria 3071, Australia, tjbrochier@gmail.com), Amanda Fullerton, Adam Hersbach (Res. and Development, Cochlear, Melbourne, Victoria, Australia), Harish Krishnamoorthi (Res. and Development, Cochlear, Denver, CO), and Zachary Smith (Res. and Development, Cochlear, Sydney, New South Wales, Australia)

Noisy conditions make understanding speech with a cochlear implant (CI) difficult. Speech enhancement (SE) algorithms based on signal statistics can be beneficial in stationary noise, but rarely provide benefit in modulated multi-talker babble. Current approaches using deep neural networks (DNNs) rely on a data driven approach for training and promise improvements in a wide variety of noisy conditions. In this study a DNN-based SE algorithm was evaluated in CI listeners. The network was trained on a large database of publicly available recordings. A double-blinded acute evaluation was conducted with 10 adult CI users by assessing intelligibility and quality of speech embedded in a range of different noise types. The DNN-based SE algorithm provided significant benefits in speech intelligibility and sound quality in all noise types that were evaluated. Speech reception thresholds, the SNR required to understand 50% of the speech material, improved by 1.8 to 3.5 dB depending on noise type. Benefits varied with the SNR of the input signal and the mixing ratio parameter that was used to combine the original and de-noised signals. The results demonstrate that DNN-based SE can provide benefits in natural, modulated noise conditions, which is critical to CI users in their day-to-day environment.

1aPP3. Peripheral neural synchrony in pediatric cochlear implant users. Shuman He (Otolaryngol. - Head and Neck Surgery, The Ohio State Univ., 915 Olentangy River Rd., Columbus, OH 43212, shuman.he@osumc.edu), Jeffrey Skidmore, and Yi Yuan (Otolaryngol. - Head and Neck Surgery, The Ohio State Univ., Columbus, OH)

We recently developed a noninvasive method to quantify neural synchrony in the electrically stimulated cochlear nerve (i.e., peripheral neural synchrony) using an index named the phase locking value (PLV). The PLV is a measurement of trial-to-trial phase coherence in the summated activity of cochlear nerve fibers. Larger PLVs indicate better/stronger peripheral neural synchrony. This tool allows for investigating this important phenomenon in cochlear implant (CI) users for the first time in the literature. The aim of this study was to characterize peripheral neural synchrony in a large group of pediatric CI users with various etiologies. To date, study participants included 17 children with cochlear nerve deficiency, 28 children with Gap Junction Beta-2 mutation, nine children with normal inner ear anatomy and idiopathic hearing loss, three children with enlarged vestibular aqueducts, three children with Mondini malformation and two children with Usher syndrome. All participants use Cochlear™ Nucleus® devices in their test ears. For each implanted ear tested in each participant, peripheral neural synchrony was assessed at the maximum comfortable level and multiple electrode locations across the electrode array. Our preliminary results demonstrated substantial variations in peripheral neural synchrony among pediatric CI users. These variations appear to depend on hearing loss etiologies.

1aPP4. Utility of electrocochleography for cochlear implant recipients and its potential to predict post-operative hearing thresholds objectively. Lei Xu (Shandong Provincial ENT Hospital, No. 4 West Duanxing Rd., Jinan, Shandong 250022, China, sdphxl@126.com), Ruijie Wang, Jianfen Luo, Xiuhua Chao, and Haibo Wang (Shandong Provincial ENT Hospital, Ji'nan, China)

The relationship between postoperative hearing threshold and CM measured using AIM system developed by Advanced Bionics was investigated in this study. Seven participants (eight ears) with hearing thresholds better than 85 dB HL at 500 Hz were enrolled so far. All were unilaterally or bilaterally implanted with Advanced Bionics HiFocus Mid-Scala or SlimJ electrode arrays. Electrocochleography (ECochG) amplitudes were recorded both intraoperatively and postoperatively at different follow-up sessions. Objective hearing thresholds predicted by the AIM system were compared to the participants' behavioural hearing thresholds. The trace of CM amplitude showed patterns of either rising or steady responses, except decrease in one case (response amplitude dropped for 12 μV). For the eight participants, the mean difference between the objective hearing levels (ECochG thresholds) measured immediately post-operatively and the preoperative behavioural threshold across 125 Hz, 250 Hz, 500 Hz, and 1000 Hz is 4.90 dB, while its difference to postoperative behavioural threshold is 3 dB. AIM can assist the surgeon to optimize electrode array insertion and thus reduce the possibility of insertion trauma or translocation. Key Words: Hearing preservation, electrocochleography monitoring, cochlear microphonics, cochlear implantation.

1aPP5. Cochlear implant pitch and lexical tone perception in quiet and noise using F0-rate coding. Andrew E. Vandali (Cochlear Ltd., Level 1, 174 Victoria PDE, East Melbourne, Victoria 3002, Australia, avandali@cochlear.com), Zachary Smith (Cochlear Ltd., Sydney, New South Wales, Australia), Komal Arora (Cochlear Ltd., East Melbourne, Victoria, Australia), Lei Xu, Jianfen Luo, Ruijie Wang, Xiuhua Chao (Otolaryngol. Head and Neck Surgery, Shandong Provincial ENT Hospital, Jinan, Shandong, China), and Yi Zheng (Cochlear Medical Device Co., Ltd., Beijing, China)

Fundamental frequency (F0) cues to voice/musical pitch can be coded by pulse-rate or amplitude modulation (AM) rate of electrical pulse-trains in cochlear implant (CI) systems. However, for most clinical strategies (e.g., ACE), temporal cues to F0 are often poorly coded, particularly for female and children's voices and/or in background noise. Strategies such as Optimized Pitch And Language (OPAL) and Fundamental Asynchronous Stimulus Timing (FAST) enhance temporal F0 cues. For OPAL, F0 is estimated and used to control the AM rate coded by electrical pulse-trains in channel containing harmonics of F0. Significant benefits of OPAL to pitch ranking, speech intonation, and Mandarin lexical tone recognition have been observed compared to ACE. For FAST, peak-timing information in channel envelopes is used to derive pulse-timing in each channel and therefore much lower stimulation rates are provided resulting in longer battery life. In bilateral CI systems, it provides better ITD sensitivity, lateralization, and spatial release from masking compared to ACE. Presented here are the results of two studies, the first explored pitch perception of harmonic tones presented in quiet and noise using experimental OPAL and FAST strategies, and the second examined Mandarin lexical tone perception in quiet and noise using ACE and OPAL in a clinical processor.

1aPP6. Impact of incomplete feedback on response bias. Evelyn M. Hoglund (Speech and Hearing Sci., Ohio State Univ., 104a Pressey Hall, 1070 Carmack Rd., Columbus, OH 43210-1234, hoglund.1@osu.edu) and Lawrence L. Feth (Speech and Hearing Sci., Ohio State Univ., Columbus, OH)

The goal of most laboratory studies is to produce a bias-free estimate of sensitivity. Forced-choice procedures with trial-by-trial feedback, equal *a priori* probability of signal occurrence, and a balanced pay-off matrix were designed to assess listener sensitivity with response bias forced to zero. (Green, 1960; Green and Swets, 1964; 1988). Real-world detection tasks rarely provide N-interval forced-choice opportunities. The probability of signal occurrence may vary from near zero in vigilance tasks to near 100% for some clinical testing. Further, many real-world listening situations do not lend themselves to trial-by-trial feedback. Davis (2015, 2017) conducted a single interval yes-no, tone-in-noise experiment measuring response bias with incomplete feedback. Liu (2020) used Bayesian analysis to account for participant variability in the Davis data. Results from different subjects were combined to draw inferences about the effect of each experimental condition on listener bias. As expected, complete feedback drives the response criteria toward the unbiased point, and incomplete feedback conditions result in various degrees of deviation from the optimal criterion. Critical incomplete feedback conditions included responses on signal trials, "yes" responses, and correct responses. Implications for the design of experiments intended to approximate listening conditions in realistic detection tasks are discussed.

1aPP7. Validating a model of bimodal hearing. Brett A. Swanson (R&D, Cochlear Ltd., 1 University Ave., Macquarie University, Sydney, New South Wales 2109, Australia, bswanson@cochlear.com) and Amanda Fullerton (R&D, Cochlear Ltd., Melbourne, Victoria, Australia)

Bimodal hearing is defined as using a cochlear implant (CI) on one ear and a hearing aid (HA) on the other ear. This study tested a model of bimodal hearing proposed by Dieudonné and Francart (2020) in 23 bimodal listeners. Firstly, the relative hearing performance of the CI and HA ears was characterized by scores for Consonant-Nucleus-Consonant (CNC) words in quiet. Secondly, Speech Reception Thresholds (SRTs) were measured for CI alone and bimodal listening, with speech (sentences) from the front, and speech-weighted noise either from the front or the HA side. Bimodal benefit (the improvement in SRT when adding a second ear to the subject's best ear) was 1.2 dB ($p < 0.001$***) averaged across all subjects. However, for the subset of 18 subjects who obtained better scores with CI alone than HA alone ("CI dominant"), there was no significant bimodal benefit. For the subset of 8 CI-dominant subjects who obtained bimodal benefit when speech and noise were co-located at the front, the benefit was lost when the noise was moved to the HA side, demonstrating that binaural cues were not being used.

1aPP8. Peripheral neural status and auditory temporal resolution in cochlear implant users. Jeffrey Skidmore (The Ohio State Univ., 915 Olentangy River Rd., Columbus, OH 43212, Jeffrey.Skidmore@osumc.edu), Yi Yuan, and Shuman He (The Ohio State Univ., Columbus, OH)

Auditory temporal resolution is important for understanding speech with a cochlear implant (CI). The aim of this study was to determine the relative contributions of peripheral neural synchrony and peripheral neural survival to perceptual temporal resolution in CI users. To date, study participants included 8 post-lingually deafened adult CI users. All participants used Cochlear™ Nucleus® devices in their test ears. For each implanted ear tested in each participant, auditory temporal resolution, neural synchrony, and neural survival were evaluated at multiple electrode locations along the array. Auditory temporal resolution was estimated using single-channel, psychophysical gap detection thresholds (GDTs), peripheral neural synchrony was estimated using the phase-locking value (PLV), and peripheral neural survival was estimated using the sensitivity of the electrically-evoked compound action potential to changes in interphase gap (i.e., IPG effect). After adjusting for multiple comparisons, our preliminary results indicated that GDT was significantly correlated with the PLV but not with the IPG effect. The PLV and the IPG effect were not significantly correlated. These preliminary results suggested that the degree of peripheral neural synchrony appeared to play a more important role in determining auditory temporal resolution than peripheral neural survival. [Work supported by NIH: 1R01 DC016038, 1R01 DC017846, R21 DC019458]

1aPP9. Balance of peripheral input and integrity of the electrode-nerve interface in a large cohort of children with bilateral cochlear implants. Carina J. Sabourin (Biological and Biomedical Eng., McGill Univ., 3655 Promenade Sir-William-Osler, Montreal, QC H3G 1Y6, Canada, carina.sabourin@mail.mcgill.ca), Sharon L. Cushing, Blake C. Papsin (Dept. of Otolaryngol.–Head & Neck Surgery, The Hospital for Sick Children, Toronto, ON, Canada), Stephen G. Lomber (Biological and Biomedical Eng., McGill Univ., Montreal, QC, Canada), and Karen A. Gordon (Dept. of Otolaryngol.–Head & Neck Surgery, The Hospital for Sick Children, Toronto, ON, Canada)

The study aim was to determine whether asymmetries in the electrode-nerve interface were present in children using bilateral cochlear implants (CIs). Bilateral CIs improve hearing over unilateral CIs in children with profound deafness in both ears but benefits decrease with increasing asymmetric activity and function between ears. Effects of asymmetry between bilateral CI arrays, CI stimulation parameters, impedance and transimpedance measurements, and electrophysiological thresholds could exacerbate asymmetries at the electrode-nerve interface but these factors have not been assessed in children using bilateral CIs. Clinically generated data from a large cohort of Canadian children (n = 669 children, n = 1332 devices) was gathered retrospectively for analyses. To account for repeated measures, a mixed effects modeling analysis was conducted. Despite CI electrical stimulation levels and electrophysiological thresholds being largely stable over time, asymmetries based on implantation sequence were observed. The observed bilateral differences suggest spread of current and sensitivity to electrical stimulation vary between devices in the same individual. These peripheral asymmetries may coincide with previously reported plasticity-induced cortical changes to increase asymmetries in auditory function between ears and restrict access to binaural hearing cues.

1aPP10. A subjective evaluation of different types of music in music therapy using an emotion equalization app. Man Hei Law (Comput. Sci. and Eng., Hong Kong Univ. of Sci. and Technol., Clear Water Bay, Kowloon, Hong Kong, mhlawaa@connect.ust.hk), Hoi Ting Leung, and Andrew B. Horner (Comput. Sci. and Eng., Hong Kong Univ. of Sci. and Technol., Hong Kong, Hong Kong)

Music therapy has been shown to be effective at improving mood (i.e., valence) in subjects in previous studies. In this study, we investigate five different ways of generating playlists of therapeutic music to see which is most effective at improving mood, tested over a large number of subjects. We conducted online experiments under three proposed approaches using playlists of consoling, relaxing, and uplifting music, and compared them to two baseline methods using natural sounds and a random mix of songs (i.e., music that could be either relaxing, uplifting, or consoling). We questioned subjects before and after listening to the music to see how valence and arousal changed, and analyzed the results with ANOVA. The results showed there were positive changes in listeners' valence levels for natural sounds, as well as for consoling, relaxing, and uplifting music playlists. There were no significant changes in arousal for most sources, but there was a lowering in arousal with relaxing music and a rising in arousal with uplifting music. Additionally, relaxing and uplifting music playlists both showed significant effects in moving listeners from negative to more positive emotional states.

1aPP11. Enhancing emotional well-being through active music listening: A study on the effects of music rhythm games on mood improvement in Hong Kong University Students. Hoi Ting Leung (Comput. Sci. and Eng., Hong Kong Univ. of Sci. and Technol., Hong Kong, Hong Kong), Man Hei Law (Comput. Sci. and Eng., Hong Kong Univ. of Sci. and Technol., The Hong Kong University of Sci. and Technol., Clear Water Bay, Kowloon, Hong Kong, mhlawaa@connect.ust.hk), and Andrew B. Horner (Comput. Sci. and Eng., Hong Kong Univ. of Sci. and Technol., Hong Kong, Hong Kong)

Anxiety and depression are growing problems worldwide, and the massive disruption of mental health care services during the COVID-19 pandemic has highlighted the need for more accessible and convenient digital tools for mental health care. Music therapy has been recognized as an effective non-invasive treatment for mental illnesses, but little research has been done on the therapeutic effects of active music therapy in digital formats. This paper investigates the effectiveness of active music therapy in a music rhythm game format for improving mood and examines the impact of five different song sequencing methods (random, all-relaxed, all-upbeat, upbeat-to-relaxed, and relaxed-to-upbeat) on therapeutic outcomes. Listening tests were conducted with 89 subjects (all university students in Hong Kong) to analyze changes in valence and arousal. The results indicate that all proposed song sequencing methods positively increased arousal levels, while all methods, except all-upbeat, significantly increased valence levels. All-relaxed and all-upbeat gave the best increase in arousal. Upbeat-to-relaxed and relaxed-to-upbeat gave the best increase in valence. All-relaxed and upbeat-to-relaxed were most effective in increasing both arousal and valence. The study also found that tailored recommendations based on subjects' initial arousal and valence levels and levels of anxiety and depression can further improve therapeutic outcomes.

1aPP12. The emotional characteristics of the piano, celeste, and harp with different pitch and dynamics. Hiu Ting Chan (Dept. of Comput. Sci. and Eng., The Hong Kong Univ. of Sci. and Technol., Hong Kong, Hong Kong), Bing Yen Chang, Man Hei Law, Andrew Horner (Dept. of Comput. Sci. and Eng., The Hong Kong Univ. of Sci. and Technol., Sai Kung, Hong Kong), and Wenyi Song (Dept. of Comput. Sci. and Eng., The Hong Kong Univ. of Sci. and Technol., Clear Water Bay, Kowloon, Hong Kong NA, Hong Kong, wsongak@cse.ust.hk)

This work investigates how pitch and dynamics influence the characters of the piano, celeste, and harp. We conducted listening tests where listeners gave absolute judgements on these sounds over ten emotional categories, and analyzed the data with logistic regression. With respect to pitch, the categories Happy, Romantic, Comic, Calm, and Shy had an arched shape that peaked around C6. Heroic, angry, and sad decreased with pitch. Mysterious did not show a clear trend with pitch. Scary had an asymmetric U-shape that was especially strong at the lowest pitch. With respect to dynamics, happy, heroic, comic, and angry were stronger for loud notes, whereas Romantic, Calm, Mysterious, Shy, and Sad were stronger for soft notes. For scary, loud and soft notes were about the same. Overall, pitch and dynamics had about an equally important effect on the piano, celeste, and harp emotional characteristics. Some characteristics such as Happy were affected more by pitch, while others such as Shy were affected more by dynamics. The emotional characteristics of the piano, celeste, and harp were largely similar for most categories, though there were some differences for categories such as Mysterious.

1aPP13. Noise and dynamic decision making. Hang Zhou, Brett R. Molesworth (Univ. New South Wales, Australia, Sydney, New South Wales, Australia), Marion Burgess (Univ. New South Wales, Australia, UNSW, Canberra, Australian Capital Territory 2610, Australia, m.burgess@adfa.edu.au), and Julie Hatfield (Univ. New South Wales, Australia, Sydney, New South Wales, Australia)

Dynamic decision-making (DDM) is a cognitive skill where real-time interdependent decisions are made in an environment which is changing, for example for emergency services, firefighting, battlefield, and aircraft operations to name a few. These are all situations where moderate noise is present, yet little is known about its effect on DDM. A series of research studies investigated the effect of 75 dBA broadband noise on performance for a simulated water distribution task that requires constant DDM. The effect of the noise on performance was found to be moderated by workload, sex, and task familiarity. The noise was detrimental to females' performance at low workload but not at high workload. Males were generally unaffected by noise regardless of the workload level. This suggests that an optimum level of arousal or stimulation may trigger individuals' effort to maintain performance in stressors. Furthermore, there may be some benefit from moderate broadband noise for individuals experiencing high cognitive workload. Not unexpectedly, increased task familiarity, through increased experience and extended task instructions, helped reduce the negative noise effect, which is consistent with Maximal Adaptability Theory.

1aPP14. Analysing the effect of non-acoustical parameters on physiological response due to traffic noise exposure. Manish Manohare (Dept. of Architecture and Planning, Indian Inst. of Technol., Roorkee, IIT Roorkee, Haridwar District, Roorkee, Uttarakhand 247667, India, mmanish@ar.iitr.ac.in), Rajasekar Elangovan (Dept. of Architecture and Planning, Indian Inst. of Technol., Roorkee, Roorkee, India), TIn Oberman, Francesco Aletta, Jian Kang (Inst. for Environ. Design and Eng., Univ. College London, London, United Kingdom), and Manoranjan Parida (Dept. of Architecture and Planning, Indian Inst. of Technol., Roorkee, Roorkee, India)

Traffic noise, as an environmental stressor, impacts both auditory and non-auditory human health through a combination of acoustical and non-acoustical parameters. This study investigates the effects of sex, noise sensitivity, and cross-cultural differences on physiological stress responses to traffic noise exposure. A psychophysiological listening experiment was conducted with 60 participants from the United Kingdom and India to analyze the variation in physiological responses to the noise stimuli. The findings reveal significant gender differences in psychophysiological responses, with females exhibiting higher physiological arousal and potentially experiencing greater stress levels at moderate loudness levels compared to males. Moreover, individuals with high noise sensitivity displayed heightened stress levels and physiological arousal. Additionally, British participants exhibited higher physiological arousal, as indicated by increased skin conductance response parameters, compared to the Indian group. This suggests that exposure to Indian traffic noise, characterized by louder sounds and increased honking, leads to heightened stress levels among British participants. In conclusion, these findings contribute to our understanding of the relationship between traffic noise and physiological outcomes, shedding light on the influence of non-acoustical parameters. It also underscore the potential health implications of chronic noise exposure on human health.

1aPP15. Impact of train horn noise on sleep disturbance and mental health. Xiaomeng Li (Ctr. for Accident Res. and Rd. Safety - Queensland, Queensland Univ. of Technol., 130 K block, Victoria Park Rd., Kelvin Grove, QLD, 4059, Brisbane, Queensland 4059, Australia, xiaomeng.li@qut.edu.au), Bryn Ellis (Queensland Univ. of Technol., Brisbane, Queensland, Australia), Melinda McDonald, and Wanda Griffin (Ctr. for Accident Res. and Rd. Safety - Queensland, Queensland Univ. of Technol., Brisbane, Queensland, Australia)

Train horns are commonly used at Australian level crossings to raise people's awareness of trains and improve road safety. However, the use of train horn could generate broader social effects than road safety. This study aims to investigate the relationships between train horn use and the sleep quality and mental health of people who live nearby an operational train environment. A questionnaire was developed to measure participants' emotional states (i.e., depression, anxiety and anxiety), sleep quality and exposure to train horns. Participants were divided into three groups according to their residence's distance to train horn use location, i.e., horn impact group, railway impact group and control group. The data collection is still ongoing, and 105 participants completed the questionnaire to the submission date. Results from primary data analysis showed that participants in horn impact group spent a longer time falling asleep and also reported lower sleep quality scores, as compared to the other two groups. No difference in the emotional states was found between groups. The study provided a lens to examine the negative impacts of train horn use in the Australasian environment and shed light on future train horn procedures and applications.

1aPP16. Hearing loss impacts aviator performance and cognitive workload during simulated flight. Heath Jones (Warfighter Performance Group, USAARL, Bldg. 6901, Fort Rucker, AL 36362, heath.g.jones2.ctr@mail.mil), Paula Henry, Jennifer Noetzel, Kyle Hale, Kichol Lee, and J. R. Stefanson (Warfighter Performance Group, USAARL, Fort Novosel, AL)

Hearing loss can render an aviator more susceptible to the adverse effects of degraded communication signals and consequently lead to an increased allocation of mental resources to hear (referred to as listening effort). Army aviation hearing standards, which are primarily based on pure tone and speech recognition test scores in quiet environments, do not necessarily predict the functional impact of hearing loss. The Army has recently adopted a new Military Operational Hearing Test (MOHT) to assess the functional impact of hearing loss. The current study aimed to validate the MOHT, specifically for aviators, and measure subjective workload, task performance, and physiological workload across different simulated hearing loss conditions in both clinical and flight simulator settings. Participants were Army aviators with normal hearing thresholds and underwent clinical testing using the MOHT for normal and hearing loss conditions. In addition to audiometric data, flight performance data were collected from pilots operating a full-motion UH-60 Black Hawk flight simulator for different levels of workloads and compared across the different hearing loss conditions. Subjective and physiological measurements of cognitive workload during simulated flight were also collected. Findings from this study will be leveraged in developing future protocols for aeromedical standards and evaluating hearing loss mitigation strategies using various headset technologies.

1aPP17. Spectral and temporal resolution in listeners with age-dependent hearing loss. Alexander Supin (Inst. of Ecology and Evolution, 33 Leninsky Prospect, Moscow 119071, Russian Federation, alex_supin@mail.ru), Olga Milekhina (Inst. of Ecology and Evolution, Moscow, Russian Federation), Marina Tomozova (Inst. of Ecology and Evolution, Moscow, Москва, Russian Federation), and Dmitry Nechaev (Inst. of Ecology and Evolution, Moscow, Russian Federation)

Two basic mechanisms perform the frequency analysis of sounds: (i) the spectral mechanism presented by a bank of frequency-tuned filters and (ii) The temporal mechanism that uses the temporal structure of the signal. The temporal structure manifests in the autocorrelation function. The roles of the spectral and temporal mechanisms in the age-dependent hearing loss were investigated using rippled-spectrum signals. It has been shown previously that discrimination of two rippled signals with different positions of ripples at the frequency scale is mostly performed the spectral processing; for discrimination between a rippled and "flat" signal, the temporal processing dominated. In cases of hearing loss, the decrease of the acuteness of frequency-tuned cochlear filters resulted in reduction of the ripple-pattern resolution by the spectral mechanism. This effect did not influence the capabilities of the temporal mechanism. However, listeners with the age-dependent hearing loss did not perceive high sound frequencies. High frequencies were more favorable for temporal processing, so the inability to perceive these frequencies negatively influenced the ripple resolution. Thus, resolution of rippled signals may reduce at any discrimination task; however, the mechanisms of the reduction differed depending on the task. [Work supported by The Russian Science Foundation, Grant 23-25-00148.]

1aPP18. Using functional near-infrared spectroscopy in measuring the hearing thresholds in sleeping infants. Demi Gao (Bionics Inst., 384-388 Albert St., East Melbourne, Victoria 3002, Australia, dgao@bionicsinstitute.org), Julia Wunderlich, Onn Wah Lee, Darren Mao, Gautam Balasubramanian, Linty McDonald, and Colette McKay (Bionics Inst., East Melbourne, Victoria, Australia)

Functional near-infrared spectroscopy (fNIRS) is a developing technology that uses near-infrared light to image brain activity in the surface layers of the cortexIt measures changes in oxy- haemoglobin (HbO) and deoxy-haemoglobin (HbR) in response to stimuli, which makes it suitable for objectively assessing hearing thresholds by examining the morphology of sound-evoked fNIRS response. This study recruited twelve sleeping infants with no known hearing loss. A natural recording of the /ba/ speech token was used as the stimulus, which was trimmed and concatenated into a 5.4 s stimulus block. The stimulus was presented monaurally between 35 and 80 dB SPL using an insert earphone. fNIRS responses were recorded from bilateral pre-frontal and temporal regions. We observed a positive peak at around 5–6 s from stimulus onset and followed by a negative trough at around 10–20 s from stimulus onset. The amplitudes and latencies of this response varied with different stimulus intensity levels. Results showed how features of the fNIRS response changed systematically with intensity level. Characterizing the response differences to different stimulus intensity levels will facilitate the development of a model for assessment of hearing thresholds in this population.

1aPP19. Effectiveness of remote post-fitting support for over-the-counter hearing devices. Peggy Nelson (Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, peggynelson@umn.edu), Harley Wheeler (Univ. of Minnesota, Minneapolis, MN), Monica Andriacchi, and Erin O'Neill (GN Adv. Sci., Minneapolis, MN)

Models of hearing device services require revamping in light of the emergence of over-the-counter (OTC) hearing aids. Much effort has gone into remote fitting for OTC customers, but other post-fitting support services are lacking. Our aim is to explore ways of translating important aspects of audiologic counseling to a format that is not dependent on in-person delivery models. Individuals with mild-to-moderate hearing loss were recruited and randomly assigned to one of three treatment groups: (1) a traditional clinical treatment model, (2) a "hands-off" OTC delivery model, or (3) a remote service model with orientation and personalized counseling materials provided via an app interface. We hypothesize that individuals receiving personalized counseling remotely will be as successful as those receiving traditional clinical counseling, and more successful than typical OTC customers. Preliminary results suggest that participants receiving tailored remote support use advanced hearing aid features more often and report similar levels of satisfaction and perceived benefit compared to the clinical delivery model. The remote delivery model also leads to greater improvement in individual hearing goals. Overall, results from this study suggest that a tailored remote delivery model could be an effective way to introduce hearing aids to new users.

1aPP20. Exploring the effect of sound scene on self-directed hearing-aid gain adjustment. Bertan Kursun, Jason Zhang, Khloe F. Sytsma, Aarushi Buddhavarapu (Univ. of Washington, Seattle, WA), and Yi Shen (Univ. of Washington, 1417 NE 42nd St., Seattle, WA 98105-6246, shenyi@uw.edu)

Self-directed gain adjustments may present a plausible solution for fitting a hearing-aid (HA) without the requirement of audiometric tests. In the current experiment, users interacted with their hearing aids using a touchscreen-enabled mobile device while a continuous speech stimulus with background noise was played in the sound field, simulating real-world sound scenes. They explored different settings by dragging a cursor on the touchscreen, experiencing real-time audio changes, and identifying their preferred amplification level. Thirteen older adults with self-identified hearing difficulties participated in the study (RHHI score $\geq$6). The self-fitting procedure was conducted in two sound scenes: Restaurant and Traffic. The speech level was set at 60 dB-A in both sound scenes, and the signal-to-noise ratios were 5 and 10 dB for the Restaurant and Traffic scenes, respectively. Following the self-fitting procedure, real-ear aided responses were measured for the amplification profiles obtained using the two sound scenes. The real-ear aided responses for the two scenes resembled each other, with a median correlation coefficient across participants of 0.81. Nevertheless, a marginally significant difference was found between them, mostly driven by higher or lower frequency regions. This suggests that scene-specific adjustments of hearing-aid fitting may be beneficial for some users.

1aPP21. Psychophysiological healing effects of mechanical massage and imagery music listening. Sang Don Park (Medical and Digital Eng., Hangyang Univ., 174 Solsaem-ro, Gangbuk-gu, Seoul 01192, South Korea, psd408@hanyang.ac.kr), Seonkyeong Kim, Yunjin Lee (Medical and Digital Eng., Hangyang Univ., Seoul, South Korea), Haram Lee, Beta Bayu Santika (Architectural Eng., Hanyang Univ., Seoul, South Korea), and Jin Yong Jeon (Architectural Eng., Hanyang Univ., Seoul, South Korea)

This study was conducted to investigate the psychophysiological healing effect and physiological effects of mechanical massage using a massage chair and listening to imagery music reminiscent of nature or urban places. All participants received three different treatments, including sitting on the massage chair and resting for 3 minutes, receiving massage from the chair, and receiving massage while listening to imagery music. The time interval between each treatment was a minimum of 1 week. Not only psychological responses, including anxiety, stress, and depression, were measured using questionnaire scales before and after each treatment but also physiological responses were measured using Heart Rate Variability (HRV) and electroencephalogram (EEG). Although all treatments were designed to psychophysiologically relax the participants, they showed significantly more positive responses when receiving massage with imagery music compared to other treatments. The results of this study may suggest the potential application of massage and guided imagery music for psychophysiological restoration and could serve as foundational data for developing content for physical healing in individuals.

1aPP22. Towards two-point neuron-driven energy-efficient multimodal open master hearing aid. Adewale Adetomi (Edinburgh Napier Univ., Edinburgh, United Kingdom), Mohsin Raza, Khubaib Ahmed (Univ. of Wolverhampton, Wolverhampton, United Kingdom), Tughrul Arslan (Univ. of Edinburgh, Edinburgh, United Kingdom), Amir Hussain (Edinburgh Napier Univ., Edinburgh, United Kingdom), and Ahsan Adeel (Univ. of Wolverhampton, Wulfruna St., Wolverhampton WV1 1LY, United Kingdom, a.adeel@wlv.ac.uk)

Here we demonstrate a two-point neuron-inspired audio-visual (AV) open Master Hearing Aid (openMHA) framework for on-chip energy-efficient speech enhancement (SE). The developed system is compared against state-of-the-art cepstrum-based audio-only (A-only) SE and conventional point-neuron inspired deep neural net (DNN) driven multimodal (MM) SE. Pilot experiments demonstrate that the proposed system outperforms audio-only SE in terms of speech quality and intelligibility and performs comparably to point neuron-inspired DNN with a significantly reduced energy consumption at any time, both during training and inferencing.

1aPP23. Potential objective measures to assess the presence of hidden hearing loss in young adults with long-term exposure to loud music. Juan P. Faundez Astudillo (Dept. of Linguist., Macquarie Univ., Balaclava Rd., Macquarie Park, New South Wales 2109, Australia, juan.faundez@hdr.mq.edu.au), Edward Hampton, Kasiea Naidoo, Mihali Shetty (Dept. of Linguist., Macquarie Univ., Macquarie Park, New South Wales, Australia), Jessica J. Monaghan (Signal Processing, National Acoust. Labs., Sydney, New South Wales, Australia), Jason Mikiel-Hunter (Dept. of Linguist., Macquarie Univ., Sydney, New South Wales, Australia), and David McAlpine (Macquarie University Hearing, Macquarie Univ., Sydney, New South Wales, Australia)

Several objective measures for identifying individuals with "hidden hearing loss" (HHL), have been proposed based on cochlear synaptopathy and the resulting central changes in neural gain. While the loss of high-threshold auditory nerve fibres may result in weaker middle-ear muscle reflexes (MEMR) in HHL sufferers, binaural processing is likely disproportionately and deleteriously impacted by central gain changes in the same group of individuals. In this study, two groups of young adults (<35 years old) with normal audiometric thresholds were recruited, one of which consisted of students from a music conservatory who co-reported tinnitus and listening-in-noise difficulties, likely as a result of HHL. Participants completed a set of questionnaires as well as undergoing both multi-frequency MEMR testing (four ipsilateral elicitors), and an electrophysiological measure of IPD processing known as the 'Interaural Phase Modulation-Following-Response ("IPM-FR") (evoked using two IPD transitions using a clinical setup). Results show that the 'HHL/Tinnitus' group presents lower scores on the Speech, Spatial and Qualities of Hearing Scale questionnaire, less steep MEMR growth functions, and smaller IPM-FRs compared to the 'control' group. This suggests that both objective measures could be clinically useful to identify populations with HHL.

1aPP24. Improving intelligibility of dysarthric speech in noise for listeners with hearing loss. Sarah Yoho Leopold (Speech and Hearing Sci., The Ohio State Univ., 1070 Carmack Rd., Columbus, OH 43210, leopold.41@osu.edu), Stephanie Borrie, Tyson Barrett (Utah State Univ., Logan, UT), and Eric Healy (Speech and Hearing Sci., The Ohio State Univ., Columbus, OH)

Noise reduction strategies based on time-frequency masking have been shown to provide substantial improvements in the intelligibility of speech in noise for both listeners with normal and impaired hearing. Previous studies on this type of noise reduction have involved intact, well-articulated speech

spoken by healthy talkers (i.e., not disordered speech). However, the prevalence of speech disorders such as dysarthria is substantial, particularly in the aging population, which is highly susceptible to the impacts of hearing loss. We recently demonstrated the considerable impact of background noise on dysarthric speech as well as the effectiveness of time-frequency masking to improve the intelligibility of this disordered speech in noise for listeners with normal hearing (Borrie *et al*., 2023). Here, we present data on the feasibility of time-frequency masking to increase the intelligibility of dysarthric speech for listeners with sensorineural hearing impairment. Preliminary results indicate that the Ideal Quantized Mask, a method of time-frequency masking, significantly improves percent words correct scores of dysarthric speech in noise for listeners with hearing loss. Results will be discussed in relation to the specific impact of background noise on dysarthric speech, and the relationship between sensorineural hearing loss and dysarthric speech.

1aPP25. Neural encoding of Cantonese and Vietnamese tones. Sam Vinh Loc (The Educational Psych. Res. Group, Ho Chi Minh City Univ. of Education, Ho Chi Minh City, Viet Nam), Patrick C. Wong (Brain and Mind Inst., The Chinese Univ. of Hong Kong, 4/F, Hui Yeung Shing Bldg., Shatin, NT, Hong Kong, p.wong@cuhk.edu.hk), Loan Vuong (Brain and Mind Inst., The Chinese Univ. of Hong Kong, Ho Chi Minh City, Viet Nam), Huynh Van Son (The Educational Psych. Res. Group, Ho Chi Minh City Univ. of Education, Ho Chi Minh City, Viet Nam), Thi Hong Dung Nguyen (Fulbright Univ. Vietnam, Hồ Chí Minh City, Viet Nam), and Nikolay Novitskiy (Brain and Mind Inst., The Chinese Univ. of Hong Kong, Shatin, Hong Kong)

The present pilot study investigates the neural encoding of lexical tones in Cantonese and Vietnamese. Better encoding of tones was found in native than non-native tone language speakers in previous research, with fewer studies examining language-specific effects. An identical equipment setup was used to measure neural responses to three lexical tones in Hong Kong (HK) (by Cantonese speakers) and Ho Chi Minh City (by Vietnamese speakers). The three tones were the high rising (35), dipping (214), and low falling (21) tones, of which the 21 and 214 tones were unique to Cantonese and Vietnamese respectively. If better tone encoding is language-specific, Vietnamese and Cantonese adult speakers are expected to show enhanced encoding of these tones. Across the majority of temporal and spectral measures, we did not find significant group differences across tonal categories, suggesting that enhancement of neural encoding found in native speakers was likely to be language general. If confirmed with a larger sample size, these findings may suggest that clinical tools utilizing speech stimuli for the assessment of spoken language processing and its related functions may not necessarily require customization for a specific language. [Work supported by ITF in HK.]

1aPP26. Using virtual sound reproduction for studying L1/L2 speech perception in varying acoustic environments. Yusuke Hioka (Mech. and Mechatronics Eng., Univ. of Auckland, 5 Grafton Rd., Auckland 1010, New Zealand, yusuke.hioka@ieee.org), C. T. Justine Hui (Mech. and Mechatronics Eng., Univ. of Auckland, Auckland, New Zealand), Hinako Masuda (Seikei Univ., Tokyo, Japan), and Catherine I. Watson (Elec., Comput. and Software Eng., Univ. of Auckland, Auckland, New Zealand)

Studying the effect of varying acoustic environments on speech perception is often challenging due to the logistical difficulties in moving participants undertaking listening tests between environments as well as lack of controllability and reproducibility in terms of the acoustics of the test environments. In addition, when the study requires to compare first and second language listeners, it often becomes infeasible due to the availability of participants with required language background in a single geographical location where the study is conducted. Virtual sound reproduction could address the challenge thanks to its ability to reproduce the acoustics of arbitrary spaces at multiple geographic locations in a controllable manner. However, it has not been studied well if the results collected using virtual sound reproduction would be a valid alternative to the results collected in real spaces. This talk will introduce a recent study that investigated the difference of speech perception in varying acoustic environments between first and second language New Zealand English listeners using virtual sound reproduction technology. The talk will particularly focus on how the results collected under virtual acoustic environments assimilates to that collected in the original real acoustic environments between first and second language listeners.

1aPP27. Mechanism of perceptual transition: A study by dichotic presentation. Seiya Funatsu (Prefectural Univ. of Hiroshima, 1-1-71 Ujinahigashi Minami-ku, Hiroshima 734-8558, Japan, funatsu@pu-hiroshima.ac.jp)

The perceptual transition occurs when a person hears a repeated single word without pause, illusory changes of the physically unchanging word are induced. We have been studying the conditions under which perceptual transitions occur. So far, it is unclear whether the perceptual transition occurs in the auditory system or in the brain processing. To elucidate this, we examined whether the perceptual transition occurred upstream or downstream of the superior olivary complex by dichotic presentation. Thirteen Japanese subjects were presented with six Japanese words. There are three types of presentation methods: (1) diotic presentation, (2) diotic presentation with ISI = stimulus length, and (3) dichotic presentation to the left and right ears alternately. We measured the time (T) until the perceptual transition occurred. If T in (3) is close to T in (1), the perceptual transition is thought to occur upstream of the superior olivary complex, and if T in (3) is close to T in (2), it is thought to occur downstream of the superior olivary complex. The results in (3) were almost equal to those in (1), suggesting the possibility that it occurs upstream from the superior olivary complex.

1aPP28. Acoustic and neural classification of multi-verbs in natural Bangla speech. Shankha Sanyal (Lang. and Linguist., Jadavpur Univ., Kolkata, West Bengal 700032, India, ssanyal.sanyal2@gmail.com), Pijush K. Gayen (Lang. and Linguist, Jadavpur Univ., Kolkata, India), Archi Banerjee (IIT Kharagpur, Kharagpur, India, India), Samir Karmakar (Lang. and Linguist, Jadavpur Univ., Kolkata, India), and Dipak Ghosh (Jadavpur Univ., Kolkata, West Bengal, India)

Multi-verb constructions occur frequently in most South-Asian languages. In this work, we focus our attention to multi-verb formations in Bengali language, which are essentially categorized into Complex Predicates (CP) and Serial Verbs (SV). For this a database of 200 recordings of CPs and SVs in Bengali has been prepared. While SVs indicate a sequence of two events, CPs indicate a single event, with the second verb denoted as light verb or its effect bleached. We aim to quantify and categorize, in terms of acoustical and neural parameters, the event construals manifested by the SVs and CPs in Bengali. For acoustic analysis, robust features like spectral centroid, skewness, kurtosis etc have been used to establish a classification paradigm using tSNE and KNN clustering algorithm to distinguish between the event construals of CPs and SVs. In the neural domain, EEG was performed on 5 participants who listened to SV and CP utterances at the word and sentential level. The spectral power and multifractality corresponding to the alpha and theta frequency domain of the EEG signals were evaluated. The results distinguish between the different cognitive loads in the frontal and temporal lobe of the human brain corresponding to the multi-verb processing in Bengali.

1aPP29. EarGenie, an innovative test to measure speech discrimination using functional near-infrared spectroscopy. Linty McDonald (Human Hearing, The Bionics Inst., 384-388 Albert St., Melbourne, Victoria 3002, Australia, lmcdonald@bionicsinstitute.org), Colette McKay, Julia Wunderlich, Demi Gao, Gautam Balasubramanian, Onn Wah Lee, and Darren Mao (Human Hearing, The Bionics Inst., East Melbourne, Victoria, Australia)

EarGenie is an innovative hearing test utilizing fNIRS (functional near-infrared light) technology to measure the brain's response to sound. By detecting variations in the brain's hemodynamic response to auditory stimuli, EarGenie can effectively assess a baby's hearing capabilities. Current audiology practices primarily rely on electrophysiology measures to determine hearing thresholds in infants. This approach poses limitations when it comes to accurately assessing infants with auditory neuropathy. There is no measure of speech discrimination for the paediatric population currently in the clinical setting, which can lead to uncertainties and challenges in providing appropriate care. The EarGenie will fulfill this unmet need to provide clinicians with an objective measure of speech discrimination. Improved clarity for clinicians and families of the diagnosis and management of infants with hearing loss is required to optimize audiology practices.

1aPP30. Speech-in-competition auditory assessment and the influence of the bilingual experience. Katia P. Bustos (Cognit. and Clinical Neuropsychology, National Inst. of Neurology and Neurosurgery MVS, Av. Insurgentes Sur 3877, La Fama, Tlalpan, Ciudad de México, CDMX, Mexico City 14269, Mexico, katia208padilla@gmail.com), Rodolfo S. Vivanco (Cognit. and Clinical Neuropsychology, National Inst. of Neurology and Neurosurgery MVS, Mexico City, Mexico), Frederick J. Gallun (Dept. of Otolaryngol., Oregon Health & Sci. Univ., Portland, OR), Aaron R. Seitz, and Esteban Sebastian Lelo de Larrea Mancera (Dept. of Psych., Ctr. for Cognit. and Brain Health, Northeastern Univ., Boston, MA)

The measurement of audition has historically focused on the detection of pure tones; however, it has been proposed that a thorough evaluation should include measures of language reception in competition. The auditory challenge in the real world occurs mostly in competitive environments that require further auditory processing, and not only rely on peripheral, but also higher-order processes. Nevertheless, innovation of speech in competition assessments has occurred primarily in English and few tools have been translated, validated, and investigated in Spanish. Likewise, the bilingual experience of subjects whose native language is not English has not been sufficiently considered when using speech audiometry in hearing evaluation, casting doubt on assessment validity. In this study, we collected repeated measures of the digits-in-noise and the spatial release from masking tasks in English and Spanish-languages on a sample of fifty-six healthy Mexican subjects with varying degrees of English experience. We found significant differences across two sessions in the spatial release from masking, but not the digits-in-noise test. Additionally, we observed that a participant's bilingual experiences, but not cognitive ability, were predictive of language-related differences in performance. We finish by discussing bilingual requisites for valid and reliable speech-in-competition assessment in linguistically diverse individuals.

1aPP31. Using consonant confusion to predict unexplained hearing loss. Joshua J. Hajicek, Sara E. Harris (Ctr. for Hearing Res., Boys Town National Res. Hospital, Omaha, NE), and Stephen Neely (Ctr. for Hearing Res., Boys Town National Res. Hospital, 555 North 30th St., Omaha, NE 68131, stephen.neely@boystown.org)

Some listeners have difficulty understanding speech in noise that is not explained by traditional audiometry. Speech perception errors can manifest as consonant confusions such as /s/ for /sh/. Consonant confusions contain information that may be used to detect or quantify unexplained hearing loss. We developed an efficient consonant confusion test, Quick-VC, based on ten vowel-consonant-vowel (VCV) phonemes. Speech spectrum noise was added to each VCV to make consonant confusions sensitive to hearing impairment. Puretone threshold averages and consonant confusion matrices were collected from 15 listeners with normal hearing and 45 listeners with hearing loss. A logistic regression with regularized coefficients was performed on principal components of the consonant confusions. After removing the effect of audibility, the model predicted unexplained hearing loss greater than 5 dB for seven individuals with a cross-validated area under the curve (AUC) of 0.757, which is clinically acceptable. Six of these seven listeners reported bilateral tinnitus and four had a history of noise exposure, which implicates the peripheral auditory system. The Quick-VC test may be a valuable tool for identifying and quantifying unexplained hearing difficulties, perhaps due to partial loss of auditory nerve fibers, for which there is currently no clinical test.

1aPP32. Selective listening in checkerboard and interrupted speech stimuli with two talkers. Jun Hasegawa (Kyushu Univ., 4-9-1 Shiobaru, Minami-ku, Fukuoka 8158140, Japan, hasegawajun586@gmail.com), Kazuo Ueda (Kyushu Univ., Fukuoka, Japan), Hiroshige Takeichi (Riken, Yokohama, Japan), Gerard Remijn, and Emi Hasuo (Kyushu Univ., Fukuoka, Japan)

A two-talker situation is tougher than a single-talker situation when one has to selectively listen to a target speech. To simplify experimental conditions, we focused on the situation that two talkers speak different sentences alternating with each other. At the same time, we introduced two types of alternations: alternation only in time (staggered speech) and alternation in time and frequency (two-talker checkerboard speech). To our knowledge, the intelligibility of two-talker checkerboard speech stimuli had been never investigated. Three steps of segment duration (20, 80, and 320 ms) for both stimulus types and three steps of the number of frequency bands (4, 8, and 20) for checkerboard speech stimuli were employed. Japanese native listeners ($N = 10$ or 22; age range: 21–34) successfully heard out a specific talker's sentence while ignoring the other for both stimulus types. The intelligibility curves of two-talker checkerboard speech stimuli exhibited characteristic U-shaped curves as found in the previous investigations with one-talker checkerboard speech stimuli (although the intelligibility for two-talker checkerboard speech stimuli went down by 5–20%), suggesting that a common auditory grouping mechanism works in this case also.

1aPP33. The influence of temporal (a)synchrony and linguistic complexity on audiovisual speech perception. Liesbeth Gijbels (Speech and Hearing, Univ. of Washington, 1715 NE Columbia Rd., Seattle, WA 98195, lgijbels@uw.edu), Kaylah Lalonde (Boys Town National Res. Hospital, Omaha, NE), Mark Wallace (Dept. of Hearing and Speech Sci., Vanderbilt Univ. Medical Ctr., Nashville, TN), and Adrian K. C. Lee (Speech & Hearing Sci., Univ. of Washington, Seattle, WA)

Audiovisual (AV) cues often lead to improved speech understanding, especially when the auditory input is impoverished (e.g., in background noise, for people with hearing loss). This AV improvement occurs when both the auditory and visual information can be integrated together effectively (e.g., through temporal synchrony). However, whether the complexity of speech signals influences this integrated percept and how this contributes to speech understanding is still unknown. This study investigates whether individual differences in perception of temporal synchrony of the auditory and visual speech signal explain individual variability in AV speech recognition, and whether this differs across different levels of linguistic processing. By implementing five different levels of linguistic complexity (i.e., varying phoneme-viseme connections, vocabulary knowledge and/or linguistic context) and introducing temporal asynchrony in a remote task with English speaking adults we ask the following questions: (1) Is temporal synchrony needed to observe AV speech perception benefits? (2) Does this vary by linguistic level? (3) Does linguistic complexity influence synchrony perception? (4) And does synchrony perception explain individual variability in AV speech perception? These findings add value in deciding how AV synchronization should be balanced in hearing aid designs as well as in augmented / virtual reality devices.

1aPP34. Interrupted mosaic speech revisited: Gain and loss of stretching on intelligibility. Kazuo Ueda (Acoust. Design, Kyushu Univ., Fukuoka, Japan, ueda@design.kyushu-u.ac.jp), Masashi Hashimoto (Acoust. Design, Kyushu Univ., Fukuoka, Japan), Hiroshige Takeichi (ADSP, R-IH, RIKEN, Yokohama, Japan), and Kohei Wakamiya (Acoust. Design, Kyushu Univ., Fukuoka, Japan)

Periodicity and temporal fine structure provide a perceptual cue to connect speech fragments when the speech is interrupted. In mosaic speech stimuli, which are noise-vocoded with stepwise amplitude and spectral

envelopes, interruption drastically reduces intelligibility because of lacking the perceptual cue. Here we report that the intelligibility of mosaic speech stimuli, consisting of 20 frequency bands and 20-ms segment duration, decreased from 95% to 78% and 33% when they were interrupted by 20- and 80-ms gaps. However, when the mosaic segments were stretched to fill the silent gaps, the intelligibility improved again to 92% and 54%, resulting in gains of 14% and 21% respectively. By contrast, intelligibility dropped by 7% with stretching and filling 160-ms gaps. The intelligibility minimums (9%–16%) were observed at the 160-ms original gap duration irrespective of stretching ratios. It is possible that stretching with short gaps effectively restored the original amplitude envelopes, whereas, with long gaps, the effect was detrimental. Furthermore, a probability summation model with a short and a long integration window, by which the intelligibility minimums were predicted for lowpass-filtered interrupted speech stimuli, seems to be highly relevant to the current results.

1aPP35. Checkerboard and interrupted speech: Critical intelligibility differences observed in factor-analysis-based checkerboard speech stimuli. Kazuo Ueda (Acoust. Design, Kyushu Univ., Fukuoka, Japan, ueda@design.kyushu-u.ac.jp), Linh L. Doan (Acoust. Design, Kyushu Univ., Hanoi, Viet Nam), and Hiroshige Takeichi (ADSP, R-IH, RIKEN, Yokohama, Japan)

It has been shown that the intelligibility of checkerboard speech stimuli, in which speech signals were periodically interrupted in time and frequency, drastically varied according to the combination of the number of frequency bands (2–20) and segment duration (20–320 ms). However, the effects of the number of frequency bands between 4 and 20 and the ways of frequency divisions on intelligibility have been basically unknown. Here we show that 8-band checkerboard speech stimuli were more intelligible than temporally interrupted speech stimuli and 4-band checkerboard speech stimuli, irrespective of segment duration ($n = 19$ and 20). At the same time, U-shaped intelligibility curves were observed for 4-band and possibly 8-band checkerboard speech stimuli. Furthermore, when the frequency divisions were determined according to regular intervals on a critical bandwidth scale rather than factor analysis results of speech power fluctuations, intelligibility improved at the 160- and 320-ms segment duration. These results suggest that the factor-analysis-based frequency bands perform as speech cue channels that are essential for speech perception and that a probability summation model based on a perceptual unit of speech perception may account for the U-shaped intelligibility curves if the model is modified to include the speech cue channels.

1aPP36. Perceptual identification of high vowels in Taiwan Mandarin. Chenhao Chiu (Graduate Inst. of Linguist., National Taiwan Univ., No. 1, Sec. 4, Roosevelt Rd., Taipei 10617, Taiwan, chenhaochiu@ntu.edu.tw), Jianzhi Huang, and Po-Hsuan Huang (Graduate Inst. of Linguist, National Taiwan Univ., Taipei, Taiwan)

Taiwan Mandarin contrasts high vowels /i/, /y/, and /u/ in the aspects of tongue backness and lip roundedness. Despite that both /y/ and /u/ are associated with the [round] feature, the postural realizations of these two vowels are not always identical, though sometimes only very subtle. The current study investigates whether native speakers can reliably identify the three high vowels from natural speech when no acoustic signal was available. The results show that the highest identification accuracy was found in /i/, followed by /u/ and then /y/. The same order was also reported in the analyses of sensitivity. On the other hand, the two rounded vowels were easily confused, as revealed by the analysis of false alarm. The results also showed that participants may pay attention to different postural cues when distinguishing /u/ and /y/, along with a slight bias towards /y/. While it is rather challenging for native speakers to identify /u/ from /y/, our results did provide a selection of postural exemplars for each vowel, which may be considered as more canonical and subject to further analyses using image processing.

1aPP37. How well do hearing aids amplify "speech glimpses" in multitalker mixtures? Virginia Best (Dept. of Speech, Lang. and Hearing Sci., Boston Univ., 635 Commonwealth Ave., Boston, MA 02215, ginbest@bu.edu)

Reductions in speech audibility are more detrimental in the presence of competing sounds, where there is little redundant speech information, than in quiet. Indeed, our previous work suggests that insufficient audibility of "speech glimpses" may partly explain the poor performance of listeners with hearing loss in multitalker mixtures. An implication of those results is that restoring audibility across the spectrum may be especially critical in such situations. Here we asked whether current hearing-aid amplification strategies adequately restore the audibility of speech glimpses. We used ideal time-frequency separation to isolate target speech glimpses from unaided and aided speech mixtures. We then measured intelligibility for the isolated glimpses and for the original mixtures. Participants were young adults with bilateral sensorineural hearing loss, and individualized non-linear amplification (NAL-NL2) was provided using the Oldenburg open Master Hearing Aid. Results show that amplification generally improved the audibility and intelligibility of the target speech glimpses. For some participants, the intelligibility improvements carried over to the mixtures. For others, the improvements were diminished in the mixtures, suggesting that there were counteracting effects. This approach may provide a useful way to unpack positive and negative effects of hearing-aid processing in multitalker mixtures.

1aPP38. Difference of phonemic restoration between professional and non-experts' voices. Maori Kobayashi (Aichi Shukutoku Univ., 2-9, Katahira, Nagakute, Aichi 480-1197, Japan, maorik@asu.aasa.ac.jp) and Masato Akagi (JAIST, Nomi, Japan)

It is generally thought that the voices of professional announcers are clearer and easier to hear than those of non-professional speakers. This study examined the perceptual mechanisms involved in the higher intelligibility of professional announcers' voice with the focus on perceptual restoration. The intelligibility of the professional announcers was 10% to 30% higher than that of the non-professional speakers in noisy conditions. The effects of the formants on the higher intelligibility were experimentally examined using test words with one of the frequency bands corresponding to the first to third formants removed. When the formant component was removed, performance declined for both speakers, whereas that of the professional announcers was maintained with noise. An additional experiment showed that a missing component in a professional announcers' voice is perceptually restored by adding noise. It is thus likely that the voices of professional announcers are rich in phonological cues, enabling phonological restoration to be achieved by adding noise. This may be one of the reasons why the higher intelligibility of their voices in noise.

1aPP39. Disentangling factors responsible for children's pronounced susceptibility to speech-in-speech masking. Margaret Miller (Boys Town National Res. Hospital, 555 North 30th St., Omaha, NE 68131, margaret.miller@boystown.org), Emily Buss (Univ. of North Carolina, Chapel Hill, NC), Angela AuBuchon (Boys Town National Res. Hospital, Omaha, NE), Aimee Miller (Univ. of Florida, Gainesville, FL), and Lori Leibold (Boys Town National Res. Hospital, Omaha, NE)

The ability to recognize masked speech follows a prolonged time course of development, particularly when the masker is also speech. Findings from prior research suggest that immature auditory segregation, selective auditory attention, and receptive language abilities contribute to these maturational effects. This study aimed to disentangle the relative contributions of these factors on speech-in-speech recognition for 5- to 8-year-old children and young adults with normal hearing. Speech-in-noise and speech-in-speech recognition were assessed using an adaptive, forced-choice procedure with a picture-pointing response. Tone detection thresholds in quiet and in the

presence of a remote-frequency, narrowband noise were estimated using an adaptive, 2 down 1 up forced-choice procedure. Standardized assessments of receptive vocabulary, executive function, and working memory were completed. Preliminary data agree with results from prior studies, indicating increased susceptibility to auditory masking in the context of speech recognition and tone detection for children relative to adults. Associations between performance on speech recognition, psychoacoustic, linguistic, and cognitive tests will be presented.

1aPP40. How does a history of mild traumatic brain injury alter speech understanding in complex auditory environments? Conner Corbett (Oregon Health and Sci. Univ., 3181 SW Sam Jackson Rd., NRC04, Portland, OR 97239, corbetco@ohsu.edu), Lauren Charney, Nicole Dean (Oregon Health and Sci. Univ., Portland, OR), Vanja Pešić (Boston College, Boston, MA), G. Christopher Stecker (Ctr. for Hearing Res., Boys Town National Res. Hospital, Omaha, NE), Aaron R. Seitz (Northeastern Univ., Boston, MA), and Frederick Gallun (Oregon Health and Sci. Univ., Portland, OR)

Listening in complex environments is a challenge for most observers, but particularly for those with mild traumatic brain injury (mTBI). Here, we explored the relationship between working memory (WM) and the ability to understand speech in competition, using three spoken sentences presented simultaneously via earphones using a Virtual Spatial Array. The three utterances were either presented at the same spatial location or different locations. Participants completed both an auditory/visual WM task and the competing speech task. The WM task involves auditory and visual list recall as well as the ability to recall lists from one modality while ignoring the other or while attending to both modalities. Fifty adults, some with and some without a history of mTBI, participated. Their ages ranged from 18 to 74 years and they either had normal hearing or moderate hearing loss. Results showed relationships between the WM and speech tasks consistent with previous results. Further analyses will be presented exploring the ways in which a history of mTBI influences the ability to use (1) spatial cues and (2) WM to make sense of competing speech utterances.

1aPP41. The effects of hearing loss and envelope processing on speech understanding in adverse acoustics during childhood. Viji Easwar (National Acoust. Labs., 16 University Ave., Sydney, New South Wales 2109, Australia, viji.easwar@nal.gov.au), Z. Ellen Peng (Boys Town National Res. Hospital, Omaha, NE), Sriram Boothalingam (National Acoust. Labs., Sydney, New South Wales, Australia), and Mark Seeto (Hearing Australia, Sydney, New South Wales, Australia)

Children with hearing loss (CHL) experience greater difficulty understanding speech in the presence of noise and reverberation relative to their normal hearing (CNH) peers despite provision of appropriate amplification. The fundamental frequency of voice (f0)—a salient temporal cuei—could play a significant role. However, the nature of deficits and its relationship with speech understanding are poorly understood. To this end, we evaluate the role of f0 encoding on speech perception abilities of CNH and CHL in the presence of noise and/or reverberation. In 14 CHL and 29 CNH, envelope following responses (EFRs) were elicited by vowels, modified to estimate f0 encoding in low (<1 kHz) and higher frequencies. Both groups demonstrated a frequency-dependent dichotomy in the disruption of f0 encoding– greater disruption at low frequencies due to noise and greater disruption at the high frequencies due to reverberation. In contrast to CNH, CHL demonstrated: (a) greater disruption of f0 encoding at low frequencies, particularly in the presence of reverberation, (b) a positive relationship between f0 encoding at low frequencies and speech discrimination. Together, these results provide evidence for the persistence of suprathreshold temporal processing deficits in children despite the provision of appropriate amplification to compensate for hearing loss.

1aPP42. Effect of energetic and informational masking on speech reception performance in middle-aged adults. Mai Yuasa (Chiba Univ., 1-33 Yayoi-cho, Inage-ku, #706, Sci. and Technol. 2, Chiba-shi, Chiba 263-8522, Japan, mai.0405@chiba-u.jp), Sho Otsuka, and Seiji Nakagawa (Chiba Univ., Chiba-shi, Chiba-ken, Japan)

Speech reception in the presence of competing sounds declines in middle age. Experiments on young normal-hearing listeners have revealed that speech reception performance in the presence of competing sounds is determined by the cumulative effects of two different types of masking: energetic masking and information masking. Here we examined how energetic and informational masking degrades speech reception in middle-aged adults. Fifteen young (seven males, eight females, 20–26 years) and seventeen middle-aged (four males, thirteen females, 47–57 years) listeners with normal hearing participated in the experiment. Speech reception in the presence of competing sounds was assessed by the Coordinated Response Measure. Participants were required to listen to a target phrase spoken by a female in the presence of noise or a single interfering phrase spoken by the same talker, same-sex talker, or different-sex talker. Speech reception performance was lower in middle-aged adults than young adults, regardless of the type of disturbing sounds. The performance differences between the same-talker and different-sex talker conditions, which reflects the effects of informational masking, was larger in middle-aged adults. These results suggest that both energetic and informational masking contributes to the degradation of speech reception in middle-aged adults.

1aPP43. Solving the cocktail party problem using Multi-modal Hearing Assistive Technology Prototype. Mandar Gogate (School of Computing, Edinburgh Napier Univ., 10 Colinton Rd., Edinburgh EH10 5DT, United Kingdom, m.gogate@napier.ac.uk), Kia Dashtipour, and Amir Hussain (School of Computing, Edinburgh Napier Univ., Edinburgh, United Kingdom)

Hearing loss is a major global health problem, affecting over 1.5 billion people. According to estimations by the World Health Organization, 83% of those who could benefit from hearing assistive devices do not use them. The limited adoption of hearing aids can be attributed to the suboptimal performance in acoustically challenging environments, such as cocktail parties, where there are multiple competing speakers and noise sources leading to poor speech intelligibility for hearing impaired listeners. This work presents a first-of-its-kind real-time, multi-modal speech enhancement system that can effectively enhance speech in challenging real noisy environments. The system exploits both audio and visual cues to isolate the target speaker's voice from interfering background noises. The system is developed using deep neural networks and is integrated with the Open Master Hearing Aid (openMHA) platform. The integrated prototype was evaluated in a cocktail party setting to isolate the target speaker's voice from interfering speakers, music, and other non-speech noise sources and was able to significantly improve the speech intelligibility of hearing impaired listeners. This work has the potential to improve the quality of life for hearing impaired listeners by facilitating effective communication in cocktail party environments.

Session 1aSA

Structural Acoustics and Vibration and Physical Acoustics: Non-Negative Acoustic Contribution Analysis

Steffen Marburg, Cochair
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Contributed Papers

11:00

1aSA1. Analysis of contribution degree of vibration transmission of bolted structure. Shuai Liu (Harbin Eng. University/The Nuclear Power Inst. of China, No. 145, Nantong St., Nangang District, Harbin, No. 328, Section 1, Changshun Ave., Shuangliu District, Chengdu, Sichuan 610213, China, liushuai_9105@163.com)

In practical engineering, the vibration transmission between bolted structures supported by cantilever beams involves many factors. The vibration transmission effect is the embodiment of the coupling of various factors, so it is impossible to accurately and quantitatively evaluate the influence degree of various factors. In this paper, the influence factors of vibration between bolted plates in the form of cantilever beam are studied by numerical calculation method. The effects of friction coefficient, bolt preload, temperature and pressure on vibration characteristics are studied. Finally, the contribution of different influencing factors to vibration transmission is given.

11:20

1aSA2. Sound energy-based surface contribution analysis for the vehicle interior noise problem. Caglar Gurbuz (Chair of Vibroacoustics, Tech. Univ. of Munich, Boltzmannstr. 15, Garching 85748, Germany, caglar.guerbuez@tum.de) and Steffen Marburg (Chair of Vibroacoustics, Tech. Univ. of Munich, Garching, Germany)

Booming noise is a major comfort problem in vehicle cabins occurring in lower frequencies. The method of panel or surface analysis serves as a viable tool to trace the sound emitting components on the enveloping chassis. Traditional techniques evaluate the surface contributions with respect to the sound pressure level at the driver's position. However, using the sound pressure level as the control objective implies two major drawbacks: Firstly, the local sound pressure is highly sensitive to the position of the evaluation point. As a consequence, the predictions of the surface contributions become unreliable at frequencies or areas with low sound pressure levels. Secondly, surface contributions in existing techniques can be positive or negative, which facilitates the event of acoustic shortcircuits. This talk presents an accurate and robust contribution analysis method for the vehicle interior noise problem. For this purpose, the sound energy density is used as the control objective. A quadratic form yields energy-based contributions, which are always non-negative and thus avoiding acoustic shortcircuits. These findings demonstrate that contributing surface are effectively identified even in regions with sound pressure levels. As such, the evidence from this study suggests that the sound energy density provides an accurate and robust control objective.

11:40

1aSA3. Contribution of jet flow noise sources to the far-field sound power. Esmaeel Eftekharian (School of Mech. and Manufacturing Eng., UNSW Sydney, UNSW Sydney High St., Kensington, New South Wales 2052, Australia, e.eftekharian@unsw.edu.au), Paul Croaker (Platforms Div., Defence Sci. and Technol. Group, Melbourne, Victoria, Australia), Richard Sandberg (Dept. of Mech. Eng., School of Eng., The Univ. of Melbourne, Melbourne, Victoria, Australia), Daniel Wilkes (Ctr. for Marine Sci. and Technol., Curtin Univ., Perth, Western Australia, Australia), Steffen Marburg (Chair of Vibroacoustics of Vehicles and Machines, Dept. of Mech. Eng., Technische Universität München, Garching, Germany), and Nicole Kessissoglou (School of Mech. and Manufacturing Eng., UNSW Sydney, Kensington, New South Wales, Australia)

Jet noise is the dominant source of noise during aircraft takeoff. This study uses direct numerical simulation (DNS) data of subsonic jet flow obtained from computational fluid dynamics (CFD) simulations to perform acoustic analyses. The non-negative contribution technique (NNC) is employed to identify aeroacoustic sources of jet noise with the greatest contribution to far-field sound power. The NNC technique uses the Lighthill source distribution with an acoustic impedance matrix constructed from radiation kernels of the free-field Green's function. The fast multipole method (FMM) is integrated with the NNC technique to reduce computational cost arising from decomposition of the acoustic impedance matrix as well as from the high number of CFD grids associated with the DNS data. The results showed that the dominant aeroacoustic sources are associated with the first component of the Lighthill tensor (T_{11}) and are concentrated along the jet centerline, downstream of the nozzle exit.

Session 1aSC

Speech Communication: Phonetics of Under-Documented Languages I (Poster Session)

Benjamin V. Tucker, Cochair
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All posters will be on display and all authors will be at their posters from 10:20 a.m. to 12:20 p.m.

Contributed Papers

1aSC1. Near-merger of /s: ʃ/ in Veneto Italian: A preliminary cross-generational acoustic investigation. Angelo Dian (School of Lang. and Linguist., Univ. of Melbourne, Babel Bldg., Rm. 422, Melbourne, Victoria 3052, Australia, a.dian@unimelb.edu.au), John Hajek, and Janet Fletcher (School of Lang. and Linguist, Univ. of Melbourne, Melbourne, Victoria, Australia)

Italian contrasts /s:/ and /ʃ/. Although in standard Italian (SI) the two phonemes have distinct places of articulation (laminal denti-alveolar and laminal post-alveolar, respectively), early descriptions of the Italian spoken in the Veneto (VI) report a near-merger due to the absence of /ʃ/ in the regional substratum dialect (Canepari 1984). Accordingly, /s:/ is retracted and aplicalized, and /ʃ/ is advanced and in some cases also apicalized in VI. To investigate this phenomenon, an acoustic and statistical analysis of spectral Center of Gravity (COG), known to increase with more advanced constriction location for coronals, is conducted on 192 /s:/ and /ʃ/ tokens produced by six VI speakers (three males and three females) of varying ages (34 to 68) producing eight repetitions of /ˈkassa kasˈsata ˈlaʃa laˈʃata/ ‘case, cassata cake, he/she/it leaves, left’ within carrier sentences. Results show that, regardless of sex, the two older speakers (50 and 68) but not the four younger speakers (34 to 44) produce statistically similar COG values for the two consonants, indicating similar constriction locations. This preliminary study suggests that the near-merger may be produced by older speakers but not younger speakers today, in an ongoing process of standardization of Italian pronunciation across regional varieties.

1aSC2. Retroflex nasals in the Mai-Ndombe (DRC): The case of nasals in North Boma B82. Lorenzo Maselli (UGent / UMons / FWO Vlaanderen, Sleepstraat 61A, Ghent 9000, Belgium, lorenzo.maselli@ugent.be)

Retroflex consonants represent a major class of language sounds, but our understanding of their phonetic and phonological behaviour is still relatively limited. From the standpoint of acoustics, recent contributions are largely lacking. Within retroflex consonants, liquids are particularly rare, and arguably very little descriptive and/or theoretical (and/or historical) research has been conducted on them, with a few exceptions for a number of Australian and Indian languages. Bantu languages have mostly been left out of broader surveys, a few cases notwithstanding. In this talk, we show why retroflex nasals could constitute a unique testing ground for phonetics, phonology, typology, and historical linguistics alike. Thanks to recent fieldwork in the Mai-Ndombe region (southwestern DRC), we were able to confirm the existence of nasal retroflexes in North Boma (B82), as initially attested by Stappers (1986). We present an acoustic description of the segments at hand, which represents the first detailed study of this kind for the Bantu languages of the region. We also hypothesise that the presence of retroflexion in the Mai-Ndombe might be a substrate feature originating in extinct hunter-gatherer languages once spoken by groups (known as “Batwa”) which still inhabit the area (Saïdi Hemedi *et al.*, 2012: 3).

1aSC3. Cross-linguistic realization of lateral ejective affricates in connected versus isolated speech. Ted K. Kye (Dept. of Linguist, Univ. of Washington, Guggenheim Hall, 3940-2425 Benton Ln., Seattle, WA 98195, tkye29@uw.edu) and Maida Percival (Univ. of Toronto, Vancouver, BC, Canada)

Typological research on ejectives has focused on the realization of stops in isolated speech (Kingston 1985, Lindau 1984). However, there has been little research on the realization of ejectives, and in particular affricates, in connected versus isolated speech. Given that lateral affricates can be produced with variable realization, this study compares the acoustics of [tɬʼ] in isolated and connected speech for speakers of three languages: Lushootseed (Coast Salish), Hul’q’umi’num’ (Coast Salish), and Dene Kədə́ (Dene/Athabaskan). Duration, spectral moment, and voice quality measurements were examined from corpus data of word lists and connected speech. Results indicated that there was greater voice onset time (VOT), longer closure duration, and a smaller frication duration to VOT ratio in isolated speech than connected speech, supporting Lindblom’s (1990) and Farnetani & Recasens’ (2013) view that words are produced more hyperarticulated in isolated speech. Cross-linguistic differences were found in the duration of frication, center of gravity, and the rise to peak amplitude of the following vowel. Dene Kədə́ had greater frication duration and a shallower intensity slope than the Salish languages, indicators of differences in place of articulation and degree of affrication. This suggests different realizations of [tɬʼ] across languages.

1aSC4. An quantitative analysis of Punjabi tones. Kiranpreet Nara (Linguist, Univ. of Toronto, 100 St. George St., Toronto, ON M5S 3G3, Canada, kiranpreet.nara@mail.utoronto.ca)

Punjabi has three tones, the falling and the rising tones developed due to the historically lost voiced aspirated consonants and word-final glottal fricative /ɦ/, and the default tone occurred elsewhere. While there has been some experimental research on Punjabi tones, there have been limitations due few stimuli and speakers. The main aim of the current study was to provide a

phonetic examination of the fundamental frequency (f0) patterns across a large number of Punjabi words produced by multiple speakers. The experiment was conducted online using 24 native speakers (9F, 15M) of Indian Punjabi. The list of stimuli contained 66 monosyllabic words (default = 18, falling = 15, rising = 33) with either /a/ or /ə/. The speakers were recorded producing the stimuli in carrier and natural sentence environments. Linear mixed effects analyses were conducted on six f0 measures: the onset, mid, and offset of the vowel, the f0 range (MaxF0-MinF0), and the f0 trajectory in the first (MidF0-BegF0) and final (EndF0-MidF0) halves of the vowel. The results confirmed three distinct pitch curves for the three tones. Each of the three tones were distinguished from one another for the onset f0 and the EndF0-MidF0 measures. Vowel quality influenced tone realization. Between-speaker and word variation was observed.

1aSC5. An investigation of ongoing tonal changes in Punjabi. Kiranpreet Nara (Linguist, Univ. of Toronto, 100 St. George St., Toronto, ON M5S 3G3, Canada, kiranpreet.nara@mail.utoronto.ca)

Punjabi developed the falling and the rising tones due to the loss of the voiced aspirated consonants and the word-final glottal fricative /ɦ/ and the default tone appeared elsewhere. A falling pitch has recently been observed to develop in words with an initial glottal fricative and both falling and rising pitch patterns have been observed in words without any apparent conditioning factor (‘novel’ words). These developments within some traditionally default tone words have been only briefly mentioned in the literature (Shackle, 2003; Schniske, 2015; and Bhardwaj, 2016) and have not been investigated experimentally. The aim of this study was to document the f0 patterns observed in the glottal-initial and novel words and determine how they compare to the f0 of the traditionally default tone words. 24 native speakers of Indian Punjabi (9F, 15M) were recorded online. The stimuli contained 35 monosyllabic words (glottal-initial = 7, novel falling = 4, novel rising = 6, default = 18) with either /a/ or /ə/. The acoustic analysis showed falling f0 for the glottal-initial and novel falling words and a rising f0 for the novel rising words. Welch’s t-tests were conducted to compare the default tone to each of the three newer pitch patterns. The results showed that nearly half of the speakers produced a non-default pitch on the glottal-initial and novel words. Considerable within- and between-speaker variation was observed.

1aSC6. Brazilian Portuguese vowel discrimination by Nungon and Tashelhit speakers. Hannah Sarvasy (The MARCS Inst., Western Sydney Univ., Penrith, New South Wales, Australia), Weicong Li (The MARCS Inst., Western Sydney Univ., Sydney, New South Wales 2052, Australia, weicong.li@westernsydney.edu.au), Jaydene Elvin (Linguist, California State Univ., Fresno, CA), and Paola Escudero (The MARCS Inst., Western Sydney Univ., Penrith, New South Wales, Australia)

The vowel system of speakers’ native languages are known to affect perception of vowels in foreign languages (Elvin *et al*., 2021). In a field-based behavioral experiment, we tested the perception of Brazilian Portuguese vowels (seven-vowel system) by speakers of the Papuan language Nungon (six-vowel system) and the Berber language Tashelhit (three-vowel system). 60 native speakers of Nungon and 38 native speakers of Tashelhit participated. The acoustics of Nungon vowels were previously analyzed using a 30-point analysis technique (Sarvasy *et al*., 2020). For the present study, we also analyzed Tashelhit vowels using the same technique. We then used discriminant analysis to analyze the relationship between speakers’ performance on the Brazilian Portuguese vowel discrimination task and the acoustics of vowels in their native languages. We present preliminary results of these analyses, and address their possible contributions to four areas: (a) description of the acoustics of under-described languages, (b) the relationship between native vowel systems and perception of foreign vowels, (c) methods for undertaking rigorous acoustic and experimental research in noisy, field environments, and (d) theoretical frameworks for phonetic learning such as the L2LP model (Escudero, 2005; van Leussen and Escudero, 2015).

1aSC7. Expanding research on ʔayʔaǰuθəm interrogative intonation. Tyler T. Schnoor (Linguist, Univ. of AB, 150 Assiniboia Hall, Edmonton, AB T6G2E7, Canada, tschnoor@ualberta.ca) and Mary A. McCarthy (Linguist, Univ. of AB, Edmonton, AB, Canada)

Relatively little research has been done on ʔayʔaǰuθəm prosody. This lack of research is due in part to the idea that stress and tone are not distinctive features in the language. However, recent studies have found evidence of suprasegmental features in ʔayʔaǰuθəm interrogatives. Additionally, previous work indicates that—as has been reported with other Salishan languages—there is no significant rise in fundamental frequency at the end of ʔayʔaǰuθəm interrogatives, though the mean vowel fundamental frequency is higher in interrogative than in declarative sentences. These findings are not unique to ʔayʔaǰuθəm, but they do contribute to growing evidence that a final rise in fundamental frequency in interrogative sentences is not universal. In the present study, we aim to address the limitations of our previous work by expanding on the amount of data used, incorporating methods for analyzing the entire fundamental frequency contour, and measuring the mean fundamental frequency of vowels from more balanced stimuli. Although we do not expect different results from the present study, it is our hope that the methodologies used will increase the validity of our findings on ʔayʔaǰuθəm interrogative intonation.

1aSC8. The vowel inventory of Nakanamanga. Shubo Li (Australian National Univ., Baldessin Precinct Bldg., 110 Ellery Crescent, Canberra, Australian Capital Territory 2600, Australia, shubo.li@anu.edu.au) and Rosey Billington (Australian National Univ., Canberra, Australian Capital Territory, Australia)

Nakanamanga is an Oceanic language spoken by approximately 10,000 speakers in central Vanuatu (Lynch and Crowley, 2001), including northern Efate island and Nguna and other small islands to the north. Previous work on Nakanamanga indicates that it has five distinctive vowel qualities (/i,e,a,o,u/), as is typical of Oceanic languages, and some evidence of long vowels (Schütz, 1969). However, whether vowel length is contrastive remains unclear, and there are suggestions that vowel length exhibits complex relationships with consonantal and prosodic patterns (Schütz, 1969; McClintock, 1991; Schmidt, 2023). We present findings of a pilot study targeting vowel length in Nakanamanga, based on phonological evidence and acoustic phonetic data. A wordlist consisting of words of CVCV structures was designed after consultation with a native speaker, and recorded with the speaker in Mere-Sauwia village on Nguna. Data was processed and analysed using Praat, EMU-SDMS and R (Boersma and Weenink, 2023; Winkelmann *et al*., 2017; R Core Team, 2023). Results for vowel duration and first and second formant frequency at vowel midpoints provide strong evidence that Nakanamanga has ten distinct vowels, with a length contrast for all five vowel qualities. Findings lay the groundwork for larger-scale investigations of the phonetics and phonology of Nakanamanga.

1aSC9. Acoustic and durational correlates of diphthong contrasts in Nafsan. Rosey Billington (Australian National Univ., Baldessin Precinct Bldg., 110 Ellery Crescent, School of Lit., Lang. & Linguist., Canberra, Australian Capital Territory 2600, Australia, rosey.billington@anu.edu.au)

The phonetic realization of diphthongs, vowels with dynamic productions involving two targets and often described as inherently long, is understudied cross-linguistically (e.g., Petersen, 2018), including within many well-documented languages. Nafsan, an Oceanic language spoken by around 6000 people on Efate island in Vanuatu, has a vowel inventory with monophthongs of five qualities, all with a length contrast (Billington *et al*., 2021), and at least six contrastive closing diphthongs (Thieberger, 2006). These include /ei/ and /oi/, and two pairs in which the primary difference is viewed as the height of the second target: /ai/ and /ae/, and /au/ and /ao/. However, there are indications that duration may be an important correlate in these pairs. This presentation reports on a study of the six closing diphthongs

based on data collected with six Nafsan speakers in Erakor village, using a wordlist of mostly monosyllabic words with diphthongs in both open syllables and syllables closed with a coronal consonant, produced in an utterance-medial frame. Results show that, in addition to differences in second formant frequency, /ae/ and /ao/ are significantly longer than /ai/ and /au/, raising questions about whether the contrast is best interpreted as a length contrast similar to that found among monophthongs.

1aSC10. The intonation of Wh-questions in two Basque varieties. Nerea Delgado Fernandez (Modern Lang. and Linguist, Florida State Univ., 625 University Way, Tallahassee, FL 32304, ndelgadofernandez@fsu.edu)

This presentation examines the intonation of Wh-questions in two under-investigated varieties of Basque: North-central Gipuzkoan (Spain) and Labourdin (France). A total of 288 sentences elicited through a semi-spontaneous task were analyzed in Praat (Boersma and Weenek, 2018). As acoustic cues, we center on final tonal movements and lengthening, two aspects for which there is limited to no information for the varieties under study. Results for final contours in Gipuzkoa showed that 58.2% of sentences were produced with rising intonation (L* H% begin the most predominant contour), while the falling contour attested in other Basque varieties (L* L%) appeared in only 41.8% of cases. In Labourdin Basque, 77.3% of sentences were produced with a rising (L* H% also being the most prevalent configuration). Speakers, however, tended to produce risings of mid height. Regarding lengthening, this cue was more relevant in Gipuzkoan Basque. We will connect these novel findings to the intonational characteristics described in prior studies (Elordieta, 2003; Elordieta and Hualde, 2014; Duguine and Irurtzun, 2020). Moreover, since Basque in these territories is in contact with Spanish and French, a brief comparison across languages will also be included. This will be the first attempt at such a comparison.

1aSC11. A sociophoentic study of intergenerational differences in Kemie language. Xirui Liu (Dept. of Lang. and Cultures, La Trobe Univ., Bundoora, Melbourne, Victoria 3083, Australia, xirui.liu@latrobe.edu.au)

Previous sociophonetic studies have primarily concentrated on European languages or other dominant languages, with Chinese dialects and minority languages receiving less attention. This article aims to examine the phonetic variations of vowels, stops and tones emerged owing to age differences in Kemie Language, an Austroasiatic language in China. 18 speakers were categorized by age into young, middle-aged and older groups with 6 speakers in each group. Acoustic parameters such as formants, VOT, f0 and duration were extracted, and linear mixed effects model and repeated measures ANOVA in R were employed to test whether the above parameters were significantly different across the groups. Results show that: (1) The monophthongs /ɛ/ /a/ /ɔ/ manifest significant variations through tongue height or frontness or both. (2) VOT values of /b/ in the above three groups are significantly different and inverse with age, while no consistent conclusions can be reached in terms of /d/ /g/. (3) Low-rising (13) and high-rising (35) tones with stop codas (cu yun wei) haven't merged across the groups, but pitch range of the two tones become narrower with decreasing age; Low-rising (13) and mid-level (33) tones with sonorant codas (shu yun wei) have merged across groups.

Session 1aUW

Underwater Acoustics: Inversions of Underwater Sound

Mark K. Transtrum, Chair
Physics and Astronomy, Brigham Young University, N283 ESC, Provo, UT 84602

Contributed Papers

10:20

1aUW1. Parameter reduction for environmental inversions in ocean acoustics. Mark K. Transtrum (Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, mkt24@byu.edu), Jay C. Spendlove (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), and Tracianne B. Neilsen (Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Environmental inversions in ocean acoustics require the simultaneous estimation of many unknown parameters. Unfortunately, many environmental parameters are unidentifiable from available measurements. Recent work in statistical physics has developed tools for analyzing such models using information geometry. The Manifold Boundary Approximation (MBAM) systematically prunes unidentifiable parameters to produce an identifiable model that can be analyzed with traditional statistical methods. In this talk, we report on progress applying MBAM to underwater acoustics models for environmental inversion, focusing on transmission loss (TL) in range-independent normal mode models. The key technical advance is the calculation of geodesics (a generalization of straight lines to curved surfaces) on a geometric representation of the model known as the model manifold. Geodesics inform which environmental parameters are constrained by ocean sounds and guide the removal of unidentifiable parameters. The result is a simplified, identifiable model of comparable accuracy. We summarize physical insights revealed by MBAM applied to ocean inversion. [Work supported by Office of Naval Research.]

10:40

1aUW2. Estimates of geoacoustic parameters and source range using airgun sound in the East Siberian Sea, Arctic Ocean. Dae Hyeok Lee (Dept. of Marine Sci. and Convergence Eng., Hanyang Univ. ERICA, 55, Hanyangdaehak-ro, Sangnok-gu, Ansan 15588, South Korea, edh0921@hanyang.ac.kr), Jee Woong Choi (Dept. of Marine Sci. and Convergence Eng., Hanyang Univ. ERICA, Ansan, South Korea), Dong-Gyun Han (Res. Ctr. for Ocean Security Eng. and Technol., Hanyang Univeristy ERICA, Ansan, Gyeonggi-do, South Korea), Wuju Son, Hyoung Sul La, and Eun Jin Yang (Div. of Ocean Sci., Korea Polar Res. Inst., Incheon, South Korea)

In a shallow-water waveguide, low-frequency sound propagating over several kilometers has dispersive properties and thus the dispersion curves can be used for the inversion of geoacoustic parameters. In September 2019, seismic survey was conducted by the icebreaking research vessel Araon, operated by Korea Polar Research Institute. A single hydrophone was moored at East Siberian Shelf, a nearly range-independent shallow water (<70 m), to receive seismic airgun sounds generated from the R/V Araon. It was observed that the arrivals corresponding to the first two modes were clearly dispersed in the spectrogram of the received signal. In order to use these dispersion curves for the geoacoustic inversion, the dispersion curves corresponding to the first two modes were extracted using the warping transform combined with the wavelet synchrosqueezing transform. Geoacoustic parameters including sound speed and density in the sediment and the distance between the source and receiver were then estimated by comparing the extracted dispersion curves to the model replicas predicted by the KRAKEN normal-mode acoustic propagation model. Finally, the inversion results and their error analysis will be discussed in this talk. [This research was a part of the project titled 'Korea-Arctic Ocean Warming and Response of Ecosystem (K-AWARE, KOPRI, 1525011760)', funded by the Ministry of Oceans and Fisheries, Korea and supported by the National Research Foundation of Korea (NRF) grant funded by the Korea government (MSIT) (2020R1A2C2007772).]

11:00

1aUW3. Application of trans-dimensional particle filtering for geoacustic inversion. Weiwen Wu (Inst. of Acoust. Chinese Acad. of Sci., No. 21 North 4th Ring Rd., Haidian District, Beijing 100190, China, 843751093@qq.com), Qunyan Ren, and Li Ma (Inst. of Acoust. Chinese Acad. of Sci., Beijing, China)

The acquiring geoacoustic model and associated geoacoustic parameters is vital for sound propagation modelling in a range-dependent environment. Conventional sequential methods, e.g., particle filtering and Kalman filtering have been widely used for geoacoustic inversion in range-dependent environment, which encounter challenges when confronted with unknown geoacoustic models that intrinsically varying with range. As an attempt to estimate geoacoustic model and associated geoacoustic parameters as well, a trans-dimensional particle filtering method is presented here. This method integrates birth-death rules into the filtering process, enabling automatic selection of appropriate geoacoustic models and estimating geoacoustic parameters in the same time. Numerical simulations were conducted based on an environment model of the South China Sea, where a sea trial was conducted in 2022. Numerical results demonstrate the efficiency in accurately estimating the geoacoustic model variations and associated parameters. Preliminary results of sea trial data processing are also presented here.

11:20

1aUW4. Extraction and analysis of three-dimensional sound scattering characteristics by body-generated internal waves. Ruixin Nie (School of Naval Architecture, Ocean and Civil Eng., Shanghai Jiao Tong Univ., No. 800, Dongchuan Minxing District, Shanghai City, Shanghai 201100, China, ruixinnie@sjtu.edu.cn), Bin Wang, and Tengjiao He (School of Naval Architecture, Ocean and Civil Eng., Shanghai Jiao Tong Univ., Shanghai, China)

The motion of an object submerged in a stratified fluid generates surface wakes, and simultaneously induces internal waves at the interface where there is a change in sound speed, known as the thermocline. As a result, spectral-temporal fluctuations occur in both the surface height and the distribution of sound velocity. While surface wakes primarily contribute to geometric acoustic scattering, the internal waves generated by the underwater object's motion can have diverse effects on sound propagation, leading to a prolonged acoustic impact that may have practical applications in underwater acoustic detection. This paper investigates the impact of body-generated internal waves on underwater acoustic propagation through the establishment of an "unfrozen field," range-dependent model using the approximated Kelvin wake theory. The model allows numerical simulations to demonstrate the spatial-temporal coherence, time-frequency modulation and directional characteristics of the three-dimensional sound field scattered by the body-generated internal wave. By analyzing the influences of thermocline depth, target motion velocity and source depth, the results presented in

this study indicate that the long-range acoustic propagation, modulated by the body-generated internal waves, can provide additional information for detecting moving targets.

11:40

1aUW5. Options for temperature selection in maximum entropy approach to obtaining source levels and seabed porosity. Jacob R. Nuttall (Phys. and Astronomy, Brigham Young Univ., Provo, UT), Tracianne B. Neilsen (Phys. and Astronomy, Brigham Young Univ., N251 ESC, Provo, UT 84602, tbn@byu.edu), Mark K. Transtrum (Phys. and Astronomy, Brigham Young Univ., Provo, UT), David P. Knobles (Knobles Sci. and Annalysis, LLC, Austin, TX), and William S. Hodgkiss (Scrips Inst. of Oceanogr., San Diego, CA)

When analyzing acoustical data, the goal often is to find source strength and characteristics of the environment. Instead of finding point estimates, a more informative approach is to find probability distributions from either a Bayesian approach or a maximum entropy approach. The Bayesian approach relies on prior distributions and an estimate of the noise covariance, while the maximum entropy approach relies on an estimate of a "temperature" derived from analogies with statistical mechanics. Different representations of the temperature are appropriate in different situations. The best choice is related to confidence in the model being used and if the model captures all the relevant physics. Examples of how the choice of temperature impacts the posterior distributions are shown using a toy model and a real-world application. Specifically, the maximum entropy approach is applied to models of transiting ship noise to obtain posterior distributions of source level and porosity of the sediment layer. These examples show that the selection of temperature significantly impacts the posterior distributions. [Work supported by the Office of Naval Research.]

Session 1pAA

Architectural Acoustics: Anomalous, Scattered and Steered Reflections

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

Densil Cabrera, Cochair
Sydney School of Architecture, Design and Planning, The University of Sydney, Wilkinson Building GO4, Sydney, 2006, Australia

Chair's Introduction—12:55

Invited Papers

1:00

1pAA1. Sound scattering at building façades. Michael Vorlaender (IHTA, RWTH Aachen Univ., Kopernikusstr 5, Aachen 52074, Germany, mvo@akustik.rwth-aachen.de), Anne Heimes, Lili Pan (IHTA, RWTH Aachen Univ., Aachen, Germany), Jonas Kempin, Iremnur Tokac, and Sigrid Brell-Cokcan (ip, RWTH Aachen Univ., Aachen, Germany)

The state-of-the-art models of soundscape planning and noise mapping are based on geometrical acoustics. Numerical wave models are possible but they exceed acceptable computation times for acoustics in practice by orders of magnitude. In geometrical acoustics, the sound is propagated either assuming Lambert's diffusion or specular reflections, which applies to randomly corrugated surfaces and to flat surfaces, respectively. For specific architectural surface design, however, no appropriate simplified sound reflection model exists. In this paper, architectural façade shape typologies are categorized, and a directional scattering coefficient concept will be introduced and applied in an example simulation and auralization of complex sound scattering phenomena.

1:20

1pAA2. Acoustic retroreflectors in architecture: From incidental cases to focusing designs. Densil Cabrera (Sydney School of Architecture, Design and Planning, The Univ. of Sydney, Wilkinson Bldg. GO4, Sydney, New South Wales 2006, Australia, densil.cabrera@sydney.edu.au), Shuai Lu (Tsinghua Univ., Shenzhen, China), Jonothan Holmes (Sydney School of Architecture, Design and Planning, The Univ. of Sydney, Sydney, New South Wales, Australia), Manuj Yadav (Inst. for Hearing Technol. and Acoust., RWTH Aachen Univ., Aachen, Germany), and Dagmar Reinhardt (Sydney School of Architecture, Design and Planning, The Univ. of Sydney, Sydney, New South Wales, Australia)

Retroreflective acoustic treatment of architectural spaces can concentrate early-reflected sound onto the respective arbitrarily located sources. Typically, this is achieved by arrays of corner reflectors (CRs) or cube corner retroreflectors (CCRs). Still greater sound concentration can be achieved by focusing retroreflectors—achieved by curving the CRs' or CCRs' faces. This paper reviews physical instances of and experiments with such retroreflective treatments in buildings. Incidental cases include building facades, Indian stepwells, and arguably the original Concert Hall of the Sydney Opera House. Designed cases include both non-focusing and focusing installations, including experiments with room and ceiling treatments. Implications of such treatments on room acoustical design are discussed.

1:40

1pAA3. Geometric solutions for focusing retroreflectors. Jonothan Holmes (Univ. of Sydney, Sydney, New South Wales, Australia, jonothan.holmes@sydney.edu.au), Shuai Lu (Tsinghua Univ., Shenzhen, China), and Densil Cabrera (Sydney School of Architecture, Design and Planning, The Univ. of Sydney, Sydney, New South Wales, Australia)

Although focusing might normally be an architectural acoustician's nightmare, combining it with retroreflection is potentially beneficial. Focusing acoustic retroreflectors concentrate early reflected sound onto arbitrarily located sources. Dihedral and trihedral retroreflectors can be made to focus by curving one or more of their faces in either one or two dimensions. The relationship between curvature and focal distance is developed empirically using analytic ray tracing and finite-difference time-domain methods. Dihedra can be made more compact by segmenting the cross-sectional curvature similar to the Fresnel lens. Complexities of fabricating doubly curved surfaces can be overcome by approximating with several flat surfaces and physical examples are constructed from bending sheet-steel. Simplified designs are shown to achieve similar performance to ideally curved focusing retroreflectors. For context, performance is compared to coherent and incoherent summation of a perfectly reflecting spherical surface of equivalent solid angle, as well as an equivalent flat surface. Results show increased speech-weighted retroreflected energy level and concentration of sound in the region surrounding the collocated source-receiver over the upper range of typical human speech (1 kHz–8 kHz octave bands) when compared to non-focusing retroreflectors, and substantial increases in reflected energy level compared to a specular reflection at normal incidence.

2:00

1pAA4. Retroreflective room acoustics: Possible treatments, their effects, and potential benefits. Shuai Lu (Shenzhen Int. Graduate School, Tsinghua Univ., Xili University town, Shenzhen 518055, China, shuai.lu@sz.tsinghua.edu.cn), Densil Cabrera, and Jonothan Holmes (Sydney School of Architecture, Design and Planning, The Univ. of Sydney, Sydney, New South Wales, Australia)

Introducing retroreflection into rooms can change the distribution of reflected energy, concentrating early reflections on the source. But what is a retroreflective room, and how could it be implemented? This paper introduces concepts including retroreflective arrays and retroreflective focusing, which achieve much more than the trivial case of a rectangular room. Examples are developed via finite-difference time-domain simulation. Treatment strategies include making use of the room corners and edges as focusing retroreflectors, retroreflective ceiling treatments, and using local retroreflective treatments such as desk reflectors. The acoustic features of the retroreflective rooms are compared with design targets of open-plan offices, classrooms and restaurants, to illustrate potential benefits of retroreflective treatment of rooms for speech.

2:20

1pAA5. Experimental characterization of scattered reflections of sound diffusers using a portable high-resolution acoustic goniometer. Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, xiangn@rpi.edu) and Ziqi Chen (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY)

Surface scattering and diffuse reflections from acoustic diffusers have recently become a significant topic of research for room acoustics. An acoustical goniometer can be implemented in the form of a circular microphone array to characterize scattered responses for various types of diffusing elements/devices. This research is also intended for experimental validation of diffraction simulations of finite-sized diffusing devices using the physical theory of diffraction (PTD). To cope with experimental challenges, an acoustic goniometer must achieve a fine enough angular resolution in addition to fulfilling far-field requirements. This paper will discuss pragmatic implementations and experimental results of portable goniometers with a radius of up to five meters and an angular resolution of 1.25 degrees. The pragmatic implementation is intended to be easily deployable in empty, indoor spaces of sufficiently large dimensions for scattered reflection characterization. This paper highlights an effort of increasing experimental efficiency of measurement routines for oblique incident characterization and challenges of characterizing full angular ranges of scattered reflections.

2:40

1pAA6. Limitations of the scattering coefficient for simulating early reflections. Stefan Feistel (AFMG, Berlin, Berlin, Germany), Michael Vorlaender (IHTA, RWTH Aachen Univ., Aachen, Germany), and Bruce C. Olson (AFMG, Brooklyn Park, MN 55444, bruce.olson@afmg.eu)

The introduction of the scattering coefficient has enabled geometrically-based acoustic simulations to produce full-length impulse responses of fairly high accuracy. However, the scattering coefficient is a random-incidence quantity that can be applied only when considering a statistical energy average over different directions. This model is inaccurate if a single reflection by a surface element is considered for which the scattered part of the sound energy depends strongly on the angle of incidence. The resulting deviation can be particularly relevant for first-order reflections as these reflections have significant influence on acoustic parameters that make use of early-energy estimates, such as D, C80 or LF. In this paper, the uncertainty of the specularly reflected energy is estimated and discussed for a number of typical surface structures and scenarios. The corresponding impact on the accuracy of predicted early-energy parameters is also discussed. Possible solutions and extensions such as an angle-dependent scattering model are outlined.

3:00–3:20 Break

3:20

1pAA7. Filter-based solutions for finite and infinite edge diffraction. Stephan D. Ewert (Medizinische Physik und Akustik, Universität Oldenburg, Carl-von-Ossietzky-Str. 9-11, Oldenburg 26129, Germany, stephan.ewert@uni-oldenburg.de) and Christoph Kirsch (Medizinische Physik und Akustik, Universität Oldenburg, Oldenburg, Germany)

Sound diffraction occurs at (building) corners, objects and openings, prominently affecting occluded and reflected sounds. Here, a physically-based, parametric filter model for diffraction at arbitrary wedges is presented. The model connects existing high-frequency asymptotic and exact solutions in geometrical acoustics using up to four half-order low-pass filters. Compact expressions for the transfer function and impulse response of the singly diffracted field are derived, with errors below 0.1 dB. A filter modification is suggested to account for the exact solution at low frequencies. The filter solution is further simplified to approximate the spectral effects of diffraction from finite wedges and objects composed thereof, referred to as universal diffraction filter approximation (UDFA). A computationally highly-efficient recursive filter implementation is presented and it is demonstrated that diffraction from flat finite objects like plates or apertures can be closely approximated. To account for effects of higher-order diffraction at the object, a heuristic filter extension is suggested. The presented filter-based diffraction solution is suited for the prediction and simulation of sound propagation including non-specular reflections in architectural acoustics. For virtual acoustics, UDFA provides an efficient and accurate approximation of edge diffraction to account for perceptually relevant frequency-dependent attenuation, extendable to arbitrary objects.

Contributed Papers

3:40

1pAA8. Challenges in the use of architectural concrete for sound diffusion—A case of 350 seats concert hall in Poland. Andrzej Klosak (Politechnika Krakowska, Fenna 3/8, Cracow 31-143, Poland, aklosak@pk.edu.pl) and Anders C. Gade (Gade & Mortensen Akustik A/S, Charlottenlund, Denmark)

This paper discusses the design, realization and acoustical performance of a recently opened 350 seats music school concert hall, located in Jastrzębie Zdrój, Poland. Several design aspects are discussed: restrictions due to cost optimization in material selection, geometrical design of "wavy" reinforced concrete walls and acoustical performance of the finished auditorium. Construction details of the interior surfaces their influence on the acoustics are also discussed.

4:00

1pAA9. Abstract withdrawn.

4:20

1pAA10. Micro-perfated membranes for cooling, heating, sound-absorbing, and lighting. Christian Nocke (Akustikbuero Oldenburg, Sophienstr. 7, Oldenburg 26121, Germany, nocke@akustikbuero-oldenburg.de)

25 years ago optically transparent sound absorbers made of micro-perforated sheets were introduced. Over the last yeas various applications and developments have been conducted. Optically transparent as well as translucent, coloured and printed sheets have been introduced. Other functions such as heating and cooling have been added into a stretched ceiling as well as modular systems. Sound absorption data according to ISO 354 for different set-ups are presented. pplications in various spaces will be presented and discussed. Day-light ceilings, mirror ceilings as well as absorbers in front of glass will be shown. Finally, some case studies will be discussed.

4:40

1pAA11. Analysis of the sound field in circular spaces using small scale models and parallel phase-shift interferometry. Mayuko Imanishi (Dept. of Intermedia Art and Sci., Waseda Univ., 3–4–1 Ohkubo, Shinjuku-ku, Tokyo, Japan, mayuko1024@asagi.waseda.jp), Haruka Nozawa, Nanase Suzuki (Dept. of Intermedia Art and Sci., Waseda Univ., Shinjuku-ku, Tokyo, Japan), and Yasuhiro Oikawa (Dept. of Intermedia Art and Sci., Waseda Univ., Shinjuku-ku, Japan)

In acoustic design, the use of circular spaces is generally avoided. This is because these spaces present several acoustic problems, for example, "whispering gallery," and low diffusivity of sound. These room acoustic problems have generally been dealt with by installing diffusers on walls. In this study, we designed the acoustics of the circular space by applying diffusers based on primitive roots and analyze the sound field. These diffusers were proposed by Schroeder and take advantage of the randomness of the numerical sequence generated from primitive roots. We designed several small circular models attached the diffusers and printed them with a 3D printer. The sound field in the models were measured and visualized using parallel phase-shifting interferometry, one of the optical sound measurement methods. As a result, "whispering gallery" phenomenon was improved, and focal points were less likely to occur by the application of diffusers based on primitive roots.

5:00

1pAA12. Acoustic reflector utilizing sound speed control by multiple obstacles. Yuma Saito (School of Informatics, Kogakuin Univ., Bldg.5 #405, 2665-1 Nakano-machi, Hachioji-shi, Tokyo 192-0015, Japan, jx20108@ns.kogakuin.ac.jp), Hidetoshi Masukawa (Graduate School of Eng., Kogakuin Univ., Tokyo, Japan), and Yoshinori Takahashi (Kogakuin Univ., Tokyo, Japan)

Acoustic diffusers have a structure that diffuses sound waves in various directions by utilizing the interference of sound waves caused by surface irregularities. For example, a phase diffraction grating based on number theory has a structure with equally spaced depressions of irregular depth, and the maximum difference between the depressions is half the target wavelength. If the reflector targets 500 Hz sound waves, the thickness of at least 34 cm is required. On the other hand, the acoustic lens proposed by Kock *et al*. in the 1950s composed by multiple spheres or other obstacles to control the speed of sound. This report proposes a diffuser without depressions of irregular depth by applying the acoustic lens principle of Kock *et al*. The difference in reflection time caused by the depressions of the diffuser can be controlled by multiple obstacles.

Session 1pAB

Animal Bioacoustics: Auditory Perception and Cognition in Animals

Rebecca Dunlop, Cochair
University of Queensland, 15 Launceston Street, Salisbury, 4107, Australia

Cynthia F. Moss, Cochair
Psychology and ISR, University of Maryland, Biology-Psychology Building 2123M, College Park, MD 20742

Chair's Introduction—1:35

Invited Papers

1:40

1pAB1. Using behavioural response experiments to measure humpback whale hearing in noise. Rebecca Dunlop (Univ. of Queensland, The University of Queensland, Brisbane, Queensland 4072, Australia, r.dunlop@uq.edu.au), Michael Noad (The Univ. of Queensland, Gatton, Queensland, Australia), and Dorian Houser (National Marine Mammal Foundation, San Deigo, CA)

Currently, there are no direct data on mysticete hearing in noise. Available data comes from anatomical modelling, the assumption they can hear their own sounds, and studies on the effects of various sources of anthropogenic noise on their behavior. This study used a behavioral response study design to quantify humpback whale hearing in natural ocean noise. Tonal signals, ranging from 250 Hz to 16 kHz, were used as the stimuli, and a change in humpback group behavior indicated the whales heard the signal. Individual whale and group behavior were quantified using a combination of land-based tracking data of groups and tag data deployed on individual whales to record fine-scale 3D movement underwater. The signal-to-noise ratio was estimated at the initial response position of the group or whale, assuming this was the level at which they first detected the tone in noise. Results confirm that humpback whales responded to signals in noise at detection levels comparable to other marine mammals and that their ability to hear signals in noise at higher frequencies is better than expected. This provides empirical data on hearing in a mysticete which can be used to better predict the acoustic impacts of anthropogenic noise on marine mammals.

2:00

1pAB2. Complex sound perception by laboratory mice. Micheal L. Dent (Psych., Univ. at Buffalo, SUNY, B76 Park Hall, Buffalo, NY 14260, mdent@buffalo.edu), Laurel Screven (Psych., Univ. at Buffalo, SUNY, Buffalo, NY), Kali Burke (Psych., Univ. at Buffalo, SUNY, Belcamp, MD), Anastasiya Kobrina, and Payton Charlton (Psych., Univ. at Buffalo, SUNY, Buffalo, NY)

Although mice are known to produce vocalizations with varying spectrotemporal characteristics, exactly how they are using these vocalizations for communication is still not entirely known. Ultrasonic vocalizations (USVs) are produced by male and female mice of many strains and researchers have parsed these USVs into different categories. Whether or not these USV categories are meaningful to the mice has been the subject of many preference and playback studies, which have weaknesses. Using an operant conditioning nose poke paradigm, we have trained laboratory mice to detect, discriminate, and identify simple and complex stimuli, including USVs used for communication. We have found that USVs are easier to detect than pure tones, although noise, aging, and blast trauma disrupts this process. Like humans listening to words, the beginnings of USVs appear to hold more importance than the middle or ends of USVs. Mice can discriminate amongst USVs, and there is evidence for categorization of at least some USVs, but this ability is much worse than that of other animals tested on similar tasks. Psychophysical experiments with awake trained mice are important for determining the perceptual limitations of the mouse auditory system and using these rodents as models for human communication disorders.

2:20

1pAB3. Perception of social vocalizations in bats. Kirsten M. Bohn (Psychol. and Brain Sci., Johns Hopkins Univ., 4411 Grandview Ave., Baltimore, MD 21211, kbohn1@jhu.edu) and Michael Smotherman (Biological Sci., Texas A&M, College Station, TX)

Although bats are well known for their highly specialized echolocation systems, the majority of produce social vocalizations whose form, function, and complexity vary widely across taxa. Here we review and compare two types of social vocalizations bats produce and perceive: infant isolation calls and complex songs. Infant isolation calls are simple (2 notes), innate vocalizations produced by the majority of mammals with little specialization in bats. Research has shown that auditory sensitivity co-evolved with isolation call frequencies across bats and bats can discriminate among individuals using isolation calls. In contrast to isolation calls, *Tadarida brasiliensis* songs are much more complex (100's of notes), hierarchically structured and highly variable across renditions. Playbacks revealed that bats rapidly perceive echolocation from passing conspecifics and sing in response. This occurs only in response to echolocation, even though

"echolocation" calls are embedded within complex song phrases. Thus, bats can rapidly discriminate between "echolocation alone" and "song with echolocation embedded" signals. We further share some evidence of differences in the syntax of songs produced in response to songs from familiar and unfamiliar bats. These studies and new discoveries of bats' songs will no doubt have a large impact on our understanding of vocal perception and cognition.

2:40

1pAB4. Acoustic behaviour in bottlenose dolphins during two different target discrimination tasks. Sara Torres Ortiz (Univ. of Southern Denmark, Calle Perdomo 1, Piso 2, puerta 6, Puerto de la Cruz, Santa Cruz de Tenerife 38400, Spain, sara.torres@bi.mpg.de), Ariana Hernandez Sanchez (Loro parque, Puerto de la Cruz, Spain), Freja Jakobsen (Univ. of Southern Denmark, Odense, Denmark), Javier Almunia (Loro parque, Puerto de la Cruz, Spain), and Magnus Wahlberg (Univ. of Southern Denmark, Kerteminde, Denmark)

Echolocation is an active sensory system that gives an acoustic representation of the surroundings by the animal emitting clicks, detecting and analysing echoes. However, little is known about dolphin biosonar in moving animals, as well as of the process of controlling click emissions, and the cues used by the animal to discriminate between different targets. We studied how bottlenose dolphins may use their dynamic sound production and hearing abilities, along with head and body movements, to detect and classify objects through echolocation. We designed two different experiments with blindfolded bottlenose dolphins (*Tursiups truncatus*) swimming freely along the pool and actively discriminating between two targets of three different materials, and a second one to determine the capability to discriminate wall thickness differences in cylindrical targets of the same material. By means of synchronized cameras and hydrophones, we observed clear differences between individuals of the same species in terms of ability to discriminate, but similar performance with the harbor porpoises in relation to the materials chosen with the difficulty of the task. Studying how the dolphins solved a simplified echolocation task, allow a better understanding of the processes behind target detection and discrimination capabilities during natural biosonar circumstances.

3:00

1pAB5. Attention modulates cortical, but not subcortical responses. Victoria Figarola (Carnegie Inst. of Technol., Carnegie Mellon Univ., 1403 N Saint Clair St., Apt. 2D, Pittsburgh, PA 15206, vif@andrew.cmu.edu), Abigail Noyce (Pyschology, Carnegie Mellon Univ., Pittsburgh, PA), Adam Tierney (Ctr. for Brain and Cognit. Development, Birkbeck College, Univ. College London, London, United Kingdom), Ross Maddox (Biomedical Eng., Univ. of Rochester, Rochester, NY), Fred Dick (Experimental Psych., Univ. College Longon, London, United Kingdom), and Barbara Shinn-Cunningham (Neurosci. Inst., Carnegie Mellon Univ., Pittsburgh, PA)

Organisms often need to attend to one signal within a crowded acoustic scene. While attentional effects in the auditory cortex are robust, what happens subcortically remains mysterious. We combined previous electroencephalography paradigms to measure whether attention modulates subcortical as well as cortical responses. Participants heard competing, temporally interleaved low and high melodies. Each melody had notes repeating at 2 Hz, with an overall note rate of 4 Hz. Participants attended to either the high- or low-pitched melody and responded whenever a 3-note pattern repeated in that stream. From past work, we expected to see strong cortical EEG responses at 2 Hz, but with a phase that shifted depending on which melody listeners attended. Importantly, we engineered the notes so that (1) low and high melodies excited different cochlear regions, and (2) each pitch period of each note elicited an auditory brainstem response (ABR). This allowed us to test whether attention altered responses along the ascending pathway in the brainstem. We found phase shifts in cortical responses with shifts of attention. While we found robust ABRs, we saw no evidence that attention modulated any of the ABR components. These results suggest that attentional effects in subcortical structures are weak, at best.

3:20–3:40 Break

3:40

1pAB6. Biosonar sampling strategies in frequency-modulated echolocating bats. Amaro Tuninetti (Cognit., Linguist., & Psychol. Sci., Brown Univ., Box 1821, Providence, RI 02912-9067, Amaro_Tuninetti@brown.edu), James Simmons (Neurosci., Brown Univ., Providence, RI), and Andrea Simmons (Cognit., Linguistic, & Psychol. Sci., Brown Univ., Providence, RI)

Echolocating bats dynamically control multiple aspects of their biosonar emissions (emission rate, frequency, bandwidth, duration, and direction) in parallel as a function of their surroundings and current goals. As environments become more difficult to navigate and tasks become more complex, bats converge on sampling strategies that have less variability in the timing and spatial changes of their echolocation. We investigated these biosonar dynamics within a virtual echo target presentation paradigm where we challenged big brown bats sitting on a platform to localize a single echo target, which was either stationary or moved unpredictably in azimuth. Acoustic clutter was also added at varying distances to investigate its effect on their sampling strategies during the task. Bats modified their biosonar strategies differently depending on the distance of added acoustic clutter and the distance that a target shifted, and individuals tended to converge on similar strategies. We also show that certain perceptual contexts can lead to unexpected biosonar strategies in individual bats, such as the use of longer-duration emissions with less emission 'clustering' when localizing a virtual target against nearby clutter.

4:00

1pAB7. Where do Egyptian fruit bats fly when auditory and visual spatial information conflicts? Nikita Finger (Psychol. and Brain Sci., Johns Hopkins Univ., Baltimore, MD), Shivam Chitnis (Neurosci., Johns Hopkins Univ., Baltimore, MD), and Cynthia F. Moss (Psychol. and Brain Sci., Johns Hopkins Univ., Biology-Psych. Bldg. 2123M, College Park, MD 20742, cynthia.moss@gmail.com)

Egyptian fruit bats, *Rousettus aegyptiacus*, present a distinct opportunity to unravel the relative contributions of auditory and visual stimuli for navigation. This species has evolved lingual echolocation, but also uses rod vision to navigate. Echolocation and visual gaze are anchored to the head, but the bat's active control over sonar beam aim, ear, and eye position can influence sampling of sensory space. We investigated the relative weighting of echolocation and vision in free-flying Egyptian fruit bats. In this experiment, we used stereo IR video and microphone array recordings to quantify the bat's trajectory, head direction and sonar beam as it flew to a landing perch.

We introduced sensory conflict by equipping the bats with goggles fit with prisms that shifted visual images by 23 degrees to the left or right. In control trials, prisms were replaced with clear or light-blocking lenses. When bats were presented with conflicting visual and echolocation information, they flew in the direction of the prism shift and often missed the perch. Preliminary analyses of sonar beam aim suggests that their navigation is largely guided by vision. These findings underscore the dominant role of vision in bats that have also evolved auditory specializations to operate in darkness.

Contributed Papers

4:20

1pAB8. Sensing in the swarm: Spectro-temporal variation may facilitate self-recognition of echoes for bats flying in dense groups. Stephen Blackstock (Univ. of Texas at Austin, Austin, TX), Robert Stevenson (Univ. of Notre Dame, Notre Dame, IN), Dieter Vanderelst (Univ. of Cincinnati, Cincinnati, OH), Michael Haberman (Univ. of Texas at Austin, Austin, TX), Paul Domski (Edgewater Tech. Assoc., Oak Ridge, TN), and Laura Kloepper (Biological Sci., Univ. of New Hampshire, 38 Academic Way, Durham, NH 03824, laura.kloepper@unh.edu)

Most biosonar models predict that bats rely on the correct assignment of an echo from a broadcast signal to successfully perceive their environment, which should be difficult for bats in dense groups. Brazilian free-tailed bats (*Tadarida brasiliensis*) form some of the largest aggregations on the planet and have flexible characteristics of their echolocation signals. We hypothesize they use subtle variations in spectro-temporal characteristics to facilitate echolocation in dense groups. To record from inside the swarm, we used both a stationary microphone that opportunistically captured the passing bat swarm, and a custom video and acoustic recorder carried by a trained hawk that flew through the swarm. We computed spectrograms from the acoustic recordings and extracted "time-frequency ridges." These ridges were fit to low-order polynomials to generate model frequency modulation functions, which in turn uniquely determine continuous time-series representations of modeled emission waveforms. Standard signal detection methods (cross correlation and background normalization) enabled quantitative estimation of detection performance, including rejection of interfering emissions and echoes. Our results demonstrate that subtle but specific variation in spectro-temporal shape can constitute the basis of call differentiation, which may be an adaptive strategy used to reject acoustic signals from conspecifics when echolocating in dense swarms.

4:40

1pAB9. The extremes of mammalian hearing: The evolution of whale hearing. Tracey L. Rogers (Ctr. for Marine Sci. and Innovation, UNSW Sydney, E26 Biological Sci. UNSW, Botany Rd., Kensington, New South Wales 2052, Australia, t.rogers@unsw.edu.au), Benjamin J. Walker, and Kobe Martin (Evolution and Ecology Res. Ctr., UNSW Sydney, Sydney, New South Wales, Australia)

The hearing of mammals spans extremes unseen in any other taxa and this diversity is best exemplified by the cetaceans. Conflicting hypotheses explain the evolution of the cetaceans' extreme acoustic biology. Both lower- and higher-frequency hearing limits have been suggested to be ancestral for cetaceans. We investigate this intriguing problem further, through a comparative analysis across 161 extinct and extant mammal species. We show that ancestral whales and mysticetes do not have ultra-low lower frequency hearing limits, and their lower hearing limit is more typical, even slightly higher than that of land mammals of their size (driven by isometric scaling). We show that odontocetes, unlike other mammalian taxa, have reversed the mammalian body-size-hearing relationship. In odontocetes, larger body size is correlated with increased low-frequency hearing limit. We show that aquatically adapted mammals typically have higher lower-frequency hearing limits. Our results suggest that the shift to high frequencies occurred progressively over time. We ask why mysticetes did not evolve higher low-frequency hearing, in line with all other aquatically adapted ears.

Session 1pAO

Acoustical Oceanography and Underwater Acoustics: Acoustic Sensing of the Indian/Southern Ocean

David Barclay, Cochair
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Christine Erbe, Cochair
Centre for Marine Science and Technology, Curtin University, CMST, B301, Kent Street, Bentley, 6102, Australia

Invited Paper

1:00

1pAO1. Revisiting acoustic detections made in the Indian Ocean at the time of the loss of MH370. David R. Dall'Osto (Appl. Phys. Lab., Univ. of Washington, 1013 N 40th St., Seattle, WA 98105, dallosto@apl.washington.edu) and Alec J. Duncan (Ctr. for Marine Sci. and Technol., Curtin Univ., Perth, Western Australia, Australia)

After an expansive multi-year undersea search, the whereabouts of airliner MH370 remains unknown. Given the energetics the crash, and the evidence (e.g., debris) that it crashed somewhere out in the open Indian Ocean, it is possible that MH370 made a detectable noise. Coincidently, as stated in the final Australian Transportation and Safety Board (ATSB) report, there was "An acoustic signature at the time of the final transmission from the engines, but which is at odds with the location determined by the satellite analysis." Following the years-long unsuccessful search, and in light of the acoustic detection of the crash of a Japanese F35 fighter jet in 2019 (received at a similar hydroacoustic station located in the Central Pacific), it seems timely to review the acoustic data to re-assess the likelihood that it is related to the loss of MH370. If the hydroacoustic evidence were included, then the probable crash area would shift significantly to a site roughly 750 km west of the Maldives.

Contributed Papers

1:20

1pAO2. Long-range underwater acoustic detection of aircraft surface impacts—The influence of acoustic propagation conditions and impact parameters. Alec J. Duncan (Ctr. for Marine Sci. and Technol., Curtin Univ., Perth, Western Australia, Australia, A.J.Duncan@curtin.edu.au) and David Dall'Osto (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

The 2019 crash of an F35 fighter aircraft off the east coast of Japan was detected at a range of approximately 3000 km by a hydroacoustic station of the Comprehensive Nuclear Test Ban Treaty Organisation's International Monitoring System. In this paper we compare the acoustic propagation conditions relating to that detection to those relating to the crash of Malaysian Airlines flight MH370 in the Indian Ocean in 2014 with a view to reassessing the likelihood that acoustic signals detected on a similar hydroacoustic station could have been related to the loss of MH370. Possible differences in source levels due to different assumptions about impact parameters are also considered. Despite several extensive searches, the location of the wreckage of MH370 remains unknown.

1:40

1pAO3. Automatic click detector for sperm and beaked whale distribution monitoring. Evgenii Sidenko (Ctr. for Marine Sci. and Technol., Curtin Univ., B301 Hayman Rd., Bentley, Perth, Western Australia 6102, Australia, evgeny.sidenko@curtin.edu.au), Iain M. Parnum, Alexander Gavrilov, and Christine Erbe (Ctr. for Marine Sci. and Technol., Curtin Univ., Perth, Western Australia, Australia)

Passive acoustic monitoring (PAM) is an effective method for monitoring marine fauna. Automatic detection, identification, and quantification of echolocation clicks by toothed whales is commonly computationally inefficient. We present an autoregression and correlation-based algorithm for detection and classification of odontocete clicks. The detection process consists of three steps: (1) detection and extraction of impulsive signals from PAM recordings, (2) cross-correlation of the extracted sounds with template signals, and (3) binary classification of detected clicks. The extraction of impulsive signals from background noise is based on automatic detection of outliers in an autoregression model of noise. The model settings can be

adjusted to detect signals of certain length. The cross-correlation analysis involves the use of template signals of high signal-to-noise ratio, which are manually identified and extracted from PAM data. Using the cross-correlation coefficient as a criterion allows distinguishing echolocation clicks by sperm or beaked whales from other impulsive sounds. Finally, binary classification utilizes thresholds determined through the receiver operating characteristic analysis of an independent subset of manually labeled, extracted signals. The approach was tested for detection of sperm and beaked whales in PAM datasets recorded on the Northwest Shelf of Australia and demonstrated low false positive rates of 3%–5%.

2:00

1pAO4. Humpback whale: Yes/no? Challenges with passive acoustic monitoring of this well-described species. Paul Nguyen Hong Duc (Ctr. for Marine Sci. and Technol., Curtin Univ., GPO Box U1987, Perth, Western Australia 6845, Australia, paul.nguyenhongduc@curtin.edu.au), Christine Erbe, and Shyam Madhusudhana (Ctr. for Marine Sci. and Technol., Curtin Univ., Perth, Western Australia, Australia)

Humpback whales sing song and produce non-song sounds. Dozens of research articles have described their acoustic repertoire—even if only considering Australian waters. While song structure changes slowly over seasons, individual notes may exhibit multi-decadal stability. Irrespectively, automatically detecting humpback whale vocalizations remains an ongoing development activity while manual validation of automated detections continues to be problematic. While analysing archival recordings from Australian waters using the NOAA/Google convolutional neural network pretrained for humpback whale song [https://tfhub.dev/google/humpback$_{whale}$/1], we found numerous instances of a particular type of sound with spectrographic features resembling those of humpback whale vocalizations. These sounds lacked the context of a song and occurred in chorus-like patterns, with the same sound recorded at varying received levels and overlapping in time—indicative of many animals simultaneously producing nothing but this specific call. To investigate these phenomena further, we (1) built a custom detector for these specific calls, (2) ran the detector on archival recordings from North-western Australia, and (3) compared geographic and seasonal presences and absences to known patterns of humpback whale occurrence. The likelihood of these scenarios being humpback whales is discussed.

2:20

1pAO5. Propagation of ship noise in shallow water over a high shear-speed seabed. Cristina Tollefsen (Ctr. for Marine Sci. and Technol., Curtin Univ., GPO Box U1987, Perth, Western Australia 6845, Australia, cristina.tollefsen@curtin.edu.au), Alec J. Duncan, and Iain M. Parnum (Ctr. for Marine Sci. and Technol., Curtin Univ., Bentley, Western Australia, Australia)

Assessing the acoustic impact of shipping close to the coast and within ports and harbours requires an understanding of the propagation of underwater sound in these environments. In the Australian context, this often means propagation over seabeds with shear speeds approaching the in-water sound speed, a situation that results in very different propagation conditions to the low shear speed sediments more commonly encountered in other parts of the world. Acoustic propagation modelling approaches were explored using a dataset of continuous underwater recordings featuring a mix of shipping, industrial, and recreational activities, acquired from 23 Jul to 16 Sep 2022 in Cockburn Sound, off the west coast of Australia. Cockburn Sound is a sheltered marine embayment with a relatively uniform water depth of 17–20 m in the 5 km $\times$ 11 km central portion of the basin, with a thin, variable thickness (0.5 m–5 m) layer of silty sediments overlying a calcarenite basement. The results of modelling the calcarenite basement as an equivalent fluid were compared to those obtained by modelling it as an elastic solid, and with measurements. Subsequent implications of the choice of geoacoustic model will be discussed

Session 1pBA

Biomedical Acoustics and Physical Acoustics: Ultrasound for Biomaterials and Bioprocessing

Muthupandian Ashokkumar, Cochair
University of Melbourne, School of Chemistry, 3010, Melbourne, Australia

Thomas Matula, Cochair
Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105

Leslie Yoo, Cochair
Chemical and Environmental Engineering, Royal Melbourne Inst. of Techology, Melbourne Vctoria 3001, Australia

Chair's Introduction—12:55

Invited Paper

1:00

1pBA1. Sonomechanobiology: High frequency cell mechanostimulation. Lizebona A. Ambattu (Chemical & Environ. Eng., Royal Melbourne Inst. of Technol., Melbourne, Victoria, Australia) and Leslie Yeo (Chemical & Environ. Eng., Royal Melbourne Inst. of Technol., School of Eng., Melbourne, Victoria 3001, Australia, leslie.yeo@rmit.edu.au)

It is well known that cells respond in many different ways to various forms of mechanical cues. To date, however, mechanostimulation has typically been carried out statically or at relatively low frequencies (several Hz), characteristic of the frequencies associated with the motion experienced by cells in their local environment, for example, in the human body (e.g., walking and running). Where higher frequency vibrational mechanotransduction pathways have been investigated, these have primarily been limited to kHz order, and it has generally been suggested that there is no significant advantage in utilising higher frequencies. In a similar manner to observations of new and often nonlinear discoveries when we couple high frequency (10–30 MHz) vibration in the form of surface and hybrid acoustic waves into fluids as well as crystalline materials, we however observe unique and novel phenomena when similar MHz-order vibrational stimuli are transmitted into cells. These include modulation of various ion and piezo channels, and transient, yet reversible, permeabilization of the cell membrane, which have implications for efficient intracellular cytosol delivery, exosome biogenesis, cytoskeletal reorganization and stem cell differentiation, at the same time maintaining very high levels (>95%) in cellular viability.

Contributed Papers

1:20

1pBA2. Mapping acoustic properties of tissues at low temperatures for ultrasound rewarming. Suzi Liang (Univ. College London, Malet Pl. Eng. Bldg., Gower St., London, London WC1E 6BT, United Kingdom, Suzi.liang.21@ucl.ac.uk), Bradley E. Treeby, and Eleanor Martin (Univ. College London, London, United Kingdom)

Cryopreservation of large volumes of cells and tissues, followed by successful rewarming, promises to revolutionize organ transplantation by extending organ preservation times. This technology addresses a critical issue: over 6,000 individuals in the UK are awaiting organ transplants, yet an alarming 60% of donated organs go unused due to current preservation time limitations. A significant challenge lies in providing a rapid and uniform rewarming method for cryopreserved tissues. Ultrasound, capable of converting sound power into heat as it propagates through tissue, offers a promising solution. Nonetheless, before tapping into ultrasound's potential for rewarming experiments, a fundamental understanding of the acoustic properties of tissues, which vary with temperature and during phase change, is essential. This study employs the multiple-reflection method (MRM), with buffer-rod positioned between the transducer and sample, to measure the acoustic attenuation coefficient and sound speed in biological relevant materials and the primary components of tissues, such as water and lipids, across a temperature range of 20 to -100°C. Concurrently, the acoustic properties of commonly used cryopreservation solutions were assessed. These data will facilitate computational simulation of acoustic power delivery and the resulting temperature distributions, enabling planning and monitoring of tissue rewarming, which is critical to its success.

1:40

1pBA3. Ultra-fast mechanics and bioeffects of acoustic droplet vaporization in tissue-mimicking hydrogels. Mitra Aliabouzar (Radiology, Univ. of Michigan, 1301 Catherine St. 6446 Medical Sci. Bldg. I Ann Arbor, MI 48109-5667, aliabouz@umich.edu), Bachir A. Abeid, Jonathan B. Estrada (Mech. Eng., Univ. of Michigan, Ann Arbor, MI), and Mario L. Fabiilli (Radiology, Univ. of Michigan, Ann Arbor, MI)

Acoustic droplet vaporization (ADV) enables phase-shift droplets to respond to focused ultrasound in a spatiotemporal manner, offering a versatile platform for theranostic applications. To better understand the ADV-induced mechanics and resulting bioeffects in real-time, we integrated ultra-high-speed microscopy (up to 10 million frames per second), time-lapse confocal microscopy, and focused ultrasound. Three monodispersed phase-shift droplets—containing perfluoropentane (PFP), perfluorohexane (PFH), or perfluorooctane (PFO)—with an average diameter of $\sim$12 μm were studied in fibrin-based, tissue-mimicking hydrogels. To assess cellular bioeffects and mechanical changes resulting from ADV, we co-encapsulated fibroblasts and tracer particles within the hydrogel. Tracking the displacement of tracer particles during and after ADV indicated a hyper-local region of influence by an ADV-generated bubble, correlating inversely with the bulk boiling point of the phase-shift droplets. Additionally, cell membrane permeabilization significantly depended on the distance between the droplet and cell (d), decreasing rapidly with increasing d. We will discuss and compare ADV dynamics, including maximum radial expansion, expansion velocity, and induced radial strain, as well as the critical distance for cell membrane permeabilization in fibrin-based hydrogels containing three different phase-shift droplets. The findings here provide useful information to optimize ADV for more tailored theranostic applications.

2:00

1pBA4. Microfluidic platform using focused ultrasound passing through hydrophobic meshes towards automatic biological experiment. Yusuke Koroyasu (Graduate School of Comprehensive Human Sci., Univ. of Tsukuba, 1-2 Kasuga, Tsukuba, Ibaraki 305-0821, Japan, koroyu@digitalnature.slis.tsukuba.ac.jp), Ruchi Gupta (School of Chemistry, Univ. of Birmingham, Edgbaston, Birmingham, United Kingdom), Tatsuya Yamamoto (Faculty of Life and Environ. Sci., Univ. of Tsukuba, Tsukuba, Ibaraki, Japan), Yoichi Ochiai (Inst. of Library, Information and Media Sci., Univ. of Tsukuba, Tsukuba, Ibaraki, Japan), Nobuhiko Nomura (Faculty of Life and Environ. Sci., Univ. of Tsukuba, Tsukuba, Ibaraki, Japan), and Tatsuki Fushimi (Inst. of Library, Information and Media Sci., Univ. of Tsukuba, Tsukuba, Ibaraki, Japan)

Laboratory automation is critical in improving productivity and data quality. Droplet-based microfluidic systems offer a solution that enables parallel handling of small samples with high reconfigurability and scalability. The most common technique is electrowetting-on-dielectric (EWOD), which manipulates droplets by exploiting the imbalance of wetting. However, this method is often limited by biofouling. In our previous study, we proposed a novel microfluidic platform using focused ultrasound passing through a hydrophobic mesh, which reduces the contact area and thus the hydrophobic interactions and electrostatic attractions of biomolecules. Our platform demonstrates the manipulation of protein-rich droplets at concentrations up to 1 mg/ml without the need for any additives. This is a significant improvement over existing EWOD methods which are limited to handling samples with protein concentrations as low as 0.005 mg/ml without additives. Here, we further investigate the influence of protein concentration, pH, and buffers using fluorescence microscopy to determine the effect on the system. Our platform also provides basic functions such as moving, merging, and splitting, as well as superior jumping capabilities. With reduced biofouling and the ability to directly transfer droplets to adjacent devices and multi-step processes, our platform opens up new possibilities for the automation of biology.

2:20

1pBA5. Noninvasive material characterization of biomaterials: Measuring viscosity and elasticity using ultrasound. Megan Anderson, Kartik Bulusu, Michael Plesniak, Lijie Grace Zhang (Mech. and Aerosp. Eng., George Washington Univ., Washington, DC), and Kausik Sarkar (Mech. and Aerosp. Eng., George Washington Univ., 800 22nd St. NW, Ste. 3000(MAE), Washington, DC 20052, sarkar@gwu.edu)

The advent of tissue engineering has led to increased interest in the viscoelastic characterization of biomaterials. Gelatin Methacrylate (GelMA) is a particularly promising biomaterial, largely due to its tunability, yet the impact of different preparation parameters on the material's viscoelasticity is not well understood. We characterized an array of GelMA scaffolds, fabricated by varying both GelMA concentration and ultraviolet (UV) light exposure time. Pulse-echo ultrasound techniques were used to noninvasively determine the sound speed and attenuation of the scaffolds, revealing significant dependence on GelMA concentration. Steady shear rate tests in addition to strain- and frequency-controlled oscillatory shear tests were performed using a rotational rheometer (Model: DHR-2, TA Instruments) to ascertain the levels of shear-thinning and viscoelasticity at a wide range of strain rates, oscillation frequencies, and amplitudes. The rheological tests show moduli dependence on both GelMA concentration and curing time. Together, this acoustic and rheological characterization can be used to inform the selection of GelMA scaffolds in tissue engineering applications, and this method can be used as a guide for characterizing other polymeric hydrogels.

Invited Paper

2:40

1pBA6. High throughput system for preparing samples for genomic, epigenetic, transcriptomic, proteomic and metabolomic analysis. Thomas Matula (Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, matula@uw.edu), Karol Bomsztyk (Univ. of Washington, Seattle, WA), and Greg Darlington (Seattle, WA)

Multi-omics considers the analysis of complex biological "omes" (genome, epigenome, transcriptome, proteome, metabolome, and others). This exponentially growing systems biology approach aims to better understand phenotypes in health and disease and as such identify biomarkers of disease and drug targets. The basic workflow is divided into an upstream sample preparation step, and corresponding downstream assays. To prepare samples for these assays biospecimen are treated to extract analytes such as chromatin, DNA, RNA, protein and metabolites. Often these sample preparation steps are done using cavitation. Current tools such as ultrasonic horns, and even many commercial sample preparation instruments, can be highly inconsistent, leading to unreliable assays. In order to overcome the problems of inconsistency, an instrument was designed and built with a transducer array (free field pressures $P+=30$ MPa, $P-=12$ MPa) capable of processing samples directly in standard 96-well microplates. An array of transducers is mounted below a microplate, coupled to a lens array that focuses acoustic energy into each well of a microplate. Intense cavitation is generated in each well. This process results in consistent analytes extraction that leads to consistently reproducible results. The authors have a financial conflict of interest. [Funded by NIH R33CA191135, R21GM111439, R01DK103849, R42HG010855, U01CA246503, and R44GM122097.]

Session 1pCA

Computational Acoustics: Numerical Methods for Vibroacoustics of Underwater Structures

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University of Technology Sydney, UTS, Sydney, 2007, Australia

Paul Dylejko, Cochair
Defence Science and Technology, Canberra, Australia

Chair's Introduction—12:55

Invited Papers

1:00

1pCA1. Vibration response of a finite airfoil under turbulent boundary layer excitation. Paul Williams (Univ. of Technol. Sydney, UTS Tech Lab32/34 Lord St., Botany, New South Wales 2019, Sydney, New South Wales 2019, Australia, paul.williams@uts.edu.au), Mahmoud Karimi (Univ. of Technol. Sydney, Sydney, New South Wales, Australia), and Paul Dylejko (Defence Sci. and Technol. Group, Melbourne, Victoria, Australia)

Turbulent boundary layers excite elastic surfaces leading to material fatigue and noise that may be transmitted into the body of a vehicle or reradiated back into the environment. Three different numerical methods, namely the spatial method, the reciprocity method and the uncorrelated wall plane wave method are employed to predict the vibration response of an airfoil excited by a turbulent boundary layer. These methods will be coupled with semi-empirical models which are used to simulate the wall pressure field beneath the turbulent boundary layer. The first method is performed in the spatial domain while the latter two are performed in the wavenumber domain. In this investigation, the efficacy of each of these will be examined when applied to a finite airfoil. The use of the two calculations will be compared to show any advantages in computational time and memory use.

1:20

1pCA2. Fast method for the determination of radiation-relevant eigenmodes of underwater structures when using FEM shell elements ("modal reduction"). Ralf Burgschweiger (Inst. for Mechatronics, Helmut-Schmidt-Univ. (HSU), Holstenhofweg 85, Hamburg 22043, Germany, burgschr@hsu-hh.de), Ingo Schäfer (Maritime Technol. and Res., Bundeswehr Tech. Ctr. for Ships and Naval Weapons (WTD71), Eckernförde, Germany), Delf Sachau (Inst. for Mechatronics, Helmut-Schmidt-Univ. (HSU), Hamburg, Germany), and Jan Ehrlich (Maritime Technol. and Res., Bundeswehr Tech. Ctr. for Ships and Naval Weapons (WTD71), Eckernförde, Germany)

A self-developed code based on the BEM for the determination of the backscattered sound pressure level in the far field was extended so that additional FEM shell elements and thus elastic material properties can be considered. For this purpose, a separate FEM equation system is built up and directly integrated into the system of the BEM via corresponding transformation matrices. A common variant for solving the FEM-specific parts is to use a solver based on eigenvalue calculations, which provides the corresponding eigenfrequencies and eigenvectors of the uncoupled FEM equation system for a given upper cutoff frequency. It can be observed that only a part of the determined modes has a considerable sound radiation efficiency and is therefore relevant for the result. An algorithm determines these modes by a fast post-processing routine considering a given percentage, reduces the size of the eigenvalue equation system accordingly, and thus shortens the solution time. Furthermore, it was investigated whether the condition of the entire system of equations is improved by removing the irrelevant modes when using iterative solution methods and whether the number of iterations can thus be reduced. The paper describes the basics of FEM coupling with BEM, presents the reduction algorithm used and shows the results obtained for corresponding test structures depending on the reduction percentage used.

Contributed Paper

1:40

1pCA3. Vibration response of a two-dimensional heavy fluid loaded plate with ABH stiffeners. Daniel Martins (Ctr. for Audio, Acoust. and Vib., Univ. of Technol. Sydney, 15 Broadway Ultimo, Sydney, New South Wales 2007, Australia, daniel.martins@student.uts.edu.au), Mahmoud Karimi (Univ. of Technol. Sydney, Sydney, New South Wales, Australia), and Laurent Maxit (Laboratoire Vibrations-Acoustique (LVA), Univ Lyon, Villeurbanne, France)

Acoustic black holes (ABHs) are a promising passive-control approach to mitigate plate vibrations in various engineering applications. By creating a gradient in the flexural waves, an acoustic black hole can trap and dissipate these waves, thereby reducing the amplitude of the vibrations in the plate. The ABH is based on gradually decreasing elastic plate thickness (to trap) and utilizing a damping layer to dissipate the wave energy. This technique has been applied to various structures, and hereby it is proposed to apply it to the stiffeners, commonly used to reinforce structures to increase their stiffness and reduce their susceptibility to vibrations. The studied structure consists of a heavy fluid-loaded infinite plate periodically stiffened and excited by a line force, i.e., a two-dimensional model. In the developed model, the stiffeners are characterized by their flexural and torsional impedances and can be estimated by a finite element analysis. These impedances are then coupled with the analytical formulation of the fluid-loaded plate problem expressed in the wavenumber domain to obtain the spectral displacement. The effectiveness of the ABH-shaped stiffeners in mitigating the plate's vibration is demonstrated by comparing against results from rectangular stiffeners.

Invited Papers

2:00

1pCA4. Making underwater noise predictions using frequency response functions generated by the Operation Round Trip method. Scott Tranter, Robbie Glachan (QinetiQ, Dunfermline, United Kingdom), and Robert Potter (QinetiQ, 2 Aquarius Court, Innova Campus, Dunfermline, Fife KY112DW, United Kingdom, rmpotter@qinetiq.com)

The Round Trip method was derived as a means to acquire frequency response functions at passive locations, where the term 'passive' specifies that no external known force has been applied. This allows frequency response functions to be generated at locations where the degree of freedom cannot be excited by conventional means. Sound and vibration produced by operational sources can increase signal-to-noise ratios in large complex systems and improve the quality of the measured frequency response functions, where it can be challenging to effectively excite the system by conventional experimental means. In this study, the application of a modified version of the Round Trip method that is based on transmissibility functions, termed the Operational Round Trip method, has been investigated to determine whether the forces generated by an operational source, such as an offshore wind turbine, could be utilized to predict the frequency response functions at and between passive measurement positions within a coupled mechano-acoustical system. These predicted frequency response functions have been used with the blocked forces of a vibratory source to make underwater noise predictions at remote positions, which were validated against measured underwater noise measurements.

2:20

1pCA5. Reduced order modelling of thickness effects on a cantilevered hydrofoil in homogeneous isotropic turbulence. Konstantinos Tsigklifis (Platforms Div., DSTG, 506 Lorimer St., Melbourne, Victoria 3207, Australia, kostastsigklifis@gmail.com), Paul Dylejko (Platforms Div., DSTG, Melbourne, Victoria, Australia), Mahmoud Karimi (Univ. of Technol. Sydney, Sydney, New South Wales, Australia), Marcus Wong, and Alex Skvortsov (Platforms Div., DSTG, Melbourne, Victoria, Australia)

The response of a cantilevered hydrofoil excited by honeycomb-generated turbulence is studied with a reduced order analytical model that considers the thickness effects of NACA sections. This is achieved by extending a gust response model of the unsteady lift to include realistic 2D sections instead of using a flat plate assumption. The general chordwise thickness profiles are included in the model using the generalized Joukowski transformation. The problem is cast into a constrained optimization problem which is solved with the use of the Lagrange multiplier method. The frequency response function (FRF) of NACA sections with the same maximum thickness to chord ratio but with different leading-edge thickness profiles are calculated to investigate the effect of the hydrofoil leading edge thickness on the high frequency response. This FRF is combined with the Uncorrelated Wall Plane Wave (UWPW) technique to simulate the pressure jump amplitude on the hydrofoil due to the turbulence interaction which excites the hydrofoil in heaving and pitching motion according to Theodorsen's hydroelastic theory. Finaly, the structural velocity spectra are compared with available experimental results of the turbulence ingestion and the effect of the different leading edge thickness profiles on the vibration response is demonstrated for various NACA sections.

2:40–3:00 Break

Contributed Papers

3:00

1pCA6. Acoustic radiation from a baffled finite shell in an underwater waveguide. Jamie Kha (Ctr. for Audio, Acoust. and Vib., Univ. of Technol. Sydney, 32/34 Lord St., Botany, New South Wales 2019, Australia, jamie.kha@student.uts.edu.au), Mahmoud Karimi (Ctr. for Audio, Acoust. and Vib., Univ. of Technol. Sydney, Sydney, New South Wales, Australia), and Laurent Maxit (Laboratoire Vibrations-Acoustique (LVA), INSA-Lyon, Villeurbanne, France)

Analytical modelling of vibroacoustic systems can help us to understand the physical phenomena involved in more complex problems, and it provides a benchmark solution and reference upon which more complex systems can be built. In the present work, the system of interest consists of a finite elastic cylindrical shell inserted in infinitely rigid cylindrical baffles and immersed in an underwater acoustic waveguide. The latter consists of a finite fluid layer bounded by an upper free surface and a lower rigid floor. In such a fluid domain, the acoustic waves radiated from the excited shell will exhibit reflections off the boundaries. This phenomenon is modelled by the image-source theory and embedded in the fluid loading term, which intervenes in the shell equations. Investigations into the influence on the finiteness of the elastic shell, types of supports (i.e., simply supported, clamped, free, and combinations of these), and depth of the waveguide on the shell's acoustic radiation are presented.

3:20

1pCA7. Target echo strength of small structured or resonant objects in water. Arne Stoltenberg (Ctr. for Water/Structure/Air Borne Sound, Bundeswehr Tech. Ctr. for Ships and Naval Weapons, Naval Technol. and Res. (WTD 71), Klausdorfer Weg 2-24, Kiel, Schleswig-Holstein 24148, Germany, ArneStoltenberg@Bundeswehr.org) and Ingo Schäfer (Numerical Modelling, Bundeswehr Tech. Ctr. for Ships and Naval Weapons, Naval Technol. and Res. (WTD 71), Eckernförde, Germany)

The Target Echo Strength (TES) of an object in water is a quantity for its ability to reflect sound waves. The TES value depends strongly on the directions of the incoming and the observed reflected wave, the structure of the object and of the frequency. In addition, resonant effects can increase its value significantly. These different effects were investigated in several TES reflection experiments in a water tank with varying small test objects of different material, scale and purpose. Devices under test were: a scaled massive steel structure with a thin hull, resonant PMMA balls and others. The experimental results are compared with the results of the BEM/FEM numerical modelling of each body. Limits of the concurrence are shown. Combinations of the effects might occur and increase the TES value beyond expectation.

Invited Paper

3:40

1pCA8. Vibration control of a cantilever plate immersed in water using shunted piezoelectric patches. Huong Cao (Ctr. for Audio, Acoust. and Vib., Univ. of Technol. Sydney, 15 Broadway, Ultimo, Sydney, New South Wales 2007, Australia, huong.q.cao@student.uts.edu.au), Mahmoud Karimi, Paul Williams (Ctr. for Audio, Acoust. and Vib., Univ. of Technol. Sydney, Sydney, New South Wales, Australia), and Paul Dylejko (Platforms, Defence Sci. and Technol. Group, Melbourne, Victoria, Australia)

Minimizing structural vibrations is an important engineering requirement in many applications. This study theoretically investigates the vibration control of an in-water cantilever plate using piezoelectric patches and resonant shunt circuits (including a resistance R and an inductance L in series or parallel). To do this, an analytical model for predicting the point-forced vibration response of a fluid-loaded cantilever plate with two piezoelectric patches is developed. The electromechanical coupling factor and optimum values for R and L are then estimated by considering the dynamics of the system in short-circuit and open-circuit conditions. The effect of piezoelectric patch size, location and electromechanical properties on the vibration response of the plate is also investigated through numerical examples. The results from this study show that optimized piezoelectric patches could provide a significant vibration reduction for underwater applications.

Contributed Paper

4:00

1pCA9. Estimation of the low-wavenumber component of turbulent boundary layer pressure field using vibration data. Hesam Abtahi (Ctr. for Audio, Acoust. and Vib., Univ. of Technol. Sydney, 13/11 Henderson Rd., Alexandria, Sydney, New South Wales 2015, Australia, Seyedhesamaldin.abtahi@student.uts.edu.au), Mahmoud Karimi (Ctr. for Audio, Acoust. and Vib., Univ. of Technol. Sydney, Sydney, New South Wales, Australia), and Laurent Maxit (INSA-Lyon, Laboratoire Vibrations-Acoustique (LVA), 25 bis, av. Jean Capelle, F-69621, Villeurbanne Cedex, Univ Lyon, Villeurbanne, France)

In structures subjected to a high-speed flow, the convective region of the wall pressure field (WPF) beneath a turbulent boundary layer (TBL) plays a crucial role in their vibration behaviour. However, when it comes to underwater structures experiencing low-speed flow, they effectively filter out the convective domain of WPF and the bending wavenumber of structures align with the low-wavenumber domain of the WPF. As a result, the primary cause of vibration in this case is the low-wavenumber components of the WPF. Thus, accurate estimation of the WPF at low-wavenumber domain is crucial for predicting vibration responses of these structures. Existing models for WPF accurately predict the convective region but differ significantly in predicting the low-wavenumber levels. This numerical study aims to investigate the feasibility of estimating the low-wavenumber WPF by analysing measured vibration data from a flat plate excited by a TBL. The WPF's cross spectrum in the wavenumber domain can be linked to the cross spectrum of the plate's acceleration. By employing regularization techniques and solving an inverse problem, the low-wavenumber components of the WPF can be then estimated. Virtual experiments are performed to evaluate the accuracy of the studied process by comparing its prediction to the input WPF model.

Session 1pEA

Engineering Acoustics, Physical Acoustics and Structural Acoustics and Vibration: Acoustics in Spatiotemporally Varying Materials

Michael Haberman, Cochair
Walker Department of Mechanical Engineering, The University of Texas at Austin, 2202 Garden Street, Unit A, Austin, TX 78702

Nicole Kessissoglou, Cochair
School of Mechanical and Manufacturing Engineering, UNSW Sydney, UNSW Sydney, School of Mechanical and Manufacturing Engineering, Kensington, 2052, Australia

Chair's Introduction—12:55

Invited Papers

1:00

1pEA1. Spatiotemporal modulations in acoustics: From nonreciprocity to mechanical computing. Mohammadreza Moghaddaszadeh, Mohamed Mousa, Revant Adlakha, Mohamed Ali Attarzadeh, Amjad Aref (Univ. at Buffalo (SUNY), Buffalo, NY), and Mostafa Nouh (Univ. at Buffalo (SUNY), 240 Bell Hall, Attn: Mostafa Nouh, Buffalo, NY 14260, mnouh@buffalo.edu)

Spatially graded materials have been a cornerstone of acoustic metamaterials for decades, enabling wave manipulation across different length scales. Likewise, structures with time-varying material properties have gained traction in wave filtering applications. Put together, systems exhibiting concurrent spatial and temporal modulations of one or more parameters (e.g., stiffness or phase) can unlock an array of new features ranging from nonreciprocal propagation to frequency-dependent wave beaming. In this talk, we will introduce the notion of dynamic phase gradients, i.e., a spatial phase shift between neighboring elements (of a metasurface) or transducers (of a phased array) which also varies in time. Through theory and experiments, we will demonstrate that the resultant systems are capable of (a) generating multiple scattered harmonics of a single input which simultaneously propagate in different directional lanes, each carrying a unique frequency footprint, and (b) exhibiting non-identical beaming patterns in transmission and reception by breaking time invariance. We will present an application of this concept in the development of a mechanical neural network via reconfigurable elastic metasurfaces, designed to perform a computational task. Since the frequency channels of a spatiotemporally-modulated metasurface are independently tunable, they can be assigned distinct tasks, thus allowing parallel operations in mechanical computing systems.

1:20

1pEA2. Nonreciprocal phase shifts in spatiotemporally varying materials. Jiuda Wu (Concordia Univ., Montreal, QC, Canada) and Behrooz Yousefzadeh (Concordia Univ., Montreal, 1455 De Maisonneuve Blvd. W., Rm. EV-4.139, Montreal, QC H3G 1M8, Canada, behrooz.yousefzadeh@concordia.ca)

Materials with spatiotemporally varying properties exhibit nonreciprocal wave propagation characteristics. Nonreciprocity is predominantly identified by a left-to-right (L-R) transmission amplitude that is different from the right-to-left (R-L) transmission amplitude; a significant difference in the transmitted amplitudes is often desirable in this context. We review nonreciprocal vibration transmission in discrete mechanical systems with spatiotemporally modulated elasticity. We discuss the importance of the transmitted phase in this context, specifically that the difference between the L-R and R-L transmitted phases can be the main contributor to breaking of reciprocity in short systems. We show that the formulation of the problem in terms of response envelopes provides a computationally efficient path for exploring the steady-state nonreciprocal transmission characteristics of spatiotemporally modulated materials. In particular, we use this technique to identify response regimes that are characterized by a nonreciprocal phase shift in transmitted vibrations while maintaining equal transmitted amplitudes or energies.

1:40

1pEA3. Physical learning of stiffness tensors by self-activated solids. Yangyang Chen (Mech. and Aerosp. Eng., HKUST, Rm. 2555, HKUST, Hong Kong, Hong Kong, maeychen@ust.hk) and Yuxuan Tang (Mech. and Aerosp. Eng., HKUST, Hong Kong, Hong Kong)

Unlike plastic deformations, elastic deformations usually cannot change mechanical properties of materials under different loading conditions. In this talk, we suggest a class of self-activated solids and derive a local learning rule so that the solid can learn the desired stiffness tensors through repeated elastic deformations it experienced. The local learning rule is inspired by contrastive Hebbian learning, which guides the learning process by modulating the bond stiffness of the solid based on its local stain. Extensive numerical tests are performed to validate the design and learning rule. We show that the self-activated solid can be physically trained to display the

desired bulk, shear, and coupling moduli and to manifest uni-mode and bi-mode materials. The solid can also realize time-varying moduli to function as time-modulated materials. The material design and learning strategy introduced are scalable and applicable to arbitrary lattice geometries and other discrete material systems. The study could lead to new physical learning systems, beneficial to autonomous materials, machines, and robots.

Contributed Papers

2:00

1pEA4. Acoustic scattering from spatiotemporally modulated domains. Benjamin M. Goldsberry, Samuel P. Wallen (Appl. Res. Labs. at The Univ. of Texas at Austin, Austin, TX), and Michael Haberman (Walker Dept. of Mech. Eng., The Univ. of Texas at Austin204 E Dean Keeton St., Austin, TX78712, haberman@utexas.edu)

Acoustic and elastic metamaterials with space- and time-dependent material properties have received significant attention recently as a means to realize systems that induce nonreciprocal wave propagation in the bulk or enable frequency and mode conversion of fields scattered from metasurfaces. A previous study derived a nonreciprocal Green's function for flexural waves in a one- and two-dimensional finite plate with a spatiotemporal modulation of the Young's modulus [Goldsberry *et al.*, *J. Acoust. Soc. Am.* **150**(4), 2021]. In this work, we generalize the nonreciprocal Green's function approach to the acoustic case of scattering from an inhomogeneity whose properties are general functions of space and time. Canonical geometries where the analytical solution is known for the unmodulated case will be investigated. Computations of the scattered field directivity pattern will then be carried out as a function of the modulation parameters to determine cases that yield a large degree of control over the scattered field directivity pattern for each generated frequency harmonic. [Work supported by ONR.]

2:20

1pEA5. Nonreciprocal vibrations of discretized finite elastic structures with spatiotemporally modulated material properties. Nathan Geib (Appl. Res. Labs. at The Univ. of Texas at Austin, 1587 Beal Ave. Apt 13, Ann Arbor, MI 48105, geib@umich.edu), Benjamin M. Goldsberry, Samuel P. Wallen, Christina Naify, and Michael R. Haberman (Appl. Res. Labs. at The Univ. of Texas at Austin, Austin, TX)

Elastic waveguides with time- and space-dependent material properties have received great attention as a means to realize nonreciprocal propagation of small-amplitude mechanical waves in unbounded elastic media. Previous works have shown that propagating waves in a modulated medium violate reciprocity by means of asymmetric frequency and wave number conversion between two counterpropagating modes. We previously investigated nonreciprocal longitudinal and transverse vibrations in a finite elastic waveguide with time- and space-dependent material properties using a semi-analytical coupled-mode theory The model employs the mode shapes of the non-modulated beam as the orthogonal basis functions to describe the nonreciprocal vibrations of beams with spatiotemporally modulated properties. The present study extends previous work to consider vibrations of Euler beams with spatially discretized stiffness that are modulated in time and phased relative to each other to mimic spatiotemporal modulation. This is achieved by approximating the discretized stiffness and then employing the mode shapes of the beam with unmodulated discretized stiffness to predict the nonreciprocal vibrational response of the beam when subjected to spatiotemporal modulation. We perform a parametric study of the system parameters and discuss how this model can be used to define requirements of future systems that can be used for experimental demonstration.

2:40–3:00 Break

3:00

1pEA6. On space-time media that compute their own inverse. Dirk-Jan van Manen (Geophys., ETH Zurich, Sonneggstrasse 5, Zürich 8092, Switzerland, dirkjan.vanmanen@erdw.ethz.ch), Johannes Aichele, Jonas Müller (Geophys., ETH Zurich, Zurich, Switzerland), and Marc Serra-Garcia (Hypersmart Matter, AMOLF, Zurich, Switzerland)

We show that it is possible to design a space-time medium such that the time scattering anticipates the space scattering in the medium and produces the exact inverse for the space scattering. Probing such a medium with a single broadband pulse thus also results in a single broadband pulse on the other side of the space-time heterogeneity. The transmission coefficient of such systems is unity. The reflection coefficient is non-zero, however, as both the time and spatial boundaries scatter waves in the backward direction. The time boundaries thus add energy. To construct a single pulse on the other side of a stack bounded by n interfaces, $2(n-1)-1$ additional well-scaled and well-timed pulses need to be emitted following the main pulse, to cancel multiple reflections in the forward direction. Scattering at a time boundary doubles the number of wavepackets propagating in a medium. Furthermore, time refraction and reflection coefficients have the form required to produce well-scaled pulses. The required $2(n-1)-1$ pulses can thus be efficiently generated by a single pulse interacting with n time boundaries. The contrasts across the time boundaries are the spatial contrasts in reverse. The time "thicknesses" of the boundaries are a simple function of the layer velocities.

3:20

1pEA7. A theoretical study on altering the frequency of high-order modes using a time-varying wall property. Bohua Huang (Dept. of Mech. Eng., The Univ. of Hong Kong, Pokfulam Rd., Pokfulam Hong Kong, Hong Kong SAR 999077, China, u3010319@connect.hku.hk), Tianyu Xu (Dept. of Mech. Eng., The Univ. of Hong Kong, Pokfulam Rd., Hong Kong, China), Bin Dong (High Speed Aerodynamics Inst., China Aerodynamics Res. and Development Ctr., No. 6, South Section, Second Ring Rd., Mianyang, Sichuan, China), and Lixi Huang (Dept. of Mech. Eng., The Univ. of Hong Kong, Pokfulam Rd., Hong Kong, China)

Noise radiation by aero-engines features high circumferential modes and high frequencies. It is rather difficult to suppress such high-order modes near their cut-on frequencies. Delicate designs are needed. When the engine operation conditions change, a design good for one set of parameters may not be optimal for another. This study explores a new path: an electromagnetic diaphragm with a MOSFET-controlled shunt circuit can be used to convert the wave frequency to any value at will by creating a digitally predefined time-varying material property. When properly organized, the incident sound energy is dispersed to several frequencies, some of which are below the cutoff frequency. In this way, it is showed that part of the energy of high-order modes can be reduced. This method creatively utilizes electroacoustic coupling to alter frequencies and control the propagation of high-order modes. Additionally, the time-variation is made on a passive basis in the sense that it requires no sensor input and no energy input to the actuator, thus ensuring stability and potential application in engineering.

3:40

1pEA8. Acoustic black hole designs for circular beams under axial loading. Shahrokh Sepehrirahnama (Univ. of Technol. Sydney, 32-34 Lord St., Techlab, Botany, New South Wales 2019, Australia, shahrokh.sepehrirahnama@uts.edu.au), Sebastian Oberst (Ctr. for Audio, Acoust. and Vib., Univ. of Technol. Sydney, Ultimo, New South Wales, Australia), and Joseph C. S. Lai (Univ. of New South Wales, Canberra, Australian Capital Territory, Australia)

Acoustic Black Hole (ABH), as a non-reflecting wave effect, has been realised in beams and plates by locally changing their thickness. Geometrical ABH designs stem from their transverse-load bearing properties, assuming isotropic material properties associated with engineering materials (e.g., steels). The underlying physics is to manipulate stiffness locally for a transversely-loaded structure, leading to a change in the elastic wave speed. However, the same ABH would have limitations for axial loading, i.e., slender beams for which longitudinal waves dominate. Here, we present a wave propagation approach using wavefront tracking to identify a potential ABH for axially-loaded circular beams. We study wave propagation in three exponential designs of finite length, monitoring the wave-travel time. Indicative of an effective ABH in finite length sections, the wave-travel time increases compared to the case of a beam without ABH. By employing the wave front tracking method for the design of an ABH with axial loading, it is possible to verify the effectiveness of ABHs. Also, various material models, e.g., orthotropic materials such as wood, and different loading conditions can be considered, which opens a new avenue in applications of ABH phenomena beyond conventional vibro-acoustic control problems.

4:00

1pEA9. Interesting effects of harmonically scattered waves from 1D functionally graded nonlinear inclusions. Pravinkumar Ghodake (Mech. Eng., Indian Inst. of Technol. Bombay, B423, Hostel 14, IIT Bombay, Powai, Mumbai, Maharashtra 400076, India, mech7pkumar@gmail.com)

The interaction of monochromatic elastic waves with nonlinear inclusions shows the presence of harmonically scattered waves. Challenges in obtaining theoretical solutions and limitations of conducting experiments to understand harmonic scattering of the nonlinear waves from functionally graded materials motivate us to conduct numerical experiments. Harmonic scattering of the elastic waves from highly local functionally graded nonlinear inclusions is exploited in this study to propose a novel building block of new nonlinear metamaterials that can effectively control harmonic responses. The simple, commonly observed, unique spatial distribution of the nonlinear parameters is proposed so that the area under the distribution remains the same for a constant-sized inclusion. Despite different spatial distributions, these functionally graded distributions show the same harmonic responses of both forward and backscattered waves during the interaction of monochromatic waves and one-way two-wave mixing. The amplitudes of all possible combinations of harmonics remain the same as long as the area under the spatially distributed nonlinear parameters curve, irrespective of the distribution curves. Reducing the area under the curve of the functionally graded nonlinear inclusion simply by reducing the maximum value of the functionally distributed nonlinear parameters, a decrease in the amplitudes of the harmonics is demonstrated.

Session 1pED

Education in Acoustics: General Topics in Acoustics Education

Jack Dostal, Chair
Physics, Wake Forest University, Olin Physical Laboratory, 1834 Wake Forest Road, Winston-Salem, NC 27109

Contributed Papers

1:00

1pED1. A focus on ocean acoustics education. Gail Scowcroft (Oceanogr., Univ. of Rhode Island, 19 Prospect Ave., Narragansett, RI 02882, gailscow@uri.edu), Kathleen Vigness Raposa (Inspire Environ., Newport, RI), Christopher Knowlton, Holly Morin (Oceanogr., Univ. of Rhode Island, Narragansett, RI), and Liesl Hotaling (Eidos Education, Highlands, NJ)

The clear need for ocean acoustics education led to two initiatives in 2023. In January, the U.S. National Academy of Sciences (NAS) formed a committee to research the state of ocean acoustics education in the country and determine needs and opportunities for the discipline. In May, the theme of the fifth Global Ocean Science Education (GOSE) Workshop was ocean acoustics. The workshop brought together 70 participants from a dozen countries. These delegates, representing the research, business, policy, and education sectors, shared their knowledge on a suite of key topics in underwater acoustics and education. Delegates discussed the importance of integrating ocean acoustics into ocean education and ocean literacy initiatives, workforce development through an ocean acoustics microcredentials program, and addressing the needs of the global ocean acoustics community. The needs identified during the NAS study and the GOSE Workshop include the imperative to raise awareness of careers in ocean acoustics in precollege education, as well as undergraduate education, more opportunities for training at the technician level, and easy to access educational resources, such as those found on the *Discovery of Sound in the Sea* website (www.dosits.org). Recommendations from the NAS study and the GOSE Workshop will be shared during this presentation.

1:20

1pED2. A case study on generative learning approaches in a studio and flipped class-room setting for increased learning outcomes. Sebastian Oberst (Ctr. for Audio, Acoust. and Vib., Univ. of Technol. Sydney, 123 Broadway, Ultimo, New South Wales 2007, Australia, sebastian.oberst@uts.edu.au) and Shahrokh Sepehrirahnama (Univ. of Technol. Sydney, Botany, New South Wales, Australia)

Teaching and learning during Covid-19 have been strongly affected by lockdowns, isolated online learning, and the sudden requirement to alternatively assess students while considering the effect internet-based information sources. Here, we present outcomes on the learning outcome of students returning from distance learning into the face-to-face mode, studying the subject "Embedded Mechatronic Systems" employing increasingly methods of Generative Learning Theory (GLT) in a flipped classroom environment, using studio and project-based learning approaches. By introducing a group project component, the formerly disconnected laboratory components become strongly connected with students being exposed to practical aspects and teamwork, generating reflected reports and videos of their practical work. To overcome the effects of Covi-d19, tighter assessments, and in-person engagements is emphasised. Viva-voces have been introduced and AI invigilated final exams have been altered to in-class room quizzes, while monitoring the cohort's performance over 3 sessions. Our data indicate that face-to-face learning and hands-on practice with peers using self-testing and self-explaining strategies enacts higher outcomes, opposed to remote modes of teaching. Our results exemplify on how to move back into face-to-face teaching with future steps to increase learning outcomes using the flipped classroom, GLT, and a studio setting being discussed.

1:40

1pED3. Advancing acoustical analysis: A suite of acoustic research-level measurement tools for professional and clinical applications. John Holik (Speech Pathol., Univ. of Sydney, Susan Wakil Health Bldg., D18 Western Ave., Camperdown, New South Wales 2006, Australia, john.holik@sydney.edu.au), Tuende Szalay (Speech Pathol., Univ. of Sydney, Camperdown, New South Wales, Australia), Duy D. Nguyen (Speech Pathol., Univ. of Sydney, Sydney, New South Wales, Australia), and Catherine Madill (Speech Pathol., Univ. of Sydney, Camperdown, New South Wales, Australia)

This presentation showcases the work of our research group in developing a suite of research-level acoustic measurement tools aimed at bridging the gap between cutting-edge research and practical applications clinical settings. Over the last four years over 100 Speech language Pathologists and Ear, Nose and Throat surgeons internationally have completed our course "Using acoustic analysis to assess and treat voice disorders in clinical and research settings." This course has focused on advanced acoustic measurement techniques with a freeware (PRAAT) for specific application to voice disorder. The online course offers a one-stop shop for clinicians, providing access to state-of-the-art acoustic quantification methods that were previously restricted to the research domain. Detailed manuals and video walkthroughs ensure users can confidently navigate the measurement process and produce reliable acoustic data for clinical use. We are now planning to expand the range of courses in clinical and research-level acoustical analysis using other freeware applications and applications in other disciplines such as speech, accent and articulation analysis. This presentation will highlight the features and capabilities of our planned suite of research-level acoustic measurement tools, emphasising the translational impact they can have on enhancing acoustical analysis in various fields.

2:00

1pED4. My favorite resources for teaching musical acoustics. Jack Dostal (Phys., Wake Forest Univ., Olin Physical Lab., 1834 Wake Forest Rd., Winston-Salem, NC 27109, dostalja@wfu.edu)

I teach a Physics of Music class at Wake Forest University. Students from a broad range of majors and backgrounds take the class to fulfill a science divisional requirement for graduation. In this talk I will describe how I incorporate some of my favorite online resources into my instruction. Some of these include the Sound and Waves section of the Physclips web platform created by Joe Wolfe at the University of New South Wales. I also frequently draw from Dan Russell's Acoustics and Vibration Animations at Penn State University. In addition, I find that tools and resources such as online spectrograms and live-streamed music fit well within my course.

2:20–2:40 Break

2:40

1pED5. Leveraging audio and music experience in a first-year engineering sequence. 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) and Benjamin Thompson (Elec. and Comput. Eng., Univ. of Rochester, Rochester, NY)

Recently, the introductory courses in Audio and Music Engineering (AME) at the University of Rochester have succeeded in attracting an academically diverse group of students, such that engineering majors often comprise a minority of those enrolled. Therefore, we see our mandate as balancing the need to prepare AME majors for subsequent courses while simultaneously engaging and supporting non-majors. To meet this challenge, we leverage students' existing experiences with sound and music to introduce foundational concepts in acoustics, circuits, and programming in a project-based environment. In the fall, students complete a series of labs where they build and conduct experiments on sections of a guitar amplifier. Each lab emphasizes a key electronics concept, culminating in a completed amplifier that students can keep. Similarly, the spring semester centers around a series of MATLAB assignments, culminating in a short-time Fourier transform project where students investigate the properties of reverberant speech. In both classes, lectures are designed to provide students with the theoretical tools necessary to fully engage with these projects. In this presentation, we share a specific approach that has been successful for us in engaging non-majors while supporting majors in STEM courses and plan to provide associated open curriculum materials.

3:00

1pED6. An acoustic design project for final year engineering students using resonant arrays. Andrew Hall (Mech. and Mechatronics Eng., Univ. of Auckland, Auckland, New Zealand, a.hall@auckland.ac.nz), Vladislav Sorokin, George Dodd, and Gian Schmid (Mech. and Mechatronics Eng., Univ. of Auckland, Auckland, New Zealand)

This paper reviews the creation of a design course for mechanical engineering students at the University of Auckland. The primary objective of the course is to introduce students to the field of acoustics, teaching basic theoretical and experimental acoustics principles, together with product design and fabrication methods, in an engaging and educational manner. During the course, students were assigned the task of developing a noise reduction duct jacket that reduced sound propagation whilst allowing free airflow, all within specified dimensions. Students were required to create designs that attenuated a predefined sound spectrum, with both narrow-band and wide-band components. Over six weeks students developed their designs using mathematical modelling and fabricated their prototypes. The performance of their suppressors was assessed through sound pressure level and impedance tube transmission loss measurements. Students then modified their designs to maximise performance before final submission. Four-person teams produced a diverse range of implementations. The most successful designs achieved impressive performance with peak transmission loss up to 70dB, 2kHz bandwidths of 10 + dB transmission loss and a delta dBA reduction of 21.5 dB. Student feedback indicates a high level of satisfaction with the course, highlighting its effectiveness in imparting key knowledge and skills in acoustics engineering.

3:20

1pED7. Undergraduate engineering education with hands-on experimental acoustic field work. Jeff Foeller (Eng., East Carolina Univ., 1000 E. 10th St., Greenville, NC 27858, foellerj@ecu.edu), Andrea Vecchiotti (Eng., East Carolina Univ., Wanchese, NC), Diego Turo, Joseph Vignola (Mech. Eng., The Catholic Univ. of America, Washington, DC), and Teresa J. Ryan (Eng., East Carolina Univ., Greenville, NC)

Offering undergraduate students access to technician-level work experience early in their academic career has successfully promoted subsequent interest in and pursuit of independent inquiry in acoustics. Over the past 9 years, approximately 70 undergraduate students have been affiliated with the East Carolina University Department of Engineering's acoustics and vibrations laboratory. Of those 70, 30 have pursued independent inquiry, 11 have attended a professional meeting, 7 have presented at a professional meeting, and 7 have pursued additional study or secured positions in a relevant field. The student work has spanned several topical areas in acoustics and vibrations. The primary effort has been collecting synchronized atmospheric and acoustic data over long ranges in littoral environments. This work presents the best practices the investigators have cultivated for training, recruiting, and retaining quality undergraduate research assistants. Both paid research positions and unpaid work done for course credit are included in the discussion.

3:40

1pED8. Flying disc noise as an introduction to acoustic signal processing. Kyle S. Dalton (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, ksd5377@psu.edu)

Flying disc sports such as ultimate frisbee and disc golf continue to grow in popularity around the world. Despite the ubiquity of the flying disc in leisure and sport, descriptions of disc flight are largely heuristic and anecdotal, with minimal quantitative investigation into in-flight behavior. This work has two primary goals. First, by throwing a variety of discs over an array of microphones, we will assess how in-flight properties of the disc impact the recorded acoustic signals. Remote acoustic sensing offers several advantages over previous flying disc experiments: No instrumentation is placed on the disc and the disc can be observed over a longer flight than is possible with most visual measurement techniques. Second, we intend to create a low-cost, easy-to-implement experiment that introduces students to acoustic signal processing topics related to rotating machinery and array theory. Frisbees are cheap, readily available, and familiar to many students, making the flying disc a fun and accessible sound source that students can engage with from data collection to data analysis. This presentation will discuss the experimental setup, provide initial results, and offer lessons learned for educators and disc sport enthusiasts attempting similar projects.

Session 1pNSa

Noise, Engineering Acoustics, Physical Acoustics, and ASA Committee on Standards: Measurement of Low-Frequency Sound and Standards

Walter Montano, Cochair
Technical, ARQUICUST, Luis Clavarino 1227, Gualeguaychú, E2820, Argentina

David S. Woolworth, Cochair
Roland, Woolworth & Associates, 356 CR 102, Oxford, MS 38655

Norm Broner, Cochair
Broner Consulling Pty Ltd, 159 Victoria Parade, Melborne, Australia

Chair's Introduction—12:55

Invited Paper

1:00

1pNSa1. Review and comments on ANSI-ASA S12.9 part 7 measurement of low frequency and infrasound. David S. Woolworth (Roland, Woolworth & Assoc., 356 CR 102, Oxford, MS 38655, dave@oxfordacoustics.com) and Roger Waxler (Infrasound, National Ctr. for Physical Acoust., University, MS)

ANSI-ASA S12.9 Quantities and Procedures for Description of Environmental Sound, Part 7: Measurement of Low Frequency Noise and Infrasound Outdoors and in the presence of Wind and Indoors in Occupied Spaces is currently being evaluated for potential revisions. This presentation will cover proposed revisions to the standard and reasoning used to arrive at these suggested revisions.

Contributed Paper

1:20

1pNSa2. Predicting low-frequency noise using its own bandwidth. Walter Montano (Tech., ARQUICUST, Luis Clavarino 1227, Gualeguaychú, Entre Ríos E2820, Argentina, wmontano@arquicust.com) and Elena I. Gushiken (Manager, ARQUICUST, Lima, Lima, Peru)

Soon after A-weighing began, an article of 1938 mentioned that as a metric it didn't represent true low-frequency loudness. In 1952, a method for "guessing" low-frequency energy was proposed and later, Beranek in 1954 recommended that if the dBC level is greater than dBA, an analysis should be performed below 150 Hz. In 1969, Botsford introduced a criterion for predicting the low-frequency level by [(dBC-dBA) > 10 dB], using the entire audible spectrum, so that this condition applies only if the spectrum is "balanced." The German Standard DIN 45680 of 1997 proposed a similar criterion, limiting this calculation to one-third octave band frequencies from 8 to 100 Hz. Since 1999, the WHO recommends using the Botsford criterion, but warns that it is not valid if the spectrum contains tones or if it is unbalanced. This paper will demonstrate that the correct way to analyze the low-frequency sound levels is to use its own bandwidth, that to have a traceable noise descriptor the ISO low-frequency definition must be used, it means to use the one-third octave band frequencies between 16 and 200 Hz, to calculate the difference (dBC-dBA) in this range to predict its actual low-frequency sound level.

Invited Paper

1:40

1pNSa3. Investigating vibration-based methods of measuring impact noise insulation to improve reproducibility at low frequencies. Sunit Girdhar (Paul S Veneklasen Res. Foundation, 1711 16th St., Santa Monica, CA 90404, sgirdhar@veneklasen.com), Benjamin M. Shafer (PABCO Gypsum, Tacoma, CA), Wayland Dong, and John LoVerde (Paul S Veneklasen Res. Foundation, Santa Monica, CA)

Standard impact noise insulation testing is based on measuring the average sound level in a reverberant field in order to estimate the incident and/or radiated sound power. While this method is straightforward, the variation under reproducibility conditions is significant, and a significant portion of the uncertainty is due to the complications of measuring the sound field in the receiving space. This variation is largest at low frequencies, where the effects of spatial variation and modes in the receiving room are well documented. Measuring the vibration on the surfaces of an assembly, or measuring the vibration propagation through an assembly, may provide another measure of the impact insulation of the assembly that is not affected by the sources of uncertainty associated with measuring the sound field in a room. This paper reports on the results of preliminary investigations on floor assemblies *in situ* while excited by impact sources and the comparison with conventional receiving room sound pressure measurements.

Contributed Paper

2:00

1pNSa4. Low frequency noise study of entertainment venues. Matthew Nolan (Lloyd George Acoust., Martinique Mews, Perth, Western Australia 6025, Australia, mnolan565@gmail.com)

Noise impacts from entertainment venues have long been a challenge to communities, local government and venue operators. A study into the decay rate of low frequency music noise from venues was undertaken in the Entertainment Precinct of Northbridge, Western Australia, for the Department of Water and Environmental Regulation (DWER). A drone mounted measurement method was utilised to gain an understanding of venue noise emissions via roof elements, combined with on-ground measurements and software noise modelling. The aerial measurements proved effective at low frequency, allowing for a more comprehensive analysis of the ten venues assessed. While the study revealed several challenges and discoveries, the concept of aerial noise measurement was proven to be useful in this application. The findings will further contribute to the development of guidelines and recommendations for noise management in Northbridge, ensuring a better acoustic environment for residents and visitors while maintaining the viability of the venues in the area.

2:20–2:40 Break

Invited Paper

2:40

1pNSa5. Development of a guideline for the assessment of low frequency noise in Victoria, Australia. Marc Buret (Air, Odour and Noise Sci. Unit, Environment Protection Authority Victoria, GPO Box 4395, Melbourne, Victoria 3001, Australia, marc.buret@epa.vic.gov.au), Elaine Just (Air, Odour and Noise Sci. Unit, Environment Protection Authority Victoria, Melbourne, Victoria, Australia), and Audrey Samuel (Environment Protection Authority Victoria, Melbourne, Victoria, Australia)

The Victorian Environment Protection Act 2017 commenced in July 2021 and includes a general environmental duty to minimise the risk of harm to human health and the environment from pollution and waste (including noise), as well as an obligation to not emit "unreasonable noise". Unreasonable noise occurs if limits set by the Environment Protection Regulations 2021 are exceeded, or having regards to factors included in the Act's definition of unreasonable noise. The Regulations prescribe 'frequency spectrum' as one of these factors. In this context, the Environment Protection Authority Victoria developed the 'Noise guideline: Assessing low frequency noise' (publication 1996) that applies to commercial, industrial and trade premises. This guideline provides the steps for assessing low frequency noise from existing and the approach to follow when planning and designing new developments in Victoria. It is the reference document to understand the risk of harm from the emission of low frequency noise and assess and address low frequency noise. The development of the guideline and the underlying considerations are presented.

Contributed Papers

3:00

1pNSa6. The application of the Psychoacoustics Perception Scale (PPS) in the measurement of soundscape with low-frequency traffic noise. Kuen Wai Ma (The Hong Kong Polytechnic Univ., 11 Yuk Choi Rd., Hung Hom, Hong Kong Nil, Hong Kong, kuen-wai.ma@polyu.edu.hk), Cheuk Ming Mak, Fu Lai Korris Chung (The Hong Kong Polytechnic Univ., Hong Kong, Hong Kong), and Hai Ming Wong (The Univ. of Hong Kong, Hong Kong, Hong Kong)

In urban environments, road traffic noise is a significant source of low-frequency sounds. Recent research has revealed that the way humans perceive sounds is mainly determined by three perceptual dimensions, namely Evaluation (E), Potency (P), and Activity (A). The Psychoacoustics Perception Scale (PPS) is a valid and reliable psychometric tool that can be used to quantitatively evaluate the human general judgment (E), sensitivity to the magnitude (P), and sensation of the temporal and spectral compositions of sounds (A), respectively. The aim of this study was to evaluate the help of the PPS in assessing the urban environments with low-frequency road traffic noise. The assessments of the urban environments are in consist of the objective measurement of acoustic and psychoacoustic metrics and the subjective measurement of using the PPS. The statistical analysis results showed that the scores of PPS were the significant predictors of the soundscape gradings of the road traffic noise. The application of the PPS will be a holistic approach to improve the measurement, analysis, and evaluation of soundscape with low-frequency traffic noise to the current standards.

3:20

1pNSa7. Analysis of the impact of low-frequency traffic noise in two South American cities. Walter Montano (Tech., ARQUICUST, Luis Clavarino 1227, Gualeguaychú, Entre Ríos E2820, Argentina, wmontano@arquicust.com), Alice E. Gonzalez (Acoust. Res., Eng. Faculty, Montevideo, Uruguay), and Elena I. Gushiken (Tech., ARQUICUST, Lima, Peru)

Traffic noise is one of the most important sources of annoyance in South American cities. The obsolescence of vehicles and the lack of controls are two of the main causes of this noisy traffic. Since the legislation considers it as a typical source of urban noise, the metric to analyze traffic noise is the A-weighted dB (so-called dBA); therefore, it is not easy to analyze its real impact on human health because low-frequency sound levels are strongly reduced by A-weighting. This article presents the results of a field study carried out in Lima (Peru) and Montevideo (Uruguay) to analyze the impact of traffic noise focused the lowest frequencies, in the range defined by ISO (1/3 octave bands between 16 and 200 Hz), and the WHO recommendation using the whole audible frequencies. The data for Lima were taken in places where a specific acoustic EIS has been required, while the Montevideo's one were for its noise map. The spectra were processed looking for a representative low-frequency traffic noise metric, so the values of (C–A) obtained by WHO recommendation and ISO recommendation are compared. The expected impact on human health is much more worrisome when low-frequency range are explicitly considered.

3:40

1pNSa8. Modernization of the instrumentation requirements to support ISO noise test standards. Jeff G. Schmitt (VIacoustics, 2512 Star Grass Circle, Austin, TX 78745, jeffs@viacoustics.com), Joseph Hook, and Michael Schaffer (VIacoustics, Austin, TX)

ISO TC43/SC1/WG28 develops the basic machinery noise test standards for products and equipment. This includes the determination of sound power levels in the ISO 3740 series and the measurement of operator/bystander sound pressure levels in the ISO 11200 series. The instrumentation requirements for these standards have historically been based on a reference the sound level meter standard, IEC 61672, supplemented by another to IEC 61260 for fractional octave filters. Both IEC standards are fundamentally based on the concept of the measurement system being a packaged sound level meter. Many modern implementations of these standards no longer use a sound level meter. Many now use multi-channel, computerized instrumentation systems and custom software. These systems provide measurement equivalence to the IEC sound level measurement standards. However, they don't fully comply with all of the requirements in these IEC standards. Many acoustic testing programs have chosen to ignore these non-compliance issues, have implemented systems that are more efficient, lower measurement uncertainty and provide measurement equivalence to IEC 61672 and IEC 61260. As industry has already decided to implement these systems, there is a need to develop modernized instrumentation requirements to support the basic sound power and sound pressure level measurement standards using current technology.

Invited Paper

4:00

1pNSa9. Benefits of using ANSI/ASA S12.9 -2020/Part 7: "Measurement of environmental sound: Measurement of low-frequency noise and infrasound outdoors and in the presence of wind and indoors in occupied spaces." George Hessler (Hessler Assoc., St George, UT) and Norm Broner (Broner Consulting, 159 Victoria Parade, Melborne, Victoria, Australia, norm@broner.consulting)

I was invited and honored by Dr. Paul Schomer to Chair working group S12/WG 54 for Part 7 of ANSI S12.9. The working group had 23 learned members all at the reasonable disposal of the Chair. At the start I knew the technical tasks that would have to be done and it would be a time-consuming process, at least a year; completion took over three years. Some of the fundamental tasks required were (a) Define and quantify wind-induced-noise (WIN) for standard wind screen measurements and with upgraded wind protection to lower WIN, (b) Define conversion corrections necessary between post-mounted above grade and ground plane mounted microphones, and (c) Define a repeatable technique for measuring pressure levels outside and inside structures to characterize noise reduction NR in 1/3 octave bands for the facade of the structure. These tasks were completed with much effort by numerous group members. The standard may be consulted for analyses and reference, particularly Annexes A through D, as well as for authoritarian standard purposes. This presentation will outline the many valuable procedures in the standard for measuring low frequency noise and infrasound.

Session 1pNSb

Noise: Ground Transportation Noise

James E. Phillips, Cochair
Intertek, 4703 Tidewater Ave., Suite E, Oakland, CA 94601

Briony E. Croft, Cochair
Acoustic Studio Pty Ltd, 27/43-53 Bridge Road, Stanmore, 2048, Australia

Contributed Papers

1:00

1pNSb1. Validation of a railway rolling noise model on a system with slab track and resilient wheels. Briony E. Croft (Acoust. Studio Pty Ltd, 27/43-53 Bridge Rd., Stanmore, New South Wales 2048, Australia, briony.croft@acousticstudio.com.au), Luke Watry (Wilson Ihrig, Seattle, WA), and Shankar Rajaram (Sound Transit, Seattle, WA)

This study investigates rolling noise emissions using an analytical model of the Seattle Sound Transit Light Rail system developed using Train Noise Expert (TNE) software. Measurements of track frequency response, decay rate, rail and wheel roughness were used to determine model input parameters. Wheel modal test results were used to characterize the wheels. Wayside noise and track vibration measurements during revenue service train passbys were used to experimentally validate the model. The average difference between measured and predicted overall passby LAeq noise at a position immediately adjacent to the track was 0.1 dBA (average from four scenarios, i.e. two surface track sites with two train types). Inspection of the predicted versus measured noise spectra indicates the model does not exactly match the measured spectrum in every frequency band, however the correlation with overall spectrum shape is good in the frequency bands that contribute most to the overall A-weighted level. It is concluded that the model provides an accurate representation of the most relevant physical rolling noise factors and is reliable with resilient wheels and light rail vehicles on ballast and slab track.

1:20

1pNSb2. The relationship between freight and passenger train noise levels and distance on the Sydney Metropolitan Rail Network across 281 measurement sites. Conrad Weber (Renzo Tonin & Assoc., Level 1, 418a Elizabeth St., Surry Hills, New South Wales 2010, Australia, conrad.weber@renzotonin.com.au) and Christopher Schulten (Freight, Transport for NSW, Parramatta, New South Wales, Australia)

Transport for NSW undertakes noise monitoring at a range of distances and locations around the NSW Government-managed rail network. The monitoring quantifies noise levels from freight and passenger train operations and informs delivery of noise reduction programs including the Transport for NSW's Freight Noise Attenuation Program, which was launched in 2015. Comprehensive noise monitoring over a typical period of two weeks has been undertaken at 281 representative locations adjacent the passenger and freight rail network around Sydney. Results were analyzed to determine the L_{AE} noise levels from each freight and passenger train passby, and the average weekday $L_{Aeq(15hour)}$ daytime and $L_{Aeq(9hour)}$ night-time levels. Curves of best fit are presented that illustrate the relationship between freight and passenger train noise levels and the distance to the near track. These curves of best fit can be utilised by Government agencies and acoustic practitioners to quantify the potential noise impact of freight and passenger train operations at a range of distances and locations close to the NSW Government-managed rail network and identify appropriate noise mitigation measures.

1:40

1pNSb3. Rail noise across three states in Australia—Operational noise assessment on Inland Rail. Arvind Deivasigamani (Acoust. and Vib., SLR Consulting Australia Pty Ltd., Level 11, 176 Wellington Pde, East Melbourne, Victoria 3002, Australia, adeivasigamani@slrconsulting.com), Aaron McKenzie (Acoust. and Vib., SLR Consulting Australia Pty Ltd., North Sydney, New South Wales, Australia), and Susan Kay (Environment Advisory, Australian Rail Track Corp., Broadmeadow, New South Wales, Australia)

Inland Rail Project is a proposed 1700 km freight rail line connecting Melbourne and Brisbane via three States in Australia–Victoria, New South Wales, and Queensland. Operational rail noise assessment is undertaken differently across these States in Australia. The assessment generally considers the existing and future noise exposure across day and night, along with single pass-by maximum levels. However, the assessment durations and statistical means of assessing single events differ across the States. The key acoustic criteria relevant to operational noise assessments and how they differ across the project footprint are discussed in this paper. Specifically, the effect of application of various criteria in the project context are discussed, along with statistical analysis of train pass-by noise data sets. A Monte-Carlo based methodology is proposed for estimating Single Event Maximum, a criterion for assessing single rail-noise events in Queensland. The methodology is discussed in the context of field data and greenfield scenarios where the method may be beneficial for acoustic consultants. The effects of inclusion of noise sources such as train horns and crossing bells are also briefly discussed. Finally, a proposed methodology for a consistent assessment of rail noise on such nationally spread projects is discussed.

2:00

1pNSb4. Acoustic roughness outcomes from rail maintenance processes. Briony E. Croft (Acoust. Studio Pty Ltd., 27/43-53 Bridge Rd., Stanmore, New South Wales 2048, Australia, briony.croft@acousticstudio.com.au), Mark Reimer (Independent Consultant, Winnipeg, MB, Canada), Aaron Miller, and David Hanson (Acoust. Studio Pty Ltd., Stanmore, New South Wales, Australia)

Rail roughness is a key determinant of the rolling noise emissions of many railway systems, particularly those systems where wheel roughness is relatively low. Rail roughness can vary considerably. The noise emissions from trains operating on very smooth, quiet, worn in systems can be 25 dBA less than the noise from identical trains operating on the same trackform but with rough or corrugated rails. Rail roughness varies over time, with changes resulting from gradual wear during normal operations but also from maintenance interventions. Rail grinding, milling or acoustic polishing can cause immediate and sometimes dramatic changes in rail acoustic surface

condition, and noticeable changes in noise character. This paper documents the acoustic roughness outcomes achieved by various rail maintenance processes, in comparison to acoustic roughness from worn in tracks. The data presented includes measurements of surface finish from conventional rail grinding, rail milling, and from newer specialist acoustic grinding technologies. The rail roughness results are used to demonstrate the differences in noise emissions following rail maintenance activities. The results confirm the importance of maintaining low rail roughness as a means of noise control at source.

2:20

1pNSb5. Hypothetical assessment of light rail vehicles idling at a terminus in New South Wales. Aaron Miller (Acoust. Studio Pty Ltd., Stanmore, New South Wales 2048, Australia, aaron.miller@acousticstudio.com.au), Briony E. Croft (Acoust. Studio Pty Ltd., Stanmore, New South Wales, Australia), Jordan McMahon, and Adam Sirianni (SLR Consulting Australia Pty Ltd., North Sydney, New South Wales, Australia)

The NSW Rail Infrastructure Noise Guideline (RING) delineates the assessment of airborne noise from light rail vehicles and from railway activities taking place at fixed locations (stationary noise sources). Airborne noise from light rail vehicles is assessed against the RING criteria, while airborne noise from stationary noise sources is assessed against the much more stringent Industrial Noise Policy (INP) which has since been superseded by the Noise Policy for Industry (NPfI). In real-life operations, light rail vehicles will often idle with their air conditioners running at terminus locations that are often located near sensitive receivers due to network constraints. This creates a pseudo-stationary noise source, which to a nearby sensitive receiver is indistinguishable from a stationary noise source. This paper hypothetically assesses the noise emissions from light rail vehicles idling at a terminus against both the RING and hypothetical NPfI criteria, and examines the likely mitigation that would be required for each assessment. Commentary is also provided regarding the suitability of the RING and the NPfI to this kind of noise source.

2:40

1pNSb6. Abstract withdrawn.

3:00–3:20 Break

3:20

1pNSb7. Review of road noise corrections in New South Wales road noise assessment method. Rebecca Warren (WSP Australia, Level 27, 680 George St., Sydney, New South Wales 2000, Australia, rebecca.warren@wsp.com)

Road noise assessments are coming under increased scrutiny from regulators as a result of intensified urban development in New South Wales (NSW), Australia. This paper presents a summary of road assessment methodology, an assessment of the assumptions used in their calculations (particularly as applied to NSW) and some of their limitations. An analysis has been completed of the suitability of the correction used to correlate modelled results to Australian conditions and potential dependencies to other variables, through an analysis of data of completed projects. This paper also aims to identify the effectiveness of recently introduced road noise guidelines, which were developed to streamline the assessment process across the industry. Recommendations are provided to revisit data collection over a wider geographic area or identification of other possible variables which would influence the modelling of real life scenarios.

3:40

1pNSb8. New road surface noise corrections for New Zealand. Richard Jackett (Waka Kotahi NZ Transport Agency, 44 Bowen St., Wellington 6011, New Zealand, richard.jackett@nzta.govt.nz), John Bull (Waka Kotahi NZ Transport Agency, Christchurch, New Zealand), Stephen Chiles (Chiles Ltd., Christchurch, New Zealand), Rob Wareing, and George Bell (Altissimo Consulting, Christchurch, New Zealand)

New Zealand's transport agency has developed a close-proximity (CPX) noise trailer and primarily operates it as a road surface research tool. Early CPX survey data revealed very high longitudinal variability in the performance of porous asphalts on the highway network (±5 dB). Further study and optimization of porous asphalt mixes has resulted in high-performance low-noise surfaces that are reliably 5 dB quieter than standard NZ porous asphalt. The new surfaces and improved surface data inspired a recalibration of NZ's implementation of the CRTN road-traffic noise model, for which a novel method based on CPX survey data and individual vehicle pass-by measurements was employed. The relationship between CPX and wayside levels was more complex than anticipated; CPX measurements did not capture all the surface attributes/interactions contributing to the wayside level (such as porosity, directivity). The methodology proved successful and a new table of surface corrections has been published, embodying the recalibration of CRTN for NZ in 2023. CRTN predictions using the new corrections have performed well when evaluated against independent road-traffic noise monitoring data from NZ highways.

4:00

1pNSb9. Acoustic labelling of tires, road vehicles and road pavements: A vision for substantially improved procedures. Ulf Sandberg (Swedish National Rd. and Transport Res. Inst. (VTI), Olaus Magnus väg 35, Linköping 58195, Sweden, ulf.sandberg@vti.se)

The EU requires vehicle tires to be tested for noise emission and labelled. Some other countries also have similar procedures. Preparations for labelling of acoustical properties of road pavements is also ongoing in EU. Road vehicles and their tires must be tested and meet noise limits worldwide. However, all cases involve problems with measurement methods and conditions, which create uncertainties higher than desirable to make fair decisions on which product to select. For tires and road vehicles the major obstacle is the test track surface, which cannot be sufficiently well defined and reproducible. Also, far from all products (tires) are measured, so commonly only the noisiest products are correctly labelled. Although attempts are made to go indoors to do the measurements, they are still for the foreseeable future going to be made outdoors, with all the influence of weather and seasons that this causes. The author has worked most of his long career with such issues and in this paper, the author outlines his vision of how most of the problems (uncertainties) can be reduced to acceptable levels, mainly by making measurements indoors, with reproducible and representative conditions for both the labelling and the actual noise emission in traffic.

4:20

1pNSb10. Unifying low and high-frequency noise mitigation: A novel dual noise barrier approach. Pawan Pingle (NVH Consultancy, Maharashtra, India), Gloria Pignatta (Red Ctr. West Wing, UNSW Sydney, 2052, Sydney, New South Wales 2052, Australia, g.pignatta@unsw.edu.au), and Samuele Schiavoni (Metexis, Perugia, Italy)

Road highways and urban areas often employ noise barriers to shield residential areas from excessive traffic noise. Despite continuous efforts, developing materials for noise barriers that can effectively mitigate noise across a wide frequency range is still a challenge. Specifically, creating a single material capable of attenuating both low and high-frequency noise has proven to be a daunting task. Current practices incorporate porous materials for absorbing low-frequency noise at specific locations. To address this limitation and offer a technique for traffic noise reduction, this research proposes a novel approach utilizing two hindrances: pores for low-frequency and bristles for high-frequency sound waves. An ANSYS Harmonic Acoustic module simulation was conducted to assess the proposed technique's effectiveness. The simulation encompassed geometry definition, material selection, boundary condition constraints, meshing, and analysis. The results indicate a significant reduction in noise at high frequencies (16 000 Hz) of approximately 30 dB, decreasing from 101.4 to 68.1 dB. This demonstration showcases the potential of a staggered arrangement of bristles and pores to handle a wide range of frequency content and effectively mitigate both low and high-frequency traffic noise. The novel dual noise barrier approach offers a promising solution to combat diverse traffic noise challenges, providing a more peaceful soundscape for urban residents.

4:40

1pNSb11. Investigating biophilic design and adaptive acoustic comfort in office spaces in the existence of indoor greenery, operable window, and road traffic noise. Merve Esmebasi (Architecture, National Univ. of Singapore, SDE-1 Office 05-12 Architecture Dept. NUS, Singapore 117566, Singapore, merve@u.nus.edu) and Siu-Kit Lau (Architecture, National Univ. of Singapore, Singapore, Singapore)

This study examines how indoor greenery and operable windows influence the annoyance caused by road traffic noise in office spaces. While existing studies have explored how urban landscapes can reduce noise annoyance in residential areas, there is a gap in understanding the effects of indoor greenery in office environments. The present study utilized a structural equation model to analyze the cause-effect relationships concerning noise perception. Thirty-two combinations of greenery levels, window conditions, and traffic noise across four different sound pressure levels (50 dBA–65 dBA) were evaluated. To explain noise perception, the model considered perceptual attributes like pleasantness, eventfulness, appropriateness, preference, and visual aesthetics. The results indicated that traffic noise level affects visual perception, and there is a two-way interaction between auditory and visual perceptions. These findings advocate for incorporating biophilic design to create visually appealing and acoustically pleasant office spaces. Moreover, the study emphasized the positive influence of open windows on the perceived appropriateness of traffic noise resulting in less annoyance. Overall, this research contributes valuable insights to discussions on creating adaptive acoustic comfort in naturally ventilated spaces and enhancing visual aesthetics through biophilic design to mitigate noise annoyance in office environments effectively.

Session 1pPA

Physical Acoustics, Architectural Acoustics and Engineering Acoustics: Acoustical Measurements and Sensors for Challenging Environments I

Cristian Pantea, Cochair
Los Alamos National Laboratory, PO Box 1663, MS D429, Los Alamos, NM 87545

Akira Nagakubo, Cochair
Engineering, Osaka university, M1-523, 2-1, Yamada-oka, Suita, 565-0871, Japan

Chair's Introduction—12:55

Invited Papers

1:00

1pPA1. Recent progress on nonlinear ultrasonic phased array for closed-crack imaging. Yoshikazu Ohara (Dept. of Mater. Processing, Tohoku Univ., 6-6-02 Aoba, Aramaki-aza, Aoba-ku, Sendai, Miyagi 980-8579, Japan, ohara@material.tohoku.ac.jp)

Closed cracks are challenging defects for ultrasonic testing since they are transparent to conventional ultrasound techniques. This can cause the underestimation or overlook of closed cracks, resulting in potential catastrophic accidents. To solve this problem, several types of nonlinear ultrasonic phased array has been developed by combining nonlinear ultrasonics with phased array (PA). In this study, we introduce the recent progress on nonlinear ultrasonic PA. The first one is called fundamental wave amplitude difference (FAD), which is based on the nonlinear incident-wave-amplitude dependence of fundamental responses. The key to the success of FAD is how high incident wave amplitude can be generated in samples to cause the contact vibration of crack faces. However, increasing the incident wave amplitude at MHz frequencies is not easy. To enhance the applicability of nonlinear ultrasonic PA, we have also developed ultra-fast imaging (MHz range) with pump excitation (kHz range). We have successfully captured high-speed crack dynamics using plane wave imaging (PWI) during the pump excitation that can generate a large displacement of more than 1000 nm. Furthermore, we will talk about ongoing work toward 3D crack imaging. [Work partially supported by JSPS KAKENHI (19K21910, 21H04592, 22K18745) and JST FOREST program (JPMJFR2023).]

1:20

1pPA2. Nonlinear resonant ultrasound spectroscopy using white-noise excitation. Paul R. Geimer (Mater. Sci. and Technol., Los Alamos National Lab., Los Alamos, NM), Luke Beardslee (Earth and Environ. Sci., Los Alamos National Laboratroy, Los Alamos, NM), and Timothy J. Ulrich (ALDW, Los Alamos National Lab., Reed McDonald Bldg., Texas A & M University, College Station, TX 77845, tju@lanl.gov)

Nonlinear resonant ultrasound spectroscopy (NRUS) is a technique for non-destructive sample evaluation by which a strain-dependent nonlinear parameter is quantified that exhibits heightened sensitivity to defects as compared to monitoring of linear resonance attributes. As NRUS measurements have traditionally used a controlled sweep of sinusoidal inputs, applications of the method have been largely limited to laboratory experiments. Here we present NRUS results which utilized a white noise signal as input, in order to better approximate ambient excitation and facilitate efficient validation using time-domain models. Studied samples were chosen to reflect a range of nonlinear responses and underlying mechanisms (acrylic, sandstone, fractured steel). We excited the studied samples using bonded piezoelectric transducers which delivered white-noise inputs at increasing amplitudes, with the nonlinear response measured with a laser doppler vibrometer. By comparing results against those obtained from traditional NRUS, we find comparable trends in the nonlinear parameter under white-noise excitation, including similar relative differences in magnitude across the samples and consistent values across similar mode shapes. While we note that white-noise strain estimation is complicated by simultaneous excitation of multiple resonant modes, NRUS results from white-noise excitation demonstrate promising viability of the technique for (1) *in-situ* inspection, and (2) a more-tractable approach to model the physics of various NRUS signals.

1:40

1pPA3. Ultrasonic *in situ* measurements for hybrid metal additive manufacturing. Joseph A. Turner (Mech. and Mater. Eng., Univ. of Nebraska-Lincoln, W342 Nebraska Hall, Lincoln, NE 68588, jaturner@unl.edu), Luz D. Sotelo (Purdue Univ., West Lafayette, IN), Nathanial Matz, W. Tanner Brandl, and Morgan Ferrin (Mech. and Mater. Eng., Univ. of Nebraska-Lincoln, Lincoln, NE)

Hybrid additive manufacturing (AM) of metals includes synergistic secondary processes such as milling or peening that impart changes to the component for improved performance. Although a valuable concept, implementation is difficult to verify due to a current lack of *in situ* inspection methods. In this presentation, two different strategies are described with respect to ultrasonic inspections for

hybrid metal AM. Both are used for inspections during laser-based directed energy deposition with milling as the hybrid process. The first uses a transducer mounted below the build plate. This configuration allows waves to interrogate the entire sample during all additive and subtractive manufacturing steps. Experiments with Ti6Al4V show the changes in ultrasonic wave speed and attenuation that track the manufacturing temporally. The second inspection approach uses ultrasonic surface waves at the top surface of the sample. These measurements are used on samples of 316L stainless steel at the end of additive steps as well as before and after milling to quantify localized information regarding the most recent layers. Wave speed and diffuse scattering information show the variability with respect to sample geometry, build height, and process parameters. Finally, advantages and limitations of both approaches and prospects for validated parts are discussed.

2:00

1pPA4. Defect detection and imaging in elastic materials with complex geometries. Takahiro Hayashi (Graduate School of Eng., Osaka Univ., 2-1 Yamadaoka, Suita, Osaka 565-0871, Japan, hayashi@mech.eng.osaka-u.ac.jp)

Additive manufacturing has made it possible to easily create parts with complex shapes, even out of metal materials. However, in order to guarantee the quality of such parts, there is a need to establish a nondestructive inspection technique that is feasible for such complex structures. In this presentation, I will introduce a technology that can visualize internal damage even in complex objects by utilizing the ultrasonic field generated and diffused inside the object with non-contact manner. Ultrasonic waves generated by laser irradiation resonate with the natural frequency of defects such as delamination near the surface of an object. By scanning the laser beam for generating ultrasonic waves and detecting vibrations at a fixed position on the surface of the object, an image of defects in the vicinity of the surface can be obtained as a distribution of vibration energy. The distribution of vibration energy does not depend on reflected or scattered waves from the object's wall, but only on information from the point where the laser is irradiated. This technique is expected to be especially useful for *in-situ* inspections of the additive manufacturing objects.

Contributed Papers

2:20

1pPA5. Laser-induced sonar: A promising approach for improved underwater acoustic sensing. Yellaiah Janapati (Adv. Ctr. of Res. in High Energy Mater., Univ. of Hyderabad, Hyderabad, Telangana 500046, India, yelluacrhem@uohyd.ac.in)

This study presents the characteristics of acoustic pressure impulses generated by nanosecond laser-induced filamentation (ns-LIF) of water, focusing on the spatial, temporal, and spectral domains. In the time domain, the peak-to-peak (P_k-P_k) overpressures increase with higher incident optical energy while the arrival time remains constant. Notably, linearly polarized pulses exhibit higher P_k-P_k overpressures than circularly polarized pulses. Acoustic measurements of ns-LIF in water demonstrate a linear correlation between filament size and incident laser energy due to multiple plasma sources along the optical beam propagation. Alongside the temporal information, the spectrogram visualizes broad-spectrum underwater acoustic pressure impulses ranging from 10 to 800 kHz, perpendicular to the optical beam propagation. The low-frequency instantaneous underwater acoustic signals generated by ns-LIF is $\sim$90 kHz, offering advantages such as extended propagation distances and reduced attenuation in water. In addition to the experimental investigation, finite element analysis is employed to visualize the propagation and interaction of underwater acoustic signals across various interfaces. This integrated approach provides valuable insights into the behavior and characteristics of underwater signals. Eventually, our findings demonstrate the successful development of remote laser-induced sonar technology.

2:40

1pPA6. Highly constrained low level fluid sensing in process pipes utilizing non-invasive ultrasonic methods. Eric S. Davis (MPA-11, LANLP.O. Box 1663, Los Alamos, NM 87545, esdavis@lanl.gov), Lalith Pillarisetti, Cristian Pantea, Abhishek Saini, and Pavel Vakhlamov (MPA-11, LANL, Los Alamos, NM)

Determination of the presence of hazardous waste in sealed pipes is a major issue of concern in many industries. Typically, when hazardous process piping needs to be opened, it is very difficult to truly know whether the pipe is empty, and thus free of hazardous material. Therefore, standard procedure is to assume worst case scenario and dedicate significant resources to Hazmat teams and containment when opening the piping or preparing it for decommissioning. In this work, we explore three separate ultrasonic-based techniques for very low-level fluid sensing in process piping. It was found that ultrasonic techniques suffer from several issues, including guided wave contamination, local resonance, overlap from other echoes, and more. To overcome these issues, three separate techniques were explored, including pulse-echo with subtraction, total focusing method (TFM) imaging, and another technique that exploits the local resonance of the pipe when it is fluid-loaded. It was found that, with proper calibration and signal processing, pulse-echo is an easy-to-use technique down to relatively low fluid levels. However, the best performance came from modified total focusing method imaging, which could detect extremely low fluid levels while being robust from false negatives (detecting no waste when there was waste).

3:00–3:20 Break

3:20

1pPA7. Robotic sensing for buried pipes with sound waves. Yicheng Yu (Dept. of Mech. Eng., Univ. of Sheffield, Mappin St., Sheffield S1 3JD, United Kingdom, yicheng.yu@sheffield.ac.uk), Kirill Horoshenkov (Mech. Eng., Univ. of Sheffield, Sheffield, United Kingdom), Rob Worley, and Sean Anderson (Dept. of Automatic Control and Systems Eng., University of Sheffield, Univ. of Sheffield, Sheffield, United Kingdom)

The length of buried sewer and drainage network in Europe is several million kilometres. Autonomous robots are being developed to inspect this massive network of pipes pervasively. These inspection technologies traditionally rely on CCTV images. However, detection of the condition of buried pipes with autonomous robots is challenging computationally. Acoustic waves provide an efficient alternative to conventional CCTV methods to detect a range of artefacts that can lead to pipe failure and map these conditions. This paper presents an acoustic method for simultaneous condition detection, localization, and classification in air-filled pipes. A microphone array is used to estimate the reflection coefficient from a range of artefacts. This information is used together with a regularization method. A wavelet basis function is adapted to enhance the fidelity of collected acoustic data. It is shown that the wavelet components can also be used to train and to test a support vector machine (SVM) classifier for the condition identification. This work can also inform the route planning and low-level control algorithms for autonomous robots which are being developed for the inspection of buried pipes.

3:40

1pPA8. Signal processing and acoustic calibration method of millimeter-wave vibration sensor. Jun Kuroda (Adv. Technol. Res. Inst., KYOCERA Corp., 3-7-1 Minatomirai, Nishi-ku, Yokohama, Kanagawa 220-0012, Japan, jun.kuroda.zs@kyocera.jp), Yuudai Matsue, Yuu Kashima (Adv. Technol. Res. Inst., KYOCERA Corp., Yokohama, Japan), Yasuhiro Oikawa, and Haruka Nozawa (Dept. of Intermedia Art and Sci., Waseda Univ., Shinjuku-ku, Tokyo, Japan)

Millimeter-wave sensors can be utilized as non-contact vibration sensors. Applications of millimeter-wave vibration sensors are the vital sensing of humans (heartbeat, respiration, body motion), factory machines, and vibration sensing of social infrastructures, such as buildings and bridges. Millimeter-wave vibration sensors can simultaneously detect the position and the waveforms of the vibration displacements of vibration objects. The primary signal processing of millimeter-wave sensors is two-dimensional fast Fourier transform (2D-FFT), detecting the arrival direction (DOA) and processing the micro-Doppler signals. The detectable areas of the millimeter-wave vibration sensors are defined using the parameters related to these three signal-processing blocks. Measured vibration waveforms by millimeter-wave vibration sensors include all components of the vibration displacements in the defined detectable area. Therefore, the waveform of the vibration displacement is influenced by the definition of the detectable area and the vibration form of the vibrating object. The measured values of millimeter-wave vibration sensors must be calibrated with attention to these influences. The vibration waveforms of vibrating objects are estimated by acoustic measuring methods, such as the laser Doppler vibrometer and the parallel-phase shifting interferometry. This paper describes calibration methods of millimeter-wave vibration sensors employing these two measuring methods.

4:00

1pPA9. Vessel and landside noise contributions from port activities: What does 2 years of data tell us? Priyadarshi Pandey (GHD Pty Ltd, Level 15, 133 Castlereagh St., Sydney, New South Wales 2113, Australia, Pri.Pandey@ghd.com)

Port Authority of NSW has developed a Port Noise Policy for the management of vessel and landside port noise at the port precinct level at Glebe Island and White Bay, Sydney, NSW. The implementation of this policy is currently underway and involves undertaking long term, unattended noise monitoring at multiple locations around the Port using directional sound level meters. Concurrent data for other key variables such as vessel movements and meteorology is also obtained from existing data APIs. This paper provides details around the methodology adopted to extract vessel and landside noise contributions from the broader ambient noise environment for Port activities. Detailed analyses of vessel and landside noise contributions is undertaken in the context of parameters such as vessel parameters and meteorology. The paper provides insights into the role of these variables in the context of measured noise levels and the spatial and temporal variations in noise around the Port.

4:20

1pPA10. Enhanced noise propagation due to atmospheric effects: Practical considerations—Part 1. Travis Hancock (Marshall Day Acoust., 10/50 Gipps St., Melbourne, Victoria 3440, Australia, thancock@marshallday.com), Liam Kemp, and Justin Adcock (Marshall Day Acoust., Collingwood, Victoria, Australia)

How far should an acoustical consultant go to investigate enhanced noise propagation due to atmospheric effects? This paper was inspired by recent updates to Victorian noise policy, which includes more stringent requirements for consideration of atmospheric effects. The authors have identified an opportunity for development of a consolidated practical method for acoustical consultants to apply in Australian conditions. Predictive calculations implementing various standards can estimate atmospheric effects based on either broad assumptions, subjective observations, or detailed weather data. Acoustical consultants are often reliant on incomplete and potentially irrelevant weather data. This includes data from sources such as public weather stations far away or local weather stations close to ground level. This paper is the first part of a two-part study designed to inform a best practice method for acoustical consultants assessing atmospheric effects with respect to enhanced noise propagation. Part one, this paper, will review the current state of knowledge including different policy approaches and predictive standards. Part two would be a field study with concurrent detailed measurements of atmospheric conditions and noise levels to inform appropriate weather data collection methods and predictive standards for Australian conditions.

4:40

1pPA11. Managing noise from port operations via the use of automated noise monitoring systems. Priyadarshi Pandey (GHD Pty Ltd., Level 15, 133 Castlereagh St., Sydney, New South Wales 2113, Australia, Pri.Pandey@ghd.com)

Ports are key pieces of infrastructure that enable the transportation of goods and passengers. Bays Port, Sydney NSW is located in close proximity to communities and balances the increased demand for Port facilities and services with community expectations, environmental requirements and regulatory approvals. For noise generated from Port activities at Bays Port, historically, individual port users have been required to monitor and evaluate noise under different environmental requirements and planning approvals, which can lead to inconsistent reporting, noise limits and regulation between operators and a lack of clarity for the community. Port Authority of NSW has implemented a first of its kind policy in Australia for the management of Port noise from both individual vessel and landside noise sources. To achieve these objectives, an automated directional noise monitoring system has been implemented to provide real-time measurement and analysis of noise contributions from vessel and landside activities. The system utilizes live data of individual vessel tracking locations, meteorology and an array of 6 direction microphones located around the port infrastructure. This information feeds into a live platform to facilitate management of port noise operations. This paper provides an overview of the Port Noise Policy and the use of unattended directional noise monitoring systems to meet the policy's aims. Case studies are also provided to illustrate the benefits of real time monitoring systems in pro-actively managing port noise.

5:00

1pPA12. Long tunnel noise propagation. Lachlan T. Woolf (Renzo Tonin & Assoc., L1/418A Elizabeth St., Surry Hills, New South Wales 2010, Australia, lachlan.woolf@renzotonin.com.au), Michael Gange (Renzo Tonin & Assoc., Surry Hills, Australian Capital Territory, Australia), Mattia Tabacchi, and Adrian Morris (Renzo Tonin & Assoc., Surry hills, New South Wales, Australia)

The number of long tunnels in Sydney has increased in recent years with the development of underground road and rail projects. Understanding the propagation of noise along the tunnels and into the environment during the construction phase is important when assessing noise for 24/7 tunnelling operations. In-tunnel noise assessment of mechanical equipment is also required as part of the project's technical criteria and approvals. The prediction of in-tunnel noise levels is required from stationary sources such as ventilation fans and jet fans. The acoustic design of these tunnels is challenging as a feature of the acoustics of long spaces is that classical room acoustic theory is not applicable, since the assumption of a diffuse field does not hold with the extreme dimensional condition. This paper details the difference between in-tunnel noise measurements from an omni-directional noise source compared with the modelled results from 3D noise modelling software packages.

Session 1pPP

Psychological and Physiological Acoustics: Understanding Hearing in a Dynamic World

Alan Kan, Cochair
School of Engineering, Macquarie University, 50 Waterloo Road, Macquarie Park, 2113, Australia

Valeriy Shafiro, Cochair
Communication Disorders & Sciences, Rush University Medical Center, 600 S. Paulina Street, AAC 1015, Chicago, IL 60612

Raymond Goldsworthy, Cochair
Biomedical Engineering, Univ. of Southern California, Los Angeles, CA

Chair's Introduction—1:15

Invited Papers

1:20

1pPP1. Human auditory ecology: Extending hearing research to the perception of natural soundscapes by humans. Christian Lorenzi (Dept d'etudes Cognitives, Ecole Normale Superieure, 29 rue d'Ulm, Paris 75005, France, lorenzi@ens.fr)

A "natural soundscape" refers to the case where the contribution of acoustic events resulting from human activity can be considered as negligeable. As a consequence, natural soundscapes are only composed of biological sounds and geophysical sounds shaped by the specific way sounds propagate within the habitat under study. Within this framework, studying soundscape perception in humans aims at unveiling the relationship between the features of sound mixtures picked up at a given place and time by the peripheral auditory system of a human listener and the characteristics of the auditory percept evoked by these features. We will present a research program based on large and ecologically-valid acoustic databases recorded in protected areas aiming to (i) better understand the mechanisms involved in auditory perception of natural soundscapes; (ii) characterize and explain the effects of sensorineural hearing loss on perception of natural soundscapes; and (iii) assess the extent to which alterations in soundscape perception can be restored back to normal *via* hearing aids. This programme combines modelling and psychophysical methods to explore our ability to distinguish between habitat, time of day and season, to detect the presence of biological sound sources and to assess levels of biodiversity in the habitat.

1:40

1pPP2. Sound in context: Comprehension of auditory scenes by listeners with normal and impaired hearing. Valeriy Shafiro (Commun. Disord. & Sci., Rush Univ. Medical Ctr., 600 S. Paulina St., AAC 1015, Chicago, IL 60612, valeriy_shafiro@rush.edu), Louisa Forrest, Emily Price (Commun. Disord. & Sci., Rush Univ. Medical Ctr., Chicago, IL), and Michael S. Harris (Otolaryngol., Medical College of Wisconsin, Milwaukee, WI)

Sounds of everyday environments are rarely heard in isolation. Usually, they occur in the context of complex auditory scenes, where they are preceded and followed by other sounds or overlap with other sounds in time. In contrast, behavioral tests of auditory function are typically focused on isolated and often artificially constructed sounds which do not represent real world objects and events or the context in which they occur. Thus, results of such tests may have limited generalizability to important aspects of everyday listening. This presentation will review alternative approaches that focus on comprehension of auditory scenes and require integration of semantic information across multiple sounds. For listeners with hearing loss, our results suggest that comprehension of auditory scenes constitutes an important aspect of auditory assessment which, combined with findings from more traditional auditory tests, can provide a basis for auditory rehabilitation.

2:00

1pPP3. Assessment of environmental sound perception among listeners with hearing loss and cochlear implants. Michael S. Harris (Otolaryngol. & Commun. Sci., Medical College of Wisconsin, 8900 W. Doyne Ave., Milwaukee, WI 53226, msharris@mcw.edu), Saba Anwer, Nathan Luzum (Otolaryngol. & Commun. Sci., Medical College of Wisconsin, Milwaukee, WI), Valeriy Shafiro (Commun. Disord. & Sci., Rush Univ. Medical Ctr., Chicago, IL), and Laurie M. Heller (Psych., Carnegie Mellon Univ., Pittsburgh, PA)

Perception of environmental sounds (PES), encompassing informationally and/or aesthetically salient nonspeech, nonmusical sounds, is crucial for safety, independence, and quality-of-life among listeners with hearing loss. The aims of this study were (1) to systematically review methodologies used to assess PES among cochlear implant (CI) users and identify performance trends in this

population, and (2) to present preliminary data using a novel PES task requiring inference of materials and actions generating environmental sounds. For aim 1, PES in quiet using open- or closed-set response formats were most commonly used. PES accuracy in pediatric (3 studies) and adult (16 studies) CI users was highly variable but generally mediocre (mean correct: 31–87%). Most studies were cross-sectional; only two evaluated PES prospectively before and after CI. No significant differences in accuracy were reported between CI candidates and CI users. PES correlated in with measures of speech perception and spectro-temporal processing. For aim 2, we present preliminary findings on identification of materials and actions responsible for generating environmental sounds among normal hearing listeners and listeners with hearing loss using hearing aids or CIs. We discuss the types of materials and actions that are most difficult for listeners with hearing loss and common sources of confusion.

2:20

1pPP4. What we can do with sounds. Matthew Rodger (School of Psych., Queen's Univ. Belfast, 18-30 Malone Rd., Belfast, Antrim BT95BN, United Kingdom, m.rodger@qub.ac.uk)

Auditory perception can be dynamic when sound sources are in motion or changing, when the listener is moving relative to the source of sound, or both. Moreover, our engagement with the acoustic environment is often active rather than passive. That is, sounds can be brought into use to support and guide perceptual-motor behaviour. This presentation will review examples of recent empirical work on the role of auditory perception in the control of motor behaviour: *what we can do with sounds*. Data from experiments on intercepting moving objects without vision, acoustic navigation and orientation in people with visual impairments, exploration and learning of novel digital musical instruments, and rhythmic sounds in gait rehabilitation, all demonstrate different ways that sounds can convey action-relevant information for skilful listeners. Furthermore, they point to a need to consider the auditory system not merely as a passive antenna for inert distal stimuli, but as integrated within an active meaning-seeking organism interacting with a dynamic environment.

2:40

1pPP5. Studies of multisensory integration in sound localization and updating of auditory spatial attention during head motion. Ewan A. Macpherson (National Ctr. for Audiol., Western Univ., Elborn College EC 2262, 1201 Western Rd., London, ON N6G 1H1, Canada, ewan.macpherson@nca.uwo.ca)

In daily life, our heads are in continual motion, but much of our knowledge of spatial hearing is based on experimental paradigms in which this behavior is discouraged or prevented. Head motion can benefit spatial hearing though the creation of dynamic acoustical cues, but also creates challenges such as the need to update head-centered spatial representations or the locus of spatial auditory attention. To utilize acoustic information generated by, or to compensate for, head movements, the auditory system must integrate self-motion information provided by other sensory systems. This presentation will review and contextualize a series of studies from the author's laboratory focused on the psychophysics of dynamic sound localization and the weighting of sensory information from vestibular, proprioceptive, and visual modalities in dynamic localization and maintenance of spatially selective auditory attention during head rotation. Notable findings include a velocity-independent ~100-ms minimum stimulus duration for disambiguation of front/rear location in dynamic localization and the apparent dominance of vestibular information in the interpretation of dynamic localization cues and in attentional updating. The approach taken provides a step towards understanding the effects of naturalistic behavior on spatial hearing while maintaining significant experimental control and repeatability of stimuli and head movements.

3:00

1pPP6. Binaural fusion with ecologically valid stimuli. Justin Aronoff (Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, 901 S 6th St., Champaign, IL 61820, jaronoff@illinois.edu), Prajna BK (Speech & Hearing Sci., UIUC, Champaign, IL), Simin Soleimanifar (Speech and Hearing Sci., Univ. of Illinois at Urbana Champaign, Champaign, IL), and Leslie Bernstein (Univ. of Connecticut, Farmington, CT)

Binaural fusion, or the perception of a single auditory "image" can arise from simultaneous inputs across ears. Fusion is fundamental to binaural abilities, including sound source localization. One potent determiner of degree of fusion is the interaural correlation of the binaurally presented waveforms. Noise waveforms can be used to manipulate the interaural correlation, but the normalized correlation of the envelopes of such noises is limited to between 1.0 and 0.785 (π/4). As a result of differing properties between noises and speech, pairs of speech sounds can be generated having normalized interaural envelope correlations well below 0.785. We measured cochlear implant users' perceived degree of binaural fusion while varying the interaural correlation of speech-like signals from 0.4 to 1.0. Stimuli were generated using speech-based envelopes from multi-channel vocoders that modulated 1000 pulse-per-second pulse trains. Envelopes derived from different vocoder channels were extracted to yield interaural correlations that varied over a wider range than would be possible using noises. Preliminary results suggest that the speech-based stimuli, while differing qualitatively from noises and being more ecologically valid, yield similar effects of interaural correlation on binaural fusion, while allowing a larger range of interaural correlations to be employed.

3:20–3:40 Break

3:40

1pPP7. Assessing listening effort in children: A child-appropriate dual-task paradigm investigating the influence of different noise conditions. Julia Seitz, Karin L. Loh (Inst. for Hearing Technol. and Acoust. (IHTA), RWTH Aachen Univ., Aachen, Germany), and Janina Fels (Inst. for Hearing Technol. and Acoust. (IHTA), RWTH Aachen Univ., Kopernikusstr. 5, Aachen 52074, Germany, Janina.Fels@akustik.rwth-aachen.de)

In classrooms, children are often challenged by competing speech, loud background noises, or unfavorable room acoustics when listening to their educators. These challenging listening situations require more cognitive, attentional, and perceptual processing resources to understand speech, i.e., higher listening effort. This study introduces a dual-task paradigm to evaluate listening effort, specifically in children. The paradigm encompasses a word recognition task as the primary task and a serial recall task as the secondary task. The

influence of multi-talker babble noise on listening effort is studied in an anechoic and reverberant environment with different signal-to-noise ratios. It further incorporates aurally-accurate sound reproduction allowing plausible listening situations appropriate to children's smaller anthropometric sizes. The aim is to validate the newly developed paradigm in an experiment with children. The listening effort in noisy conditions is compared to a noise-free condition. Moreover, the influence of anechoic versus reverberant noise scenarios on listening effort is investigated.

4:00

1pPP8. Allocation of cognitive resources for older adults with normal or impaired hearing when listening to temporally degraded and masked speech. Daniel Fogerty (Dept. of Speech and Hearing Sci., Univ. of Illinois Urbana-Champaign, 901 S. Sixth St., Champaign, IL 61820, dfogerty@illinois.edu) and Judy R. Dubno (Dept. of Otolaryngology-Head and Neck Surgery, Medical Univ. of South Carolina, Charleston, SC)

Hearing impairment may affect how older listeners allocate cognitive resources when communicating in challenging environments. Older adults with normal or impaired hearing completed three speech recognition studies. The studies consisted of 15-16 measures of temporally filtered speech with (1) degraded spectral cues, (2) competing speech-modulated noise, and (3) speech vocoding in speech-modulated noise. Generalization was assessed with measures of speech interruption, segregation, and multitalker masking. To account for audibility differences, all listeners received spectral shaping fit to their hearing thresholds. Measures in each study were reduced to a single principal component (PC) that represented speech recognition abilities. Participants also completed a cognitive test battery that yielded domain-specific PCs and a measure of speech glimpsing. For each study, hierarchical linear regression analysis was completed separately for subgroups of older adults with normal or impaired hearing using cognitive PCs as predictor variables, followed by speech glimpsing, age, and hearing thresholds. Vocabulary knowledge was the best predictor of speech recognition for older adults with normal hearing, whereas working memory consistently explained ~30%-40% of the variance in recognition for older adults with hearing loss. Thus, even with sufficient audibility, hearing impairment may alter allocation of cognitive resources for older adults listening to degraded speech.

4:20

1pPP9. Sensor-fusion to understand communication difficulty during conversations in noise. Kelly Miles (ECHO Lab., Macquarie Univ., 1.602, 16 University Ave., Australian Hearing Hub, Macquarie University, New South Wales 2109, Australia, kelly.miles@mq.edu.au), Ronny Ibrahim, Yvonne Tran, Alan Kan, and Joerg M. Buchholz (ECHO Lab., Macquarie Univ., Macquarie Park, New South Wales, Australia)

Difficulty communicating is the most challenging consequence of living with hearing loss, substantially affecting personal and professional relationships. While hearing devices help to redress this challenge, there is often a mismatch between performance measures obtained in clinical and laboratory settings and observed real-world behaviour. This discrepancy is likely due to an array of parameters, with the most notable being unrealistic speech stimuli (e.g., contrived sentence materials), artificial background noise, and tasks that do not reflect real-world communication behaviour or scenarios (e.g., sentence recall). To bridge this gap, we used sensor-fusion to understand communication difficulties in familiar communication partners engaged in natural, unrestricted conversations while listening to different levels of realistic background noise. We tallied communication breakdowns as a robust, overt metric of communication difficulty and fused data from an array of sensors including microphones, eye and motion trackers, and wearables that detect autonomic nervous system activity to objectively index communication difficulty. Our approach aims to find biomarkers that may predict the communication difficulties faced by individuals with hearing loss in the real-world. Ultimately, this research will contribute to enhancing the effectiveness of hearing devices, leading to improved social connection and quality of life for people with hearing loss.

4:40

1pPP10. Assessment of cochlear implant hearing outcomes using ecological momentary assessment in both controlled and real-world settings. Jessica J. Monaghan (National Acoust. Labs., 16 University Ave., Australian Hearing Hub, Sydney, New South Wales 2109, Australia, jessica.monaghan@nal.gov.au), David Meng, Marisa Poulos (National Acoust. Labs., Sydney, New South Wales, Australia), Zachary Smith (Cochlear Ltd., Sydney, New South Wales, Australia), and Jorge Mejia (National Acoust. Labs., Sydney, New South Wales, Australia)

Laboratory tests of speech understanding demonstrate high effectiveness of cochlear implants (CIs) for individuals with severe-to-profound hearing loss. However, understanding real-world benefits remains challenging. Retrospective questionnaires may be unreliable due to recall bias and limited acoustic information. To address this, we used an Ecological Momentary Assessment (EMA) smartphone app to collect subjective feedback and acoustic data from CI recipients in lab and real-world settings. We characterised challenging acoustic environments for CI users and the influence of internal factors on self-ratings of environmental noisiness, device benefit, and speech understanding. Twenty adults underwent cognitive tests, questionnaires, EMA, and the NAL Dynamic Conversation Test in the lab. Acoustic scenes were reproduced in an anechoic chamber to assess comprehension, familiarise participants with EMA, and establish benchmarks for real-world EMA data. Over four weeks, participants recorded experiences using the EMA app. Analysis of real-world EMA data revealed perceived noisiness was influenced by spectro-temporal properties and sound level. Self-reported communication factors were associated with environmental sound level and predictive of self-reported CI device benefit. Communication factors were also correlated with CI experience and cognitive performance. EMA offers valuable insights into CI recipients' real-world experiences, aiding customization and efficacy evaluation.

Session 1pSA

Structural Acoustics and Vibration: General Topics in Structural Acoustics and Vibration

Trevor W. Jerome, Cochair
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Benjamin J. Halkon, Cochair
University of Technology Sydney, 15 Broadway, Ultimo 2007, Australia

Contributed Papers

1:00

1pSA1. Sliding friction acoustics. Trevor W. Jerome (NSWCCD, 9500 MacArthur Blvd., Bldg. 3 RM 329, West Bethesda, MD 20817, twjerome@gmail.com), Maggie Kurdle (Univ. of Georgia, Athens, GA), and Anthony Bonomo (NSWCCD)

Contact interactions generate acoustic energy in complicated ways, dictated by dozens of system parameters. The ability to accurately predict these stochastic friction forces and resultant acoustic energy from first principles is currently lacking. A summary of experimental tests with sliding and stick-slip of metal-on-metal contact are presented to provide insight into the influence of roughness, sliding velocity, and normal load on the generated acoustics. Normal load and normal displacement increase in dynamic amplitude under stick-slip conditions relative to smooth sliding. In the current configuration, stick-slip also produces an excess of up to 30 dB SPL, relative to smooth sliding. Relationships between normal load, surface roughness, tangential velocity, and acoustic output are explored, and parameters are tuned to fit these variables using an existing empirical relationship from the literature.

1:20

1pSA2. An improved measurement of frequency-dependent shear properties of highly compliant materials using a dynamic Kibble's method. Nicholas Vlajic (Penn State Appl. Res. Lab., P.O. Box 30, State College, PA 16803, nav5000@psu.edu), Noah Robertson, and Benjamin Beck (Penn State Appl. Res. Lab., State College, PA)

While methods exist to characterize the Young's modulus and loss factor of viscoelastic materials, there are fewer methods to characterize the shear modulus and loss factor in shear. We previously presented an experimental apparatus that applied dynamic torque on rod-like viscoelastic materials with diameters of 0.5 cm to 2 cm and lengths of 10 cm to 30 cm. The torque was applied using Lorentz forces and measured using a dynamic version of Kibble's method (similarly, the Watt method). Here, we present a modified version of this apparatus that has several notable improvements over the previously reported version. Improvements include a stronger, more uniform magnetic field and a direct measurement of the angular response of the specimen using a laser and photo-diode configuration. Moreover, the direct measurement of the angular response eliminates the need to know other geometric quantities (such as the radius of the electromagnetic coil), which further reduces the uncertainty of the measured torque. The measurement of the angular response and applied torque are fit to models in order to determine the shear moduli and loss factor. We demonstrate this improved technique on rod-like specimens of different highly-compliant viscoelastic materials.

1:40

1pSA3. Impact localization on structures using remote sensors. Allison M. King (Mech. Eng., Univ. of Michigan, Ann Arbor, MI) and David R. Dowling (Naval Architecture and Marine Eng., Univ. of Michigan, 2600 Draper Dr., 215 NAME Bldg., Ann Arbor, MI 48109, drd@umich.edu)

Acoustic waves provide a wealth of information about their source and the environment in which they propagate. For large structures, it is sometimes desirable to use acoustic remote sensing methods to collect structure-borne sound because the microphone array and structure can then be maintained separately. Traditional time-of-flight array signal processing techniques used to localize acoustic sources are ill-suited for structures due to the potential for complicated propagation paths, the dispersive propagation of acoustic waves in structures, and the coupling of the vibrating structure and surrounding medium. Thus, source localization experiments were conducted using Matched Field Processing for a 6.4-mm thick circular aluminum plate excited by the impact of a 12.7-mm-diameter stainless-steel ball bearing dropped from 76-mm above the plate. This impulsive excitation of the plate contained frequencies between 0 and 5 kHz, resulting in wave speeds in the plate up to approximately 550 m/s. A linear array of 14 microphones with 51-mm spacing located 90-mm above the plate was placed in a random location such that it was not centered over the plate. Source localization results in both a quiet environment and an environment with additive white Gaussian noise are presented. [Sponsored by a SMART Scholarship, and the NEEC.]

2:00

1pSA4. Analyzing symmetry-based techniques to reduce the computation time of sound power estimates obtained from vibration measurements. Ian C. Bacon (Phys. & Astronomy, Brigham Young Univ., 333 W 100 S, Provo, UT 84601, icbacon@byu.edu), Micah Shepherd, Scott D. Sommerfeldt (Phys. & Astronomy, Brigham Young Univ., Provo, UT), and Jonathan D. Blotter (Mech. Eng., Brigham Young Univ., Provo, UT)

Two significant drawbacks have been identified in an emerging vibration-based sound power (VBSP) method when compared to its pressure and intensity-based competitors. Firstly, the measurement time increases significantly when dealing with large structures or higher frequencies. Secondly, sound power is computed through the matrix triple product $\mathbf{v_e}^{H}(\omega)\mathbf{R}(\omega)\mathbf{v_e}(\omega)$ which has expensive temporal costs as the structure becomes more complex and for dense scans. This work focuses on the latter concern pertaining to the computation time of the R matrix. Previous research has shown that leveraging the inherent symmetries arising from reciprocity between element radiators across the surface of a structure

expedites the construction of the R matrix. Additionally, theoretical work has revealed further symmetries in unbaffled structures, offering potential improvements in computational efficiency. To demonstrate this reduced computation time, simple case studies implementing the R matrix symmetries were conducted using experimental data from various unbaffled geometries. Furthermore, approaches to reducing the temporal costs of this specific matrix triple product, which exhibits these symmetries, will be investigated to enable real-time results within minutes after conducting dense scans. Reducing the computation time will contribute to the VBSP method becoming an efficient and reliable technique for measuring sound power in various applications. [Funding for this work was partially provided by the National Science Foundation (NSF).]

2:20

1pSA5. (Re-)exploring the normal modes of axisymmetric structures: An English church bell case study. Robert Perrin (Univ. of Technol. Sydney, Ultimo, New South Wales, Australia), Benjamin J. Halkon (Univ. of Technol. Sydney, 32-34 Lord StreetUTS Tech Lab, Botany, New South Wales, Australia, benjamin.halkon@uts.edu.au), and Zimu Guo (Univ. of Technol. Sydney, Botany, New South Wales, Australia)

The normal modes of axisymmetric structures are of interest to structural engineers, physicists and musical acousticians. Previously, some of the present authors have made studies of church bells, hand bells, elephant bells, various gongs and rings. Group theoretical arguments have been used, with considerable success, in classifying the normal modes of these structures and in understanding how "beats" arise from split degenerate doublets. It is now pointed out that further information can be obtained from group theory using a variety of additional arguments. In particular, it infers a basic distinction between "in-extensional" and "extensional" types of modes. In the present work, we concentrate on the case of an English church bell, as an example axisymmetric structure, whose normal modes were previously measured in a frequency range of up to about 10 kHz. In that earlier work, the results were analyzed with what was then considered a state-of-the-art finite-element package. We now repeat this exercise with a modern finite-element package to explore the differences between the types of modes and validate the Group theory observations. The agreement with experiment is much improved and some new level of understanding of the spectrum of the bell is achieved.

2:40

1pSA6. Correction of scanning laser Doppler vibrometer measurements when subjected to six degree-of-freedom base motion. Benjamin J. Halkon (Univ. of Technol. Sydney, 32-34 Lord StreetUTS Tech Lab, Botany, New South Wales, Australia, benjamin.halkon@uts.edu.au), Abdel Darwish (Univ. of Technol. Sydney, Ultimo, New South Wales, Australia), Steve Rothberg (Manchester Metropolitan Univ., Manchester, United Kingdom), Mahdi Mohammadi (Univ. of Technol. Sydney, Ultimo, New South Wales, Australia), and Sebastian Oberst (Ctr. for Audio, Acoust. and Vib., Univ. of Technol. Sydney, Ultimo, New South Wales, Australia)

Scanning laser Doppler vibrometer (SLDV) measurements are affected by sensor head vibrations as though they are vibrations of the target surface itself. Previous work has established a fully general theoretical analysis which shows that the only measurement required for measurement correction is of the vibration velocity at the incident point on the final steering mirror in the direction of the outgoing laser beam. Two practical—but not quite perfect—options for measurement correction were presented (one more suitable to manufacturer implementation, one more applicable to the vibration engineer end user). In both cases, placement of the correction transducer is critical with correction working for totally arbitrary instrument vibration and scan angle. Experimental validation, employing frequency-domain based processing, has been completed for one degree-of-freedom, on-axis vibration. Simultaneously, advancements in the data processing approach have realised improved correction in practice, especially for lower frequencies and for transient, as opposed to statistically stationary, vibration. In this paper, extension of the experimental validation to six degree-of-freedom instrument vibration is presented for the first time. In combination with the latest data processing approaches, reductions in the measurement error of 29.4 and 28.2 dB for the frequency- and time-domain processing techniques, respectively, are realised.

3:00–3:20 Break

3:20

1pSA7. A review of structure-borne noise from elevated rail structures. David Larner (Mott MacDonald, 707 Collins St., Docklands, Victoria 3008, Australia, david.larner@mottmac.com), Graham Brown (Mott MacDonald, Sydney, New South Wales, Australia), and Damian McGuckin (Pacific ESI, Glebe, New South Wales, Australia)

In the general case, noise from elevated rail systems consists of direct and reverberant airborne noise radiated from wheels and rails, noise from on-train sources, aerodynamic noise, and structure-borne noise reradiated from the viaduct or bridge structure. This paper presents a method of predicting structure-borne noise radiating from viaducts due to rail operations. The structure-borne noise is calculated by first determining the dynamic force input to the elevated structure at the rail attachment points due to excitation from wheel-rail roughness. Vibration transfer between structural elements, such as parapets and structural webs is estimated using Statistical Energy Analysis (SEA), which is also used to calculate the sound power radiating from the structure. The prediction of receiver sound pressure levels is carried out using simple noise propagation models that account for the geometric relationships between structural elements and receiver locations.

3:40

1pSA8. A study on the ground loss and foundation coupling loss for railway vibration prediction in greater Perth area of Western Australia. Ying Liu (Perth, SLR Consulting Australia, Subiaco, Victoria, Australia) and Luke Zoontjens (Perth, SLR Consulting Australia, Level 1, 500 Hay St., Subiaco, Western Australia 6008, Australia, lzoontjens@slrconsulting.com)

Railway noise and vibration have emerged as significant concerns affecting the well-being of residents living in proximity to railway tracks and the structural integrity of nearby infrastructure. As such, accurate and reliable predictions of railway vibration levels are necessary in order to properly assess the risks and develop effective mitigations. The propagation of railway vibration from the source to the receiving point involves a complex process, wherein ground loss and foundation coupling loss play crucial roles. Therefore, comprehensive capture of these factors is necessary for improving the accuracy of railway vibration prediction. This paper presents a study focused on the measurements of ground damping loss and foundation coupling loss in great Perth area of Western Australia. The measurements have been undertaken in generally sandy soil conditions. The foundation coupling loss was primarily measured at single-storey residential dwellings. The collected field data were subsequently analyzed to identify the characteristics of ground loss and foundation coupling loss. The study revealed that ground loss and foundation coupling loss exhibited variability dependent on the specific location, ground conditions, and structural characteristics of the dwellings. These measured results can serve as input for modeling efforts, thereby enabling more accurate prediction of railway vibration levels. Furthermore, the study provides valuable insights into the characterization of vibration propagation in sandy soils.

4:00

1pSA9. Structural stability analysis for a Maglev design. Zeyi Zhang (Jiangxi Univ. of Sci. and Technol., Ganzhou, China), Ying Hu (The Univ. of Hong Kong, Pok Fu Lam Rd., Central And Western District, Hong Kong, huying1226@connect.hku.hk), Zhe Zhang (The Univ. of Hong Kong, Hong Kong, Hong Kong), and Lixi Huang (The Univ. of Hong Kong, Hong Kong, China)

Magnetic levitation train has the advantage of low vibration and noise compared with ordinary railway systems. It also has the potential to become more energy efficient if permanent magnets are properly incorporated. However, suspension forces provided by the magnetic fields may have dynamic instability that needs to be overcome before such design can be utilized. This talk presents a design of permanent magnet group that overcomes the instability of at least one degree of freedom compared with rudimentary levitation configuration. The magnetic forces between components are derived analytically and the system eigen-value problem is presented. It is shown that, in the rudimentary design, unstable modes occur in two degrees of freedom (lateral and rotational), and the new design suppresses one of the two instability modes (rotational). Initial experimental demonstration is also provided.

4:20

1pSA10. Comparing user human comfort vibration on key commuter cycling routes in Melbourne Australia. Simon Ritchie (GHD, GHD, Level 9, 180 Lonsdale St., Melbourne, Victoria 3000, Australia, simon.ritchie@ghd.com) and Craig Evenden (GHD, Newcastle, New South Wales, Australia)

Cycling as an active mode of travel is recognised as an important part of the transport mix within Melbourne Australia. This city has a number of identified key commuter cycling routes in the vicinity of the CBD. Human comfort, including vibration level, is a factor in the selection of routes by an end user. This study aims to quantify the measured vibration experienced by a cyclist on a set of these main cycling routes informed by ISO2631-1 methods. Overall vibration values are then compared between routes. The results are combined with GPS position and speed data to enable mapping and visual route comparison.

4:40

1pSA11. Vibrational timber characterization through the use of model updating. Thien Tran (Ctr. for Audio, Acoust. and Vib. (CAAV), Univ. of Technol. Sydney, Ultimo, New South Wales, Australia), Can Nerse (Ctr. for Audio, Acoust. and Vib. (CAAV), Univ. of Technol. Sydney, 123 Broadway, Ultimo, New South Wales 2007, Australia, can.nerse@uts.edu.au), Sebastian Oberst (Ctr. for Audio, Acoust. and Vib. (CAAV), Univ. of Technol. Sydney, Ultimo, New South Wales, Australia), Benjamin J. Halkon (Univ. of Technol. Sydney, Botany, New South Wales, Australia), Nader Sawalhi (Ctr. for Audio, Acoust. and Vib. (CAAV), Univ. of Technol. Sydney, Ultimo, New South Wales, Australia), and Shahrokh Sepehrirahnama (Ctr. for Audio, Acoust. and Vib. (CAAV), Univ. of Technol. Sydney, Botany, New South Wales, Australia)

The characterization of orthotropic materials has challenged the vibration and acoustics community for quite some time. Complex composite materials such as wooden structures require attention to factors including moisture, grain boundaries in addition to macroscopic features. Here we devise a basic model developed by measuring the vibrational response in two separate axes to determine the material characteristics of a timber dowel. A proposed benchtop procedure utilises vibrometers and accelerometers to gather data before the updating process, for which, FEMtools was used. Based on the input material parameters, uncovered by previous studies, provide a starting point for the model updating procedure where experimental mode shapes and frequency responses are correlated to the finite element model. With the focus on radiata pine, the results show radial and tangential values converge similar to previous literature but with variation in the longitudinal direction and shear planes. Overall, this study provides a solid foundation to the characterization process of orthotropic materials like timber which can be further expanded into fields of structural health monitoring, damage detection and potential use in digital twins. The authors acknowledge the support of the Australian Research Council Linkage Project LP200301196.

Session 1pSC

Speech Communication: Phonetics of Under-Documented Languages II

Benjamin V. Tucker, Cochair
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Chair's Introduction—1:15

Invited Paper

1:20

1pSC1. Prominence and focus: The tension between acoustic cues and phonological structure. Sasha Calhoun (School of Linguist and Appl. Lang. Studies, Te Herenga Waka - Victoria Univ. of Wellington, PO Box 600, Wellington, N/A 6140, New Zealand, Sasha.Calhoun@vuw.ac.nz)

Across many languages, prosodic prominence signals focus. For example, "Does SHE like it?" has different implications to "Does she LIKE it?" (capitals indicate prominence), as the prominence marks focus and therefore a different implied contrast (with *she* versus *like*). In prosody research, there has always been a tension in the extent to which prominence is marked directly by psychoacoustic cues (higher pitch, loudness, length, etc.), which may be universal across languages, versus prominence being part of phonological structure, and therefore language-specific (e.g., Ladd, 2008). I review my research on the production and perception of focus in the understudied Polynesian languages Samoan and te reo Māori (Calhoun, 2015, 2016; Calhoun *et al*., 2021, 2023, in press), as well as other diverse languages (e.g. see Kügler & Calhoun, 2020). This research shows very clearly that cues to prominence vary substantially across languages and must be related to the phonological system of a given language. Nonetheless, questions about whether psychoacoustic cues directly signal focus, or its pragmatic sister, (paralinguistic) emphasis, alongside or within phonological structure, remain unresolved (e.g., Ladd and Arvaniti, 2023). I consider recent research offering new approaches to resolving this tension in relation to results from understudied languages.

Contributed Papers

1:40

1pSC2. Acoustic correlates of oral and glottal stops in Tahitian. Janet Fletcher (Univ. of Melbourne, School of Lang. and Linguist., University of Melbourne, Parkville, Victoria 3010, Australia, janetf@unimelb.edu.au), Adele Gregory (La Trobe Univ., Melbourne, Victoria, Australia), and Ben Volchok (Univ. of Melbourne, Melbourne, Victoria, Australia)

The Eastern Polynesian language Tahitian spoken in French Polynesia, is described as having a simple phonological inventory with a single stop series: /p, t, ʔ/. Little is known about the acoustic properties of these consonants compared to well-resourced languages. Five female speakers of Tahitian produced multiple repetitions of tokens that varied in terms of stress location and prosodic phrase position. Acoustic analyses of stop duration and VOT, as well as voicing measures including voicing fraction and strength of excitation were conducted. As predicted, oral stops were significantly longer than glottal stops by 63 ms in word medial contexts with no effect of stress. VOT values were on average 21–26 ms (with /t/ marginally longer than /p/) confirming these are short lag stops (Cho and Ladefoged, 1999). Stress was marginally significant with shorter VOT values in stressed syllable onsets with no affect of prosodic position, unlike Hawaiian (Davidson and Parker Jones, 2022). Like Hawaiian, however, (Davidson, 2021), the glottal stop in Tahitian has gradient realizations although the majority are realised with full glottal closure with low voicing fraction and strength of excitation values. There are also clear effects of greater coarticulation in unstressed versus stressed syllables suggesting localised hyperarticulation in the latter context.

2:00

1pSC3. An acoustic examination of Drehu vowels. Catalina Torres (Comparative Lang. Sci., Univ. of Zurich, Zurich, Switzerland), Weicong Li (The MARCS Inst., Western Sydney Univ., Sydney, New South Wales 2052, Australia, weicong.li@westernsydney.edu.au), and Paola Escudero (The MARCS Inst., Western Sydney Univ., Penrith, New South Wales, Australia)

Drehu is an indigenous language spoken in New Caledonia. Compared to other Oceanic languages, it has rather large vowel and consonantal inventories. The vowel system consists of 14 vowels, distinguishing seven vowel qualities and an additional length distinction. Previous phonological studies showed divergent proposals for two of the vowel qualities /ɛ/ and /ə/ based

on impressionistic descriptions. For this study, data was collected through a controlled reading task that was conducted during a fieldwork trip. We investigate the Drehu vowels for the first time, using acoustic data from eight native speakers (4 female). Vowel duration, formant values and dynamic formant properties were measured to examine the phonetic correlates of the proposed phonological vowel quality and length distinctions. In this study, we report seven timbres identified across the fourteen vowels of the system. Our results confirm a distinction between phonologically long and short vowels based on duration differences. We provide a detailed acoustic description of the vowel system and propose a revised vowel inventory based on a detailed analysis of formant structure of open-mid /ɛ/ and the central vowel /ə/.

2:20–2:40 Break

2:40

1pSC4. Articulatory and acoustic characteristics of occlusivized sonorants in Shakou Hakka. Jonathan Havenhill (Dept. of Linguist., The Univ. of Hong Kong, 930 Run Shaw Tower, Centennial Campus, University of Hong Kong, Pokfulam, Hong Kong, jhavenhill@hku.hk), Ming Liu, Robert Marcelo Sevilla, J. J. Perry, and Arthur Lewis Thompson (Dept. of Linguist., The Univ. of Hong Kong, Hong Kong, Hong Kong)

This study examines a pattern of sonorant occlusivization in a dialect of Hakka Chinese spoken in Shakou Township, Guangdong, China. This variety exhibits variable occlusivization of /l/ (realized as centralized [l^{d} ∼ ^{l}d ∼ d]) and denasalization of initial /m, ŋ/ (realized as prenasalized stops [?b, ?g]). Mid-sagittal and coronal ultrasound with synchronized audio, EGG, and nasalance recordings were collected from one speaker, along with acoustic recordings from 5 speakers. Partial occlusivization of /l/ occurred in 74.8% of words with high /i, y, u/ but does not occur in words containing low or mid vowels. Occlusivization of /l/ has an average duration of 42ms and is characterized by a significant decrease in mid-frequency acoustic energy (−6 dB, $p < 0.001$). Ultrasound data reveal tongue body raising and parasagittal contact during occlusivization, indicative of overlap between the tongue gestures required for /l/ (with lateral airflow) vs. /i, y, u/ (with lateral bracing). Denasalization, which has a similar mean duration of 39.5 ms, occurs in 42.9% of words containing non-high vowels and 86.25% of words containing high vowels. Although these patterns differ with regard to their phonological conditioning, gestural interaction during sonorant-vowel sequences may provide a common underlying mechanism for both processes.

3:00

1pSC5. Northern Lisu vowels & tones, and their interactions. Rael Stanley (La Trobe Univ., 7 Parklane Mews, Coburg, Victoria 3058, Australia, r.stanley@latrobe.edu.au), Marija Tabain, David Bradley, and Defen Yu (La Trobe Univ., Melbourne, Victoria, Australia)

Lisu is a Sino-Tibetan language of Central Ngwi branch. It has multiple tonal contrasts and a typologically unusual vowel inventory. In this talk, I present an acoustic phonetic study based on data collected from eleven speakers of Northern Lisu, spoken in Nujiang Autonomous Prefecture in Yunnan, Southwestern China. The study focuses on the vowels and tones of the language, and their interaction. Northern Lisu has six tones, including four modal and two creaky-voiced tones; it has ten vowels, including five front vowels, four back vowels, and a central "fricative" vowel. Tone and vowel interaction is analyzed using Esling's Laryngeal Articulator Model. Results show all speakers produce a lower f0 in the retracted vowel context, but it is not the case that all tones are creakier in this same vowel context. Simultaneously, male and female speakers show differences in terms of voice quality. Examination of the vowel space shows a reduction in many vowel contrasts. It is suggested that the vowel space is becoming regularised, with perceptually difficult contrasts being neutralised. In addition, it is shown that the fricative vowel contains minimal frication compared to the fricative consonants, and as such is better described as a syllabic retroflex approximant.

Invited Paper

3:20

1pSC6. On the small flat vowel systems of Australian languages. Andrew Butcher (Speech Pathol. & Audiol., Flinders Univ. of South Australia, Flinders University, GPO Box 2100, Adelaide, South Australia 5001, Australia, andy.butcher@flinders.edu.au)

The vowel systems of Australian languages could best be characterized as "small and flat." Small, because about 50% of Australian languages have a three-vowel system with another 20% or so distinguishing four or five vowel qualities. Flat, because, while there is the usual degree of separation between front and back vowels along the horizontal (F2) axis of the vowel space (700–2400 Hz), variation along the vertical (F1) axis is typically restricted to a range of 350-900 Hz, leaving the 'close' vowel area noticeably under-exploited. Thus, rather than the [i, a, u] values as traditionally transcribed, 3-vowel languages have the phonetic (Cardinal Vowel) qualities of [e, ɐ, o] and 5-vowel languages have [e, ɛ, ɐ, ɔ, o]. I suggest that, as with the atypical consonant inventories, the lack of close vowels in these languages may be explained in terms of the historical influence of the hearing status of their speakers. Reduction in vowel space in individual speakers is one consequence of hearing impairment caused by *otitis media*, which can reduce hearing levels under 500 Hz by up to 40 dB. The Australian Aboriginal population has the highest prevalence of *otitis media* in the world by far (50 to 90%).

Session 1pSP

Signal Processing in Acoustics: Signal Processing Poster Potpourri (Poster Session)

Manton J. Guers, Chair
Acoustics, Penn State Univ, PO Box 30, State College, PA 16804

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

Contributed Papers

1pSP1. An underwater sound event detection algorithm based on Temporal Graph Convolutional Network. Chong Hyun Lee (Ocean System Eng., Jeju National Univ., Jeju, Jeju Special Self-Governing Province, South Korea), Kibae Lee (Ocean System Eng., Jeju National Univ., Jejudeahakro 102, jejusi, Jeju 63243, South Korea, kibae0211@gmail.com), and Guhn Hyeok Ko (Ocean System Eng., Jeju National Univ., Jeju, South Korea)

Accurate underwater sound event detection (SED) and real-time underwater situation awareness plays an important role in intelligent sonar system. However, little research on underwater SED has been done because underwater acoustics have very complex and dynamic spectral and temporal characteristics which make underwater SED difficult to solve. The goal of SED is to recognize type and time of the input audio signal and most of SED research have been focused on airborne sound so far. To address underwater SED problem, we adopt temporal graph convolutional network (T-GCN), which is originally designed for traffic prediction and is composed of GCN and gated recurrent unit (GRU) To capture spectral and temporal information simultaneously, the GCN is used to learn topological structures to capture spectral correlation and the GRU is used to learn changes of input sound to capture temporal correlation. The proposed model utilizes sound spectrogram and its annotation with temporal information as input and then the GCN and GRU of the model is employed to solve underwater SED problem. Experiments using urban SED dataset and underwater sound dataset, demonstrate that our proposed model expresses connection of spectral and temporal information effectively and shows reliable underwater SED performance accordingly.

1pSP2. Research on vessel noise signal compression sensing algorithm based on Particle Swarm Optimization and Variational Mode Decomposition. Danni Wei (Ocean College, Zhejiang Univ., Dinghai District, Zhoushan, Zhejiang 316021, China, 22234088@zju.edu.cn) and Haocai Huang (Ocean College, Zhejiang Univ., Zhoushan, China)

Due to the complexity of marine environmental factors and strong background noise, the collected vessel signal becomes intricate and challenging to transmit and reconstruct using conventional parameter estimation methods. Additionally, identifying vessel targets becomes difficult. In order to detect the vessel target and obtain the corresponding orientation information, it is necessary to eliminate the interference of noise as much as possible and reconstruct the original signal more accurately. In this work, we propose an optimized signal denoising method PSO-VMD (Particle Swarm Optimization–Variational Mode Decomposition). By conducting simulations using experimental data and evaluating the model, we demonstrate the superior denoising capabilities of this method. Then we apply this method to denoise the real vessel sound signal and then compress and reconstruct the signal using compressed sensing. The experimental results show that the signal processed by PSO + VMD method has the maximum SNR and the minimum RMSE and MAE. What's more, the selected suitable denoising method can effectively suppress the environmental noise in the signal and improve the reconstruction efficiency of compressed sensing.

1pSP3. Tank test results of synthetic aperture sonar image reconsturction using two frequency bands. Sea-Moon Kim (Ocean and Maritime Digital Technol. Res. Div., Korea Res. Inst. of Ships and Ocean Eng., 32 Yuseong-daero 1312 beon-gil, Daejeon 34103, South Korea, smkim@kriso.re.kr), Pan-Mook Lee (Ocean and Maritime Digital Technol. Res. Div., Korea Res. Inst. of Ships and Ocean Eng., Daejeon, South Korea), Ji-Eun Lee (Acoust. Lab Co., Ltd., Seoul, South Korea), Kyungmin Baik (Korea Res. Inst. of Standards and Sci., Daejeon, South Korea), Chong-Moo Lee (Ocean and Maritime Digital Technol. Res. Div., Korea Res. Inst. of Ships and Ocean Eng., Daejeon, South Korea), Yeon-Seong Choo, and Jeong-Bin Jang (Korea Univ. of Sci. and Technol., Daejeon, South Korea)

Synthetic aperture experiments have been done in water tanks using two different frequency bands. Their center frequencies are around 25 kHz and 300 kHz, respectively. The lower frequency system has eight equally spaced receiver channels with one wavelength distance and the higher frequency one has sixteen receiver channels with about seven wavelength spacing. Five different types of structures were selected as sonar targets. They are sphere, cylinder, octahedron-shape structure, star-shape structure, and an in-plane grid target. The sphere and cylinder were floating in the water column while the others were located on the tank bottoms. As the transmitting signals linear frequency modulation was applied. Sonar images were reconstructed by synthetic aperture processing and the deviated motion from the perfect rectilinear trajectory due to the vibration of the vertical support was compensated by using motion sensor data and receiver signals. Synthetic aperture images were experimentally generated using the two different frequency bands and compared. [This research was supported by KIMST funded by the Agency of Korea Coast Guard (KIMST-20210547).]

1pSP4. A novel audio-visual based beamformer using a 360° camera and planar microphone array system. Hoang Phuc Phan (School of Eng., Macquarie Univ., 44 Waterloo Rd., Macquarie Park, New South Wales 2113, Australia, hoangphuc.phan@hdr.mq.edu.au), Ronny Ibrahim (MU Hearing, Macquarie Univ., Macquarie Park, New South Wales, Australia), Joerg M. Buchholz (MU Hearing, Dept. of Lingusitics, Macquarie Univ., Macquarie Park, New South Wales, Australia), and Alan Kan (School of Eng., MU Hearing, Macquarie Univ., Macquarie Park, New South Wales, Australia)

Beamforming systems that rely only on microphone arrays may suffer from poor performance in complex acoustic environments with multiple sources and room reverberation. To overcome this limitation, we propose a novel method that combines a planar microphone array and a 360° camera to steer the beamformer. The camera provides visual information about the location and orientation of sound sources, which can be used to adjust the beamformer parameters. We will present results that compare the signal-to-noise ratio (SNR) of the proposed method with a conventional beamforming microphone-based method in a simulated room with different source positions, orientations, and noise levels to investigate the benefit of visual information for beamforming in different environments. The proposed method

has the potential to improve the performance of beamforming systems in applications such as speech enhancement, speech recognition, and speaker localization.

1pSP5. A combined method for multi-channel active noise control based on online adaptive estimation and offline convolutional neural network. Qinxuan Xiang (Acoust. Lab., School of Phys. and Optoelectronics, South China Univ. of Technol., Guangzhou, China), Yijing Chu (State Key Lab. of Subtropical Bldg. and Urban Sci., South China Univ. of Technol., Guangzhou, China), Ming Wu (Univ. of Chinese Acad. of Sci., Guangzhou, China), and Guangzheng Yu (Acoust. Lab., School of Phys. and Optoelectronics, South China Univ. of Technol., Wushan Rd., Tianhe District, Guangzhou, Guangdong 421300, China, scgzyu@scut.edu.cn)

Multi-channel active noise control (ANC) has been widely used to attenuate low-frequency noise in relatively large spatial areas. Traditional multi-channel systems are achieved by optimizing the controller weights with adaptive algorithms that minimize the power of all error signals. However, nonlinearity in ANC systems can significantly degrade the performance of conventional adaptive algorithms, which is even more severe in multi-channel systems. Researchers suggested using neural networks (NN) to deal with the nonlinear problems. However, applying NN to multi-channel ANC systems is challenging in real-time processing because of the high computational burden. Therefore, a multi-channel ANC method that combines NN and adaptive method is proposed. The proposed controller consists of both linear and nonlinear parts, where the linear part is estimated adaptively to track the change of primary noise in real applications while the nonlinear part is modeled offline with a small-scale convolutional NN (CNN). This method has the advantages of solving the nonlinear problem in ANC systems and achieving the capability in tracking primary noise source positions. At last, the performance of the proposed algorithm is verified through simulation experiments.

1pSP6. Examining the voice verification system resistance developed for banking to attacks employing the voice cloning. Andrzej Czyzewski (Multimedia Systems, Gdansk Univ. of Technol., Narutowicza 11/12, Gdansk 80-233, Poland, andczyz@gmail.com)

The developed system for verifying the speaker identity in banking branches is based on voice biometrics employing DeepSpeaker neural network model. The testing of its resistance to a potential attack using voice cloning also employed neural networks, such as SV2TTS, Tacotron, WaveRNN, and GE2E. Subjective listening tests indicated that people might easily get confused and point out cloned recordings as an original sample. Nearly 50% of respondents pointed to the wrong answer, confusing the synthesized recording with the original one. Subjects in most cases declared that the quality of cloned and original recordings is similar (Good or Fair according to ITU-R BS.1534 recommendation) and, in some cases, even better than the original (graded as Good for synthesized sample, Fair for original one). Meanwhile, nearly all verification attempts with cloned samples failed (98.8% of samples were rejected). It proves that voice biometrics based on deep neural networks can identify cloned samples better than human listeners. Methods and results of testing the resistance of the developed voice biometrics system to voice cloning attacks are discussed in the paper. This research was funded from the budget of project No.POIR.01.01.01-0092/19 subsidized by the Polish National Centre for Research and Development (NCBR).

1pSP7. Event-driven online-learning using the CAR-FAC cochlea model. Ying Xu (Int. Ctr. for Neuromorphic Systems, The MARCS Inst., Western Sydney Univ., Kingswood, Locked Bag 1797, Penrith, New South Wales 2751, Australia, ying.xu@westernsydney.edu.au), Yeshwanth Bethi, Saeed Afshar, and van Schaik André (Int. Ctr. for Neuromorphic Systems, The MARCS Inst., Western Sydney Univ., Kingswood, NSW 2751, Australia, Penrith, New South Wales, Australia)

This work explores using a local spike-timing-dependent adaptation of thresholds and weights to learn to classify spectro-temporal representations of audio represented by neural spikes. We use the Cascade of Asymmetric Resonators with Fast Acting Compression (CAR-FAC) cochlea model and Leaky Integrated-and-Fire (LIF) neurons to generate the spikes from audio. This event stream is fed into the Optimised Deep Event-driven Spiking Architecture (ODESA) to learn spectro-temporal features in a hierarchical architecture. The cochlear events provide robust spectro-temporal representations of audio, and the ODESA performs online learning without needing error back-propagation or the calculation of gradients. Using simple local adaptive selection thresholds at each node, ODESA rapidly learns to appropriately allocate its neuronal resources at each layer in the hierarchy to learn features at different spatial and temporal scales. Information transmission throughout the system is event-based and all computing is performed asynchronously and online. We tested the approach on the TIDIGTIS benchmark and compare the performance with existing event-driven TIDIGTIS datasets and feature extraction and classification algorithms. With a three-layer ODESA, the proposed approach achieves 84.2% accuracy.

1pSP8. Automated Koala call detection using a cochlea model. Hadi Jamshidi (Int. Ctr. for Neuromorphic Systems, The MARCS Inst., Western Sydney Univ., Kingswood, New South Wales 2751, Australia), Ying Xu (Int. Ctr. for Neuromorphic Systems, The MARCS Inst., Western Sydney Univ., Locked Bag 1797, Penrith, New South Wales 2751, Australia, ying.xu@westernsydney.edu.au), Saeed Afshar (Int. Ctr. for Neuromorphic Systems, The MARCS Inst., Western Sydney Univ., Kingswood, Penrith, New South Wales 2751, Australia), Brad Law (Forest Sci. Unit, NSW Dept. of Industry-Lands, Parramatta, New South Wales, Australia), and van Schaik André (Int. Ctr. for Neuromorphic Systems, The MARCS Inst., Western Sydney Univ., Kingswood, Penrith, New South Wales 2751, Australia)

Estimating koala populations is challenging, making surveys costly and often imprecise. This study examines the feasibility of recognising koala calls from audio recordings using a bio-inspired acoustic approach. In the work, we propose to use the Cascade of Asymmetric Resonators with Fast-Acting Compression (CAR-FAC) cochlea model to pre-process koala recordings and a support vector machine (SVM) for classification. The frequency-dependent gain and bandwidth properties of the CAR-FAC model in pre-processing noisy data are discussed and investigated. To evaluate the effectiveness of the proposed method, it is compared with an FFT-based system as a baseline method. The baseline methodology uses a Mel-spectrogram pre-processing and the same linear classifier as the proposed method as the back-end. In summary, we achieved 96.7% accuracy for the proposed system.

1pSP9. Graph signal processing is combined with deep learning for detection of damaged wind turbine blades. Xiang Pan (College of Information Sci. and Electron. Eng., Zhejiang Univerity, No. 38 Zheda Rd., Xihu District, Hangzhou 310027, China, panxiang@zju.edu.cn) and Chenhui Zhang (College of Information Sci. and Electron. Eng., Zhejiang Univ., Hangzhou, China)

For early warning the damaged blade of wind turbines, an emission noise processing framework is proposed based on combination of Graph signal processing and Deep Learning. A microphone array is utilized to receive the noise emitted by the wind turbine blades. The weak abnormal signal from the damaged blade is enhanced by beamforming techniques. The enhanced signal is transformed into the graph domain by Graph Fourier Transform, from which the Mel filter bank features are extracted as inputs of a Multi-scale Feature Aggregation Conformer (MFA-Conformer) for damage detection. The MFA-Conformer combines Transformers and convolution neural networks (CNNs) to capture global and local features from the frequency or Graph domain. And the multi-stage aggregation strategy is utilized to exploit hierarchical context information. The reduction in the computational cost is achieved in the CNNs-based damage detection due to the real-valued features extracted from graph domain. The MFA-Conformer neural network is trained on the dataset which is created by applying data augmentation to the training samples. With the Mel filter bank features extracted from the frequency and graph domains, the MFA-Conformer neural network performs well in the five wind-farm data tests, with 2.55 % improvement in accuracy over the residual networks.

1pSP10. Performance evaluation on a convolutional neural network with adaptive interference cancellation for detecting a buried person's voice in a disaster situation. Jungyu Choi (Information and Commun. Eng., Soongsil Univ., 369, Sangdo-ro, Dongjak-gu, Seoul 06978, South Korea, cjg@soongsil.ac.kr) and Sungbin Im (School of Electron. Engr., Soongsil Univ., Seoul, South Korea)

In a disaster situation, rapid detection of victims is critical to prevent life-threatening accidents. However, it is not easy to detect the target's sound due to noise and interference which are difficult to predict. In addition, it is difficult to collect acoustic data in disaster situations. Therefore, this study proposes a target detection scheme in which untrained noise is added through a binary classification model trained only with a clean target signal and additive white Gaussian noise (AWGN). The feature uses the Mel spectrogram of the signal, and the model adopts a convolutional neural network (CNN). Two acoustic sensors are used in this study to adaptively remove environmental noise and a CNN to detect target signals. The first process trains a CNN using the target signal and AWGN data. In the next step, environmental noise is removed by compensating for the gain difference and delay characteristics between the two acoustic sensors through adaptive filtering. The input signal passes through preprocessing to eliminate noise through an adaptive filter. This signal is converted into a Mel spectrogram, which is then detected through a CNN model. The simulation results show that the model improves from 50% to 99% in detection rate at SNR 0 dB. In a catastrophe, there are many sounds that are difficult to discern. However, the model cannot learn from noise. Therefore, it shows that the target detection method using adaptive filter-based noise cancellation is effective in disaster situations.

Session 1pUW

Underwater Acoustics: Underwater Acoustic Propagation and Modelling

Robert Taylor, Cochair
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Thomas S. Jerome, Cochair
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Contributed Papers

1:00

1pUW1. Three-dimensional raytrace modeling of the New England seamounts acoustics experiment. Thomas S. Jerome (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, thomas.jerome@arlut.utexas.edu), Megan S. Ballard, Jason D. Sagers, Andrew R. McNeese (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), James H. Miller (Dept. of Ocean Eng., Univ. of Rhode Island, Narragansett, RI), and Julien Bonnel (Appl. Ocean Phys. & Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

The 2023 New England Sea Mounts Acoustics (NESMA) experiment took place from April to June at the Atlantis II Seamount Complex. One goal NESMA seeks to address is to further the understanding of acoustic propagation around prominent bathymetric features and associated diffraction and refraction effects. The bathymetry of the experimental site exhibits extreme variability with a difference in elevation of three kilometers from the abyssal plane to the peaks of the seamounts and slopes upwards of 15°, which has a significant influence on bottom-interacting acoustic propagation. As part of the experiment, multiple broadband impulsive sound sources were deployed around the seamounts, including explosive charges, seismic air-gun, and rupture induced cavitation events. Three-dimensional ray trace models of the experiment are presented for comparison with the measured acoustic receptions. Model results are primarily concerned with signals received on three single-channel acoustic recorders deployed on the abyssal plane near the seamounts. Comparisons provide insight into variation in the structure of the measured arrivals associated with varying ray paths refracted around and between the seamounts for different source positions. The influence of geotechnical properties on the measured signals is also assessed through modeling. [Work supported by ONR.]

1:20

1pUW2. Impact of internal waves on underwater acoustic propagation. Angus J. Coyle (Undersea Systems, Defence Sci. and Technol. Group (DSTG), Third Ave., Edinburgh, South Australia 5111, Australia, angus.coyle@defence.gov.au), Md Ayub, Daniel Boettger (Undersea Systems, Defence Sci. and Technol. Group (DSTG), Adelaide, South Australia, Australia), Manuel Cervera, and Andrew Mackinnon (Univ. of Adelaide, Adelaide, South Australia, Australia)

The propagation of internal waves in the ocean can produce significant fluctuations in the local sound speed field. Understanding how these fluctuations affect acoustic propagation is an area of considerable interest in underwater acoustics. Previous studies have indicated that large fluctuations (of the order of 20 dB) in transmission loss (TL) of acoustic waves can occur due to focusing and defocusing effects as the acoustic waves propagate through an internal wave. This work looks to extend some of these studies by exploring the frequency and directional dependence of these fluctuations through the implementation of a 3D acoustic model (FOR3D) suitable for modelling low frequency (<1 kHz) acoustic propagation. The study utilises sound speed data produced using the non-hydrostatic model MITgcm, simulating a nonlinear three-dimensional internal wave field that was verified using *in situ* mooring observations. By also considering results from a study utilising an acoustic ray model (Bellhop3D) on the same data, this work gives a comprehensive picture of the internal wave induced TL fluctuations across a range of acoustic frequencies.

1:40

1pUW3. Diurnal measurements of sound attenuation (100 Hz—10 kHz) in a shallow (1-2 m), seagrass (*Zostera marina*) meadow are impacted by photosynthetic activity and water column height. Emily B. Barosin (Thayer School of Eng., Dartmouth College, 0350 Hinman, Hanover, NH 03755, emily.b.barosin.25@dartmouth.edu) and Joseph D. Warren (School of Marine and Atmospheric Sci., Stony Brook Univ., Southampton, NY)

Seagrass meadows are important habitats that provide a variety of ecosystem services. As the seagrasses photosynthesize, they fill their vascular tissues with oxygen which is also released as bubbles into the water column. Measurements of sound attenuation were made in August 2022 in a shallow (1–2 m water depth) *Zostera marina* meadow in Shinnecock Bay, New York. Sound sources included tones (100 Hz–2 kHz) transmitted over an 8 hour period (0800–1600 local time) and a 10 kHz Fishtek pinger operating over a 3 day period. Hydrophones were placed in the seagrass meadow at ranges of 1, 3, and 7 m from the source to determine attenuation. Vertical profiles of temperature, salinity, and dissolved oxygen were measured at the study site during the 8 hour tonal source experiment, and solar irradiance and tide level for Shinnecock Bay were obtained for the 3 day study period. Peak to peak sound pressure levels varied by more than 31 dB over a diurnal period for the 10 kHz signals. Sound attenuation of lower frequency tones varied from 9 to 30 dB both with dissolved oxygen (a proxy for photosynthetic activity of the seagrass) as well as water column height.

2:00

1pUW4. Acoustic characteristics of the seismic airgun sound propagation in the East Siberian Sea. Dong-Gyun Han (Res. Ctr. for Ocean Security Eng. and Technol., Hanyang Univeristy ERICA, 55, Hanyangdaehak-ro, Sangnok-gu, Ansan, Gyeonggi-do 15588, South Korea, dghandg@gmail.com), Sookwan Kim (Korea Inst. of Ocean Sci. and Technol., Busan, South Korea), Martin Landrø (Norwegian Univ. of Sci. and Technol., Trondheim, Norway), Wuju Son (Korea Polar Res. Inst., Incheon, South Korea), Dae Hyeok Lee (Dept. of Marine Sci. & Convergence Eng., Hanyang Univ., Ansan, South Korea), Young Geul Yoon (Res. Ctr. for Ocean Security Eng. and Technol., Hanyang Univeristy ERICA, Ansan, South Korea), Jee Woong Choi (Dept. of Marine Sci. & Convergence Eng., Hanyang Univ., Ansan, South Korea), Eun Jin Yang, Yeonjin Choi, Young Keun Jin, Jong Kuk Hong, Sung-Ho Kang, Tae Siek Rhee, Hyoung Chul Shin, and Hyoung Sul La (Korea Polar Res. Inst., Incheon, South Korea)

A passive acoustic monitoring has been conducted by Korea Polar Research Institute (KOPRI) since 2017 in the East Siberian Sea using an autonomous passive acoustic recorder. Seismic airgun sounds, generated from the multichannel seismic survey system of the ice-breaking research vessel Araon, were measured from 18.6 km to 164.2 km in September 2019. During the survey, the R/V Araon maintained the ship's trajectory at a constant heading and speed to obtain useful acoustic data for investigating the acoustic propagation properties with a distance between the source and receiver in shallow water. The decrease in sound pressure levels as a function of distance was compared with the least-square regression curves and numerical propagation modeling based on the parabolic equation. The broadband sounds in a low-frequency range were propagated over long distances with distinctive characteristics such as precursor arrivals, modal dispersion, and rapid decrease in spectrum level at low frequencies. Our results will be discussed in comparison with the sediment structure and geological history of the measurement region. [This research was supported by Korea Institute of Marine Science & Technology Promotion (KIMST) funded by the Ministry of Oceans and Fisheries (20210605, Korea-Arctic Ocean Warming and Response of Ecosystem, KOPRI and 20210632, Survey of Geology and Seabed Environmental Change in the Arctic Seas, KOPRI).]

2:20

1pUW5. A simple formula for coherent specular reflection loss of underwater sound from rough ocean surfaces. Zhiyong Zhang (Defence Sci. and Technol. Group, DSTG EDN Site, Edinburgh, South Australia 5111, Australia, yong.zhang@defence.gov.au)

The interaction of underwater sound with rough sea surfaces affects its propagation. The interaction may involve several coupled mechanisms: (1) a rough air-sea interface scatters and spreads the incident sound energy over a range of angles, (2) entrapped sub-surface air bubbles absorb and scatter sound energy, and (3) medium of bubble clouds refract sound energy upwards. In this paper, we ignore the effect of bubbles and consider the effect of rough air-sea interface only. We further limit our consideration to the coherent component in the specular direction. Two simple closed-form formulas are often used to model the coherent reflection loss from rough ocean surfaces. One is derived from perturbation approximation and suitable for smaller grazing angles. The other is from Kirchhoff approximation and suitable for larger grazing angles. More accurate approximations over a wide-range of grazing angles require numerical evaluation of integrals or complex special functions. In this work, we simplify and adapt earlier results by other authors to obtain a simple expression that transits from the perturbation to the Kirchhoff formulas as grazing angle increases. The new simple analytical formula is easy to use and compares well with results from complex numerical methods.

2:40

1pUW6. Acoustic signature of strong deep-water currents in the vicinity of the New England Seamounts. Tsu Wei Tan (Marine Sci., ROC (Taiwan) Naval Acad., No. 669, Junxiao Rd., Zouying Dist., ROC Naval Acad., Kaohsiung 81345, Taiwan, ttan1@nps.edu), John Joseph (Oceanogr., Naval Postgrad. School, Monterey, CA), OLEG A. Godin, and Matthew Walters (Dept. of Phys., Naval Postgrad. School, Monterey, CA)

This paper presents some initial findings from the New England Seamount Acoustics (NESMA) Pilot experiment, focusing on the data obtained in April—June 2023 with three moored autonomous noise recorders (MANRs). Two MANRs were each equipped with a tilt current meter (TCM). Unexpectedly strong near-bottom currents were observed at depths of 2500m and 4443m, with the flow speeds reaching 110 cm/s and 80 cm/s, respectively. At 2500 m, recorded current speeds exceeded 20 cm/s approximately 5% of the time throughout the entire eight-week measurement period. Collocated ambient sound measurements revealed episodic large increases in the low-frequency noise intensity. A strong correlation is found between the noise intensity in the 2–20 Hz band and the measured current speed. As expected for flow noise, the intensity and the characteristic frequency of the power spectrum peaks increase with the increasing current speed. Observations of the flow noise by the MANR not equipped with a TCM indicate the presence and timing of strong current events and allow for an estimate of the current speeds at that location. The possible origins of the strong near-bottom currents and the flow noise effect on operation of deep-water acoustic systems will be discussed.

3:00–3:20 Break

3:20

1pUW7. The acoustic propagation environment surrounding the New England seamounts. Robert Taylor (Appl. Res. Labs, The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, rtaylor119@utexas.edu)

The New England Seamounts lie in a complex oceanographic environment that frequently sees both the colder Slope Sea waters and the warmer Sargasso Sea waters. The Gulf Stream, a meandering current that separates these two bodies of water, can be difficult to locate and frequently produces cold-core and warm-core eddies, complicating ocean modelling efforts. Regional temperature and salinity profile measurements gathered by the Argo program enable an investigation of the accuracy of the Hybrid Coordinate Ocean Model (HYCOM), a popular ocean circulation model. Simple range-independent calculations of acoustic propagation conditions provide a means of characterizing the acoustic environment by detailing common propagation paths associated with the New England Seamounts region. Contrasting the range-independent calculations for the Argo measurements with those of the oceanographic model indicates that persistent temperature features absent from the model impact predictions of the acoustic propagation environment.

3:40

1pUW8. Time spreading of a target echo from a short pulse in a waveguide. Dale D. Ellis (Phys., Mount Allison Univ., 18 Hugh Allen Dr., Dartmouth, NS B2W 2K8, Canada, daledellis@gmail.com)

The normal mode approach has proven very useful for calculating reverberation in a shallow water waveguide. Generally the calculation is done as a function of range r, then the time dependence t is obtained via $t = 2r/c$, where c is some average sound speed. This assumes that all the multipath returns from a particular scattering point arrive at the same time. A better time dependence can be obtained by using mode group velocities [D. Ellis,

J. Acoust. Soc. Am. **97**, 2804–2814 (1995)]. For reverberation from a uniform bottom this only has a small effect compared to the simple approach. However, for a compact scattering feature or target, the time spreading for a short pulse can be quite significant. Analytical results for an isospeed waveguide have been previously obtained using an energy flux approach. Here, a model using modal group velocities is used to calculate the time dependence of an echo from a target or compact reverberation feature. The mode approach extends to a general sound speed profile. One can convolve the impulse response with the amplitude envelope of the transmitted pulse to obtain the echo time decay. [Work supported by Mount Allison University, and US Office of Naval Research.]

4:00

1pUW9. Influence of the sound speed profile below the isothermal surface duct on the duct mode leakage rates. Alex Zinoviev (Sensors and Effectors, DSTG, West Ave., Edinburgh, South Australia 5111, Australia, alex.zinoviev@defence.gov.au)

The isothermal surface duct is an important feature in the sound speed profile (SSP) in ocean, as it allows acoustic signals to propagate to long distances. Usually, such a duct is modeled as two intervals of sound speed with respect to depth with N^2-linear SSP approximating linear SSP. The interval above the duct boundary has a positive sound speed gradient over depth, while the interval below has a negative gradient. At the duct boundary the gradient normally has a discontinuity. In this paper, based on the author's previously derived solution for the depth-separated Helmholtz equation in the environment with the N^2-exponential sound speed profile, he shows how this SSP can be used to model the smooth transition between the two intervals. Also, three SSPs with nearly identical sound speed between the duct boundary and the surface but with different profiles below the duct are considered, and energy leakage from the duct is calculated for the first three duct modes in a range of frequencies. It is shown that the shape of the sound speed profile below the duct can significantly affect the modal leakage rates.

4:20

1pUW10. Sensitivity of phase difference between short-range acoustic multipaths to changes in sound speed and source depth. Jacob P. DeFilippis (Oceanogr., Univ. California San Diego, 9500 Gilman Dr., La Jolla, CA 92093, jdefilip@ucsd.edu), Bruce Cornuelle, and William S. Hodgkiss (Oceanogr., Univ. California San Diego, San Diego, CA)

Ocean acoustic tomography often relies on precise clocks and knowledge of the range between acoustic instruments to estimate ocean-driven sound speed changes. In this study, we explore an alternative method utilizing the difference in the carrier phase between two broadband pulse arrivals to estimate small-scale changes in source depth and sound speed. By leveraging acoustic multipathing, this phase difference approach obviates the need to know the acoustic propagation's precise range and absolute travel time. Observations from a mid-frequency (1–10 kHz) acoustic experiment at short ranges (2–3 km) demonstrate the sensitivity of the difference in the 4 kHz carrier phase between two arrivals to changes in both source depth and sound speed. The difference in the arrival phase is modeled using measurements of the time-varying source depth and sound speed. The comparison between the modeled and observed arrival phase differences reveals a significant agreement, supporting the potential of using phase difference as a reliable estimator for perturbations in either source depth or sound speed for short-range acoustic propagation.

4:40

1pUW11. Efficient parabolic equation based travel time computation. Kevin D. Heaney (Appl. Ocean Sci., 11006 Clara Barton Dr., Fairfax Station, VA 22039, oceansound04@yahoo.com)

In this paper an efficient method for computing the travel time of an acoustic wave as a function of range using the parabolic equation model. The frequency derivative of the acoustic phase is the differential travel time associated with a propagation in range. By taking this difference across closely spaced frequencies (0.02 f_0) and integrating in range, this method computes the travel time of the dominant acoustic arrival. The method compares well with other travel time methods for four different cases, including deep water (shallow source, axial source), upslope and shallow water, and a global 3D propagation environment.

5:00

1pUW12. Acoustic transmission loss observed on a tomographic array in the Beaufort Gyre during 2016–2017. Peter F. Worcester (Scripps Inst. of Oceanogr., Univ. of California San Diego, University of California, San Diego, 9500 Gilman Dr., 0225, San Diego, CA 92093-0225, pworcester@ucsd.edu), Matthew A. Dzieciuch, Heriberto J. Vazquez (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA), John A. Colosi (Dept. of Oceanogr., Naval Postgrad. School, Monterey, CA), and Richard A. Krishfield (Woods Hole Oceanographic Inst., Woods Hole, MA)

The Arctic Ocean is undergoing dramatic changes. The 2016–2017 Canada Basin Acoustic Propagation Experiment (CANAPE) was conducted to assess the effects of the changes in the sea ice and ocean structure in the Beaufort Gyre on low-frequency underwater acoustic propagation and ambient sound. An ocean acoustic tomography array with a radius of 150 km that consisted of six transceivers and a long vertical receiving array measured the impulse responses of the ocean every four hours using broadband signals with center frequencies that ranged from 172.5 to 275 Hz. Ice-profiling sonar data showed a gradual increase in ice draft over the winter with daily median ice drafts reaching maxima of about 1.5 m, suggesting that the ice was first-year ice. The transmission loss of early, resolved ray arrivals from steep ray paths with lower turning depths below 500 m was lowest when open water was present and increased as the ice draft increased. The transmission loss per surface reflection increased with center frequency and surface grazing angle. The values are greater than observed during the 1988–1989 Greenland Sea Tomography experiment, but the ice conditions likely differed significantly.

Session 2aAAa

Architectural Acoustics and Noise: Airborne and Impact Noise in Buildings I

Benjamin M. Shafer, Cochair
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Invited Papers

8:00

2aAAa1. The use of professional audio external USB sound cards for building acoustics measurements. John L. Davy (School of Sci., RMIT Univ., Private Bag 10, Clayton, Victoria 3169, Australia, johnldavy@gmail.com)

CSIRO purchased an open application programming interface for hardware with an ethernet connection from a major acoustical instrument manufacturer. Using this interface, the author developed software for fractional octave band filtering with linear and exponential averaging. He then extended this software to automatically measure reverberation time. Fast Fourier Transform and Stepped Sine software was also developed. He then realized that CSIRO also owned two professional audio external USB sound cards. One of these had eight analogue inputs and ten analogue outputs. The other had two analogue inputs and two analogue outputs. Both could supply 48 V phantom power on their analogue inputs. It is possible to purchase phantom power preamplifiers for professional measurement microphones, but CSIRO decided to purchase phantom power to IEPE pre-amplifier adaptors. This enabled the use of professional pre-polarized microphones with IEPE microphone preamplifiers and accelerometers with built-in IEPE preamplifiers. CSIRO also had external USB hardware from another acoustical instrument manufacturer. Upon opening this instrument, it consisted of a professional audio external USB sound card and an IEPE front end for the two analogue input channels. The author decided to produce a version of his software that could be used with these three external USB sound cards.

8:20

2aAAa2. Effect of stud dimension and material properties on predicted and tested sound insulation. Wayland Dong (Paul S Veneklasen Res. Foundation, 1711 16th St., Santa Monica, CA 90404, wdong@veneklasen.com), Benjamin M. Shafer (PABCO Gypsum, Tacoma, CA), Sunit Girdhar, and John LoVerde (Paul S Veneklasen Res. Foundation, Santa Monica, CA)

A common wall design is gypsum wall board (GWB) cladding on each side of a row of steel studs, which may or may not include insulation. Steel studs are available in a range of metal thicknesses, dimensions, and structural properties. Most published laboratory testing on these walls uses a small subset of the available stud types, and the acoustical effect of changes to stud parameters has not been completely studied or documented. As a result, theoretical models of stud walls have not been evaluated over the full set of stud parameters. This work is a continuation of previous laboratory testing programs to systematically investigate the acoustical effects of stud properties. This paper analyzes the effects of stud depth and structural properties on third-octave transmission loss values and single-number ratings and investigates the capability of existing models for wall transmission loss to predict performance of walls with an expanded set of stud parameters. Our findings indicate that the parameters currently used to describe stud walls in acoustical test report are not sufficient to characterize the assembly.

8:40

2aAAa3. Acoustic design considerations for vertical education buildings. MD Nifarul Islam (Norman Disney & Young, Werribee, Victoria, Australia) and Kanvin Chen (Norman Disney & Young, 115 Batman St., Melbourne, Victoria 3003, Australia, k.chen@ndy.com)

In recent years, the design and construction of vertical (multi-story) education buildings are gaining preference over traditional single storied multi buildings precincts. This change is driven by the need to incorporate modern architectural design features, optimize spatial use, and construct sustainable, energy efficient buildings for education precincts. Vertical education precincts pose multiple acoustic design challenges, including consideration for adequate acoustic separation between noise sensitive adjacencies, noise emissions from rooftop mechanical plant to outdoor learning areas, and the provision of adequate reverberation treatment while aligning with the interior

design intent. Also, with the use of cross laminated timber (CLT) gaining in popularity, this presents challenges of its own. This paper discusses typical acoustic design challenges faced while designing vertical education precincts and provides recommendations adopted to resolve each design challenge.

9:00

2aAAa4. The booming party walls being constructed in the skyscrapers of Toronto. Jeffrey Mahn (National Res. Council Canada, 1200 Montreal Rd., Ottawa, ON K1A0R6, Canada, jeffrey.mahn@nrc-cnrc.gc.ca), Markus Müller-Trapet, Sabrina Skoda, and Iara Cunha (National Res. Council Canada, Ottawa, ON, Canada)

With eight new 300 m or higher buildings and another ninety shorter skyscrapers either under construction or proposed for the booming Toronto real estate market, there is no shortage of work for acoustic consultants in Toronto at the moment. Most of the high-rise buildings allocated for residential use include concrete floors and ceilings, glass curtain walls, and lightweight, steel stud gypsum board interior walls. A party wall design that has become popular among architects is a double steel stud wall with one or more layers of gypsum board attached to the studs in the gap between the rows of studs in addition to the gypsum board attached to the exterior of the studs. Typically, the transmission loss values for the wall designs used to demonstrate compliance with the building code are those for single stud walls with equivalent stud gauges and numbers of exterior layers of gypsum board. Due to the lack of available data, the National Research Council of Canada measured the transmission loss of many of these wall designs and found that adding the gypsum board in the cavity between the rows of studs has a significant effect on the transmission loss below 200 Hz.

9:20

2aAAa5. An investigation of the sound transmission class rating system using large-volume, high-variability datasets. Benjamin M. Shafer (Tech. Services, PABCO Gypsum, Tacoma, CA, ben.shafer@quietrock.com)

From shortly before, ASTM International introduced the E90-61T classification method in 1961 to the present time, the use of the sound transmission class (STC) rating system in North America expanded arguably more than any other building acoustics metric available today. From building codes to architectural specifications, the STC rating continues to connect acoustical engineers to building designers and occupants as a method for conceptualizing sound isolation in buildings, notwithstanding a valid evidence basis for significant flaws in the STC calculation methodology. Modern innovation in acoustic treatment introduces a far greater variety in achievable sound isolation performance than when the STC was first introduced. PABCO Gypsum has performed thousands of standardized sound transmission loss (STL) testing, providing a large and varied sample from which to observe trends. This paper utilizes this large-volume sample to compare tested assemblies with the same or similar STC ratings. STL data will be observed and compared to probable loudness level differences with the hope that this information will motivate further research and change in sound isolation classification methodology.

9:40–10:00 Break

10:00

2aAAa6. Recent progress in acoustic metamaterials with ventilation. Linus Yinn Leng Ang (Inst. of High Performance Computing (IHPC), Agency for Sci., Technol. and Res. (A*STAR), Singapore, Singapore) and Fangsen Cui (Inst. of High Performance Computing (IHPC), Agency for Sci., Technol. and Res. (A*STAR), 1 Fusionopolis way, #16-16 Connexis North Tower, Singapore 138632, Singapore, cuifs@ihpc.a-star.edu.sg)

To develop sustainable cities, natural ventilation is being explored as one of the ways to reduce energy consumption and carbon dioxide emissions. Recently, acoustic metamaterials with ventilation have garnered attention for their potential to provide both ventilation and noise control. In addition, this class of acoustic metamaterials can be used as a machine enclosure for dissipating thermal load and reducing noise emission. This presentation focuses on the recent progress in acoustic metamaterials with ventilation over the last five years, highlighting three types: metamufflers, metapanels, and metacages. Furthermore, the challenges faced by metacages for real-world applications are presented.

Contributed Papers

10:20

2aAAa7. A case study on flanking noise transmission through curtain wall in an office building. Mahbub A. Sheikh (Acoust., ACOR Consultants Ptd. Ltd., Unit 10, Level 1, No. 1 Maitland Pl., Baulkham Hills, Sydney, New South Wales 2153, Australia, msheikh@acor.com.au)

Curtain walls are a popular architectural element for the design of modern commercial building façade. With the advanced technologies and recent developments in building materials, curtain walls are developed with different lightweight materials, providing a spectacular visual identity of buildings and a number of functional design advantages to building interiors, including acoustic comfort, weather resistance, and thermal comfort. With the availability of high-performing glazed curtain walls these days, external noise in a commercial building adjacent to busy roads and railway lines is easily controlled, and acoustic comfort at internal space be achieved to suit the design intent. However, the lack of building design integration between building envelop, structural, and interior systems would often lead to airborne flanking noise transmission between different occupancies resulting in poor acoustic privacy. Such a post-design issue is often undesirable to the occupants and is an expensive exercise to retrofit. In this paper, a case study is conducted in a building where office workers in a three-story office building raised concerns about noise annoyance and acoustic privacy between two different tenancies on levels one above the other. This paper discusses the acoustic issues causing the compromised acoustic privacy, the method of acoustic investigations, the recommended noise control mechanisms, and finally the outcome of mitigation measures.

10:40

2aAAa8. Comparison of facade noise reduction measurement methods: Field results. Randy Waldeck (Resonance Acoust., 475 Sansome St., Ste. 800, San Francisco, CA 94111, randy@resonanceac.com)

Many projects require acoustical measurements of facade noise reduction be conducted to verify that Building Code, "green" rating systems (e.g., LEED), or other project interior noise criteria are met. In addition, federally funded airport sound insulation programs in the United States require facade noise reduction measurements be conducted to verify the eligibility of residences receiving sound insulation treatments. Currently, there exists a variety of standards to measure facade noise reduction (e.g., ASTM E966, ISO 16283-3, SAE ARP 6973). These standards employ various measurement methods: artificial exterior noise source, calibrated exterior noise source, line source, etc. In addition, these standards provide guidance on the location of exterior microphones (flush versus near facade) and correction factors to be applied to the data. Recently, a field comparison of three measurement methods was made at multiple residences: artificial exterior noise source, artificial interior noise source, and flush versus near microphone position. This is follow-on research from the National Academies of Science Airport Cooperative Research Project 02-51 and its purpose is to determine whether valid facade noise reduction measurements can be made with the loudspeaker located at the interior of the residence. A comparison of results across the measurement methods will be presented.

11:00

2aAAa9. Development on acoustic louvre to enhance the noise reduction performance of acoustic balconies. King Kwong Lu (Westwood Hong & Assoc. Ltd., 2404 Tung Wai Commercial Bldg., 109-111 Gloucester Rd., Wanchai 852, Hong Kong, kkiu@wha.com.hk), Maurice Yeung (ESEA Chapter, Acoust. Society of America, Hong Kong, Hong Kong), Kit Wong (Westwood Hong & Assoc. Ltd., Hong Kong, Hong Kong), and Fanni Lin (Westwood Hong & Assoc. Ltd., Hong Kong, Hong Kong)

A balcony in a domestic flat provides residents an outdoor living space as well as better ventilation to improve indoor air quality. However, balcony door when open for natural ventilation will allow outdoor noise transmit into the residential flat. Innovative acoustic balcony design by installation of sound absorptive panels onto the balcony ceiling and side walls of balcony has been implemented in Hong Kong for mitigating road traffic noise. Recently, combined balcony with an extension of utility platform for installation of air-conditioning outdoor unit has become popular. However, the parapet of the extension used a three-sided air grille with 70% permeability for heat dissipation purpose. Acoustically, the issue is noise leakage through the air grille comparing to a solid parapet. This paper discusses the development of acoustic louvres to replace the air grille in order to improve the noise reduction abilities of acoustic combined balconies. Several acoustic louvres of different configuration are developed. Validation of improvement in noise reduction of the acoustic combined balcony in a mock-up flat is conducted in accordance with ISO 16283-3.

11:20

2aAAa10. BIM-based sound insulation performance evaluation of interior walls. Youngmin Kwon (ENG Div., Samsung C&T, 26 Sangil-ro 6-gil, Gangdong-gu, Samsung GEC - Tower B, Seoul 05288, Republic of Korea, iam0min.kwon@samsung.com), Changhyun Kim (Bldg. & Residential BIM Group, Samsung C&T, Seoul, Republic of Korea), and Gyung Kim (BIM-Peers, Seoul, Republic of Korea)

As a general contractor, acoustical design confirmation of a project at construction stage is rather limited especially if it is large-scale. Among others, many interior walls of the building projects are found to be improperly designed in terms of sound insulation in that chances are hardly given for the most projects to be intervened by an acoustic expert at their design stage. In many cases, they turn out to be either underperformed or overperformed. This study developed a tool for holistically evaluating sound insulation design of interior walls across the building areas. It was designed on the basis of BIM modeling to be fast-tracked in supporting either the architects or general contractors for a quick and broad design review. Mapping of the information of room use and wall types with their predicted or measured sound insulation data on to a BIM model is set out to compare to the design criteria according to room use and adjacency. The results are simulated on both 2-D and 3-D BIM models, visually presenting the sound insulation performance. A report is instantly generated on performance qualification with the information of wall types, surface areas, etc. It also suggests alternative wall types as necessary.

11:40

2aAAa11. The effect of suspension systems on the sound insulation of suspended ceilings: A finite element analysis. Jesse Lietzén (Dept. of Acoust. Eng., AINS Group, Tampere, Finland), Ville Kovalainen (Dept. of Acoust. Eng., AINS Group, Turku, Finland), Lauri Talus (Dept. of Acoust. Eng., AINS Group, Tampere, Pirkanmaa, Finland), Mikko Kylliäinen (Dept. of Acoust. Eng., AINS Group, Tampere, Finland), Aitor Lopetegi (AMC Mecanocaucho, Pol. Zona A – parc.35, Asteasu 20159, Spain, alopetegi@amcsa.es), and Ander Aldalur (AMC Mecanocaucho, Asteasu, Spain)

It has been shown via laboratory measurements that the airborne and impact sound insulation of a concrete floor structure with a suspended plasterboard ceiling can be improved by using elastic ceiling suspension systems. The weighted airborne sound reduction index R_{w} was increased by 7 dB and the normalized impact sound pressure level $L_{n,w}$ was decreased by 15 dB when using elastic ceiling hangers as opposed to fixed hangers. In order to study the effect of elastic ceiling hangers on sound insulation further and to make it easier to compare different suspension systems especially in the low frequency range, a calculation model applying the finite element method (FEM) and parametric calculation methods was created. The calculation model was validated using measured data. The calculation results were then used to predict the improvement of sound insulation achieved with the different suspended ceilings. Additionally, the calculation model was used to examine the phenomena around the performance of the different ceiling hangers. The calculation results confirm the observations made from the laboratory measurements; switching from fixed ceiling hangers to elastomer ceiling hangers improved the performance of the suspended ceiling by more than 10 dB, with significant improvements beginning from the low frequencies.

Session 2aAAb

Architectural Acoustics: Student Design Competition

Robin S. Glosemeyer-Petrone, Chair
Threshold Acoustics, 141 West Jackson Boulevard, 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 nation's capital city wishes to construct a new 2,300-seat concert hall for a local resident symphony orchestra in a nation's capital city to present the full range of symphonic repertoire performances. The building will include a rehearsal room of the same area as the stage play area. The site for the concert hall is in an urban environment. The city block on which the project will be built is surrounded by high trafficked streets in all four cardinal direction and is used by emergency vehicles.

TUESDAY MORNING, 5 DECEMBER 2023 ROOM C2.3, 7:55 A.M. TO 12:00 NOON

Session 2aAB

Animal Bioacoustics and Underwater Acoustics: Session in Honor of Douglas H. Cato I

John Buck, Cochair
ECE, Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., Dartmouth, MA 02747

Michael Noad, Cochair
School of Veterinary Sci., The Univ. of Queensland, Gatton 4343, Australia

Chair's Introduction—7:55

Invited Papers

8:00

2aAB1. From ocean ambient noise to soundscape ecology. Douglas H. Cato (School of Geosciences, Univ. of Sydney, Sydney, New South Wales 2006, Australia, doug.cato@sydney.edu.au)

Ambient noise in the ocean has been studied for more than 80 years but recently there has been a tendency to use the word "soundscape" for what appears to be the same phenomenon. Ambient noise is usually defined as the background noise from all sources, excluding sounds from individual identifiable sources and is an important limitation on sonar performance and the use of sound by marine animals. The term "soundscape" has been used for decades outside of underwater acoustics with varying definitions depending on the application. Probably the most relevant is "soundscape ecology," which comes from terrestrial ecology and includes the acoustic interaction between animals and between animals and their environment. A soundscape, therefore, includes all sounds in an environment, not just the background or ambient noise. Sounds from individual identifiable sources may have particular interest to marine animals especially if they are from conspecifics, predators, or prey. This paper discusses the value of applying the broader concepts of soundscapes and soundscape ecology to the underwater environment, the advantages of recognizing the distinction between soundscapes and ambient noise and the importance of soundscape ecology in understanding animal acoustic behavior and the effects of anthropogenic noise.

8:20

2aAB2. Building bridges: Doug Cato's remarkable career spanning underwater acoustics and animal bioacoustics. John Buck (ECE, Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., Dartmouth, MA 02747, jbuck@umassd.edu)

Doug Cato's research includes pioneering studies of the underwater acoustic environment surrounding Australia and numerous contributions to our understanding of marine mammal acoustic behavior. Moreover, his relaxed avuncular manner allowed him to assemble collaborations spanning disparate disciplines while mentoring the next generation of acoustics researchers in Australia. Throughout his career, Doug built bridges between underwater acoustics and animal bioacoustics, and between Australian and international researchers. This talk highlights two international collaborations applying signal processing techniques to humpback whale songs and leopard seal calling bouts. These projects estimated the information entropy and structure of the sequences of sounds in these marine mammal vocalizations. Comparing sliding window entropy estimates with Markov models supports Doug's earlier observations that Australian humpback song follows the hierarchical structure proposed by Payne & McVay for North Atlantic humpbacks [Miksis-Olds *et al.*, JASA, 2008]. The same comparison of entropy estimators for leopard seal calling bouts reveals that most of the sequential structure of in these animals' calls is captured by Markov models.

8:40

2aAB3. Humpback whales increase the length of their songs during nearby airgun operations. Michael Noad (School of Veterinary Sci., The Univ. of Queensland, Gatton, Queensland 4343, Australia, m.noad@uq.edu.au) and Rebecca Dunlop (Univ. of Queensland, Brisbane, Queensland, Australia)

There is concern that noise from "airguns" used during oil and gas exploration may cause behavioral changes in marine mammals. One important behavior of some mysticete whales is the production of songs which are likely used as reproductive displays. Humpback whales have conspicuous songs but there are few studies on the impact of noise on their singing behavior. Here, we test the hypothesis that airgun noise causes individual humpback whales to reduce the duration of their songs. In a series of experiments off the east coast of Australia, we exposed migrating humpback whales to airguns. We tracked and recorded singing males when we towed airguns through the study area as well as when there were no vessels or airguns present. We also noted interactions between singers and nearby conspecifics. Humpbacks usually stop singing when they join with others, and so joining is both a predictor of song duration and a biologically relevant outcome. Contrary to expectations, whales that were already singing prior to airgun exposure produced significantly longer songs than unexposed singers. However, the small number of whales that started singing during airgun exposure produced significantly shorter songs.

9:00

2aAB4. Does female choice for song complexity drive sexual selection in humpback whales? Ellen C. Garland (Sea Mammal Res. Unit, Scottish Oceans Inst., School of Biology, Univ. of St. Andrews, St. Andrews, Fife KY16 9TH, United Kingdom, ecg5@st-andrews.ac.uk), Jenny A. Allen (Bio-Telemetry and Behavioral Ecology Lab., Inst. Marine Sci., Long Marine Lab., Univ. of California Santa Cruz, Santa Cruz, CA), Franca Eichenberger (Sea Mammal Res. Unit, Scottish Oceans Inst., School of Biology, Univ. of St. Andrews, St. Andrews, United Kingdom), Claire Garrigue (UMR ENTROPIE (Université de La Réunion, Université de la Nouvelle-Calédonie, CNRS, Ifremer, Laboratoire d'Excellence-CORAIL), IRD, Noumea, New Caledonia), Claire Bonneville (UMR ENTROPIE (Université de La Réunion, Université de la Nouvelle-Calédonie, CNRS, Ifremer, Laboratoire d'Excellence-CORAIL), IRD, Noumea, New Caledonia), Debbie Steel (Marine Mammal Inst. and Dept. of Fisheries and Wildlife, Oregon State Univ., Newport, OR), and Emma L. Carroll (School of Biological Sci., Univ. of Auckland - Waipapa Taumata Rau, Auckland, New Zealand)

Male humpback whales (*Megaptera novaeangliae*) sing a long, stereotyped, and culturally transmitted song display. While song likely functions in sexual selection, whether it is primarily directed at females and/or males is still debated. Most males within a single population sing the same, slowly evolving song type. However, multiple humpback whale song revolutions (where a song introduced from a neighboring population rapidly and completely replaces the existing song) have spread across the South Pacific region from the east coast of Australia to French Polynesia. How cultural evolution and sexual selection interact in this complex vocal display is not well understood. We investigated song-mediated female mate choice in humpback whales using 20 years of paternity data and song recordings from individually identified males in the small, South Pacific population of New Caledonia. As mating has never been observed in humpback whales, we assessed male reproductive success (calf paternity) and song complexity (unit repertoire and song length) of individual males. We found that males who successfully sired offspring had on average higher song complexity than males who did not sire offspring. This suggests females may select mates at least partially based on song complexity, and thus, song being primarily directed at females.

9:20

2aAB5. Air availability and expandable vocal tracts shape the duration of mammal calls. Tracey L. Rogers (Ctr. for Marine Sci. and Innovation, UNSW Sydney, E26 Biological Sci. UNSW, Botany Rd., Kensington, New South Wales 2052, Australia, t.rogers@unsw.edu.au), Jacob Dunn (Life Sci., Anglia Ruskins Univ., Cambridge, United Kingdom), Benjamin J. Walker (Evolution and Ecology Res. Ctr., UNSW Sydney, Sydney, New South Wales, Australia), Lucinda E. Chambers (Ctr. for Marine Sci. and Innovation, UNSW Sydney, Sydney, New South Wales, Australia), Kobe Martin (Evolution and Ecology Res. Ctr., UNSW Sydney, Sydney, New South Wales, Australia), and Andrea Ravignani (Dept. of Human Neurosciences, Sapienza Univ. of Rome, Rome, Italy)

The longest and shortest duration calls produced by any animal, spanning five orders of magnitude, are produced by whales. This is at odds with our understanding that the duration of an animal's call is driven by its size. In fact, we intuitively assume that large animals, with large vocal tracts, make long duration sounds, whereas small animals make shorter sounds. We compare the duration of the shortest and longest calls in the repertoire across 174 mammal species and show the relationship between body mass and vocalization length breaks down for mammals. For mammals that call underwater, those with modified vocal tracts (e.g., air sacs, expandable trachea, etc.) that move air between expandable reservoirs produce longer calls than predicted for their size. Those mammals without modified vocal tracts that call underwater produce much shorter calls than expected. We explore here the different evolutionary pressures that shape vocal signaling.

9:40

2aAB6. Cetacean audiogram models. Darlene R. Ketten (Biology, WHOI, WHOI, Woods Hole, MA 02543, dketten@whoi.edu), Andrew A. Tubelli (Eaton Peabody, MEEI, Boston, MA), and Aleks Zosuls (Biomechanical Eng., Boston Univ., Boston, MA)

Modelling is an established alternative to *in vivo* audiometry. We produced finite element models (FEM) of hearing in minke (*Balaenoptera acutorostrata*) and humpback (*Megaptera novaeangliae*) whales and frequency place maps (FPMs; total hearing ranges) for these species plus right (*Eubalaeglacialis*) and blue (*Balaenoptera musculus*) whales with control data for bottlenose dolphins (*Tursiops truncatus*) and harbor porpoises (*Phocoena phocoena*) Anatomically derived FPMs were obtained from microCT scans, basilar membrane histology, and nanoindentation stiffness measurements. FEM simulations of frequency response peak sensitivities used middle ear transfer function (METF) and laser Doppler measurements of stapes footplate velocities. Frequency responses were measured for stimuli at bone versus tympanic membrane/glove finger sites to assess most sensitive input locations. Odontocetes had a high pass response with corner frequencies from 10 kHz to 30 kHz. Response frequencies extended to 125 kHz, the limit of our measurement capability. Mysticetes had a bandpass response characterized by peak sensitivity from 0.5 kHz to 2 kHz for stimuli at the glove finger. Preliminary audiograms show peak responses differ among baleen whale species but were generally between 20 Hz and 5 kHz with *functional* hearing ranges of 50 Hz–35 kHz for tested species. [Work supported by the Joint Industry Programme on Sound and Marine Life.]

10:00–10:20 Break

10:20

2aAB7. Abandon FLIP! The following being a completely unembellished and sober account of a disastrous field experiment with Doug Cato, with side comments on his subtle influences on bioacoustic animal tracking. Aaron M. Thode (Scripps Inst. Oceanogr., Univ. of California San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0206, athode@ucsd.edu)

I have only been to sea once with Doug Cato, as a graduate student. It did not end well. In 1996 the R/P FLIP, a manned spar buoy, was deployed off the coast of San Diego to test advanced passive acoustic tracking methods on baleen whales, with Doug along as an invited guest. He had long been interested in acoustic means of localizing whales and was one of the first scientists to suggest using relative differences between received levels on hydrophones (instead of just the relative timing) for fixing an animal call's location [JASA **104**(3), 1667–1678]. This idea is the fundamental basis behind "matched-field processing" (MFP), the technique behind the FLIP test. The trip was supposed to collect three weeks of data; instead, it collected 42 h. This presentation will explain the challenges of abandoning a vessel at sea, and why garbage scows are underappreciated. It will also explain my brief involvement in Mike Noad's and Doug Cato's HARC field experiment as part of another MFP demonstration, and why that did not end very well either. Interlaced with these tales of disaster is a review of how Doug's thoughts on animal tracking have re-emerged under a variety of circumstances, especially in arctic bioacoustics.

Contributed Paper

10:40

2aAB8. Azimuthal, range, and depth tracking of marine mammal sounds using a drifting acoustic vector sensor and hydrophone array platform. Alison B. Laferriere (Marine Physical Lab., Scripps Inst. of Oceanogr., 9500 Gilman Dr., La Jolla, CA 92109, alaferriere@ucsd.edu), Aaron M. Thode (Marine Physical Lab., Scripps Inst. of Oceanogr., La Jolla, CA), Dieter A. Bevans (NUWC Keyport, Keyport, WA), Eric J. Berkenpas, Charles M. Shepard (Second Star Robotics, Silver Spring, MD), and Lauren A. Freeman (NUWC Newport, Newport, RI)

Deep-water acoustic tracking of marine mammals typically requires correlating hydrophone signals across both short and large-aperture hydrophone arrays. Here, we demonstrate how a single drifting, depth-controlled platform can obtain two and three-dimensional positional fixes of marine mammal sounds using an acoustic vector sensor and short-aperture arrays, both mounted on an autonomous drifting platform. In February and June 2023 two autonomous opto-acoustic drifters were deployed 50km off San Diego, CA at 250 m depth in 1030 m deep water. Both drifters were equipped with a CTD and acoustic recording system comprised of a 1.75 m aperture vertical hydrophone array, a tetrahedral hydrophone array, and either a 2-D Geospectrum M-35 or 3-D Wilcoxon VS-209 acoustic vector sensor. Throughout the two to three-day deployments, all acoustic sensors detected numerous marine mammal calls, including humpback whales and a pod of common dolphins. Estimates of azimuth and elevation were obtained from the simultaneously sampled acoustic vector sensor and hydrophone arrays, while multi-path and cross-platform processing were employed to extract range information. The results suggest that sparse deployments of drifting vector sensor platforms may be able to map biological distributions over spatial scales that would otherwise require larger numbers of hydrophone-only platforms. [Work supported by ONR TFO.]

Invited Papers

11:00

2aAB9. Multidimensional comparison of ocean soundscape properties across depth and substrate type. Jennifer Miksis-Olds (Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824, j.miksisolds@unh.edu), Dylan Wilford (Univ. of New Hampshire, Durham, NH), and Bruce S. Martin (JASCO Appl. Sci., Dartmouth, NS, Canada)

The local soundscape is determined by its physical environment, human activity, and presence of soniferous marine life and, therefore, provides insight on ecosystem processes, habitat quality, and biodiversity. Shallow coral habitats in tropical environments are hotspots of biodiversity and marine life. Deep-sea coral environments along the southeastern U.S. provide important habitat for a variety of marine animals but are generally poorly understood. An innovative Soundscape Code methodology for quantitatively characterizing soundscapes was applied to passive recordings from four locations along the US Outer Continental Shelf (OCS) and one from the Great

Barrier Reef to assess how differences in habitat, depth, and substrate manifest acoustically. Comparisons were made between (1) deep, cold-water and shallow, warm-water coral reefs and (2) deep-sea coral and sandy bottom habitats. The shallow, tropical reef soundscape differed from the deep-sea soundscapes in amplitude and impulsiveness. Differences in soundscape properties among the deep-sea soundscapes suggest deep-sea coral sites produce different soundscapes than the deep sites without live hardbottom. Biologic and anthropogenic signals influenced the deep-sea soundscapes and represent fundamental differences in habitat. This initial assessment of deep-sea soundscapes along the US OCS provides baseline acoustic properties in a region likely to experience changes in climate and human use.

11:20

2aAB10. Marine soundscape research—Are we making progress? Christine Erbe (Ctr. for Marine Sci. and Technol., Curtin Univ., GPO Box U1987, Perth, Western Australia 6845, Australia, c.erbe@curtin.edu.au)

Marine soundscape research includes, inter alia, ambient noise, marine animal bioacoustics, anthropogenic noise, sound propagation, and the effects of underwater noise. This research field is rapidly growing—in terms of its community, funding, and publications. Despite this almost exponential increase in activity, it might seem we are making too-slow progress, or in fact, running in circles. At every conference, we are hungry for new data, new research outputs. We aspire to see our research outputs translated into management outcomes and environmental impacts. But are we as a community making progress at a reasonable rate? Let us step aside from this race. Having worked with >60 peers on three ASA Press/Springer Verlag books on animal bioacoustics has given me the luxury to rejoice in the achievements we have made. It has, in fact, been a race to stay up-to-date with the literature, and regular, significant advancements have repeatedly required stepping back, reworking, updating, and editing. Join me in revisiting just some of the major developments from the last 20 years.

Contributed Paper

11:40

2aAB11. Vocalizations of two conspecific damselfish (Pomacentridae) at two Australian World Heritage sites: The Ningaloo Coast and the Great Barrier Reef. Juan Carlos Azofeifa Solano (Ctr. for Marine Sci. and Technol., Curtin Univ./The Australian Inst. of Marine Sci., 85 South Terrace, Fremantle, Western Australia 6160, Australia, eazofeifa2@gmail.com), Miles Parsons, Rohan Brooker (AIMS, Crawley, Western Australia, Australia), Robert McCauley, and Christine Erbe (Ctr. for Marine Sci. and Technol., Curtin Univ., Perth, Western Australia, Australia)

Novel methods to use Ocean Sound to address ecological questions are increasingly being developed but require optimization and validation to ensure their reliability. Source validation, increasing the number of known soniferous species of marine fauna, and characterizing their acoustic repertoires are important components of this effort. Coral reefs are remarkably complex, diverse, and noisy environments, inhabited by many soniferous species, hindering the association of vocalizations with the emitting species. Damselfish (Pomacentridae) are common and conspicuous reef fishes that have been relatively well-studied due to their abundance, territorial behaviors, and sound production. Most fish sound studies support that fish vocalizations are species-specific, often showing niche acoustic partitioning to avoid competition across frequency and time. It is imperative to record fish vocalizations in the field to avoid distortion in their acoustic attributes, which then can be used for further analyses such as automated detections. Here, we characterize the vocalizations of *Dascyllus aruanus* and *Dascyllus reticulatus* using field-experimental settings for *in situ* recordings from the Ningaloo Coast and the Great Barrier Reef. [Work supported by the BHP-AIMS Australian Coral Reef Resilience Initiative. JCAS was supported by Curtin University and BHP-AIMS in the form of a Curtin Strategic Scholarship and Top-up Scholarship.]

Session 2aAO

Acoustical Oceanography: Bubbles from Seeps

Elizabeth Weidner, Cochair
Marine Physical Laboratory, Scripps Institution of Oceanography, 7835 Trade St., Ste. 121, San Diego, CA 92121

Yoann Ladroit, Cochair
Ocean Science, Kongsberg Discovery, 23 Rue d'Anjou, Paris 75008, France

Grant B. Deane, Cochair
Marine Physical Laboratory, Scripps Institution of Oceanography, UC San Diego,
UC San Diego, 9500 Gilman Dr. #0206, La Jolla, CA 92093-0206

Alexandra Padilla, Cochair
Woods Hole Oceanographic Institution, Woods Hole, MA 02540

Chair's Introduction—8:55

Invited Paper

9:00

2aAO1. Measuring the sound of bubbles: Passive acoustic gas flux inversion. Ben Roche (ULB (Univ. libre de Bruxelles), 90 Heathwood Rd., Cardiff, Wales CF14 4BP, United Kingdom, benjohnroche@hotmail.com), Corentin Caudron (ULB (Univ. libre de Bruxelles), Brussels, Belgium), Paul White (ISVR, Southampton, United Kingdom), and Jonathan Bull (Univ. of Southampton, Southampton, United Kingdom)

There is an ever growing need to be able to detect and quantify marine gas seeps, both natural and man-made. One potential avenue for achieving this is via passive acoustic flux inversion. Such techniques use the frequency of oscillation of newly released bubbles (the Minnaert frequency) to determine the size (radius) and number of bubbles released from a seep in a given time. Passive acoustic flux inversion offers a number of advantages over active acoustics, namely, being low cost and energy making them ideal for long term monitoring of known sites. Recent studies have demonstrated how passive acoustic inversion can provided fresh insight into the variability of natural seeps, with changes being correlated to tides, surface seiches, and even day/night temperature cycles. Furthermore, we are now applying this technique to monitor volcanic activity, correlating changing rates of gas release with increasing volcanic activity, developing a new eruption monitoring tool. This talk will introduce the theory behind passive acoustic gas flux inversion, detail case studies and future projects with an open invitation for collaboration.

Contributed Papers

9:20

2aAO2. Evaluation of passive acoustic methods for ambient noise baseline and gas flow rate quantification at a proposed nearshore carbon capture and storage site in Australia. Haris Kunnath (CSIRO, GPO Box 1538, Hobart, Tasmania 7001, Australia, Haris.Kunnath@csiro.au), Najeem Shajahan (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada), Benoît Bergès (Wageningen Marine Res., Haringkade 1, 1976 CP IJmuiden, Netherlands), and Rudy Kloser (CSIRO, Hobart, Tasmania, Australia)

Measurement, monitoring, and verification (MMV) is an integral component of carbon capture and storage (CCS) projects. Within an operational MMV equipment, hydrophone-based passive acoustic techniques are used to establish ambient noise baseline and flow rate quantification at short range, specifically to facilitate "detect-attribute-quantify" sequence of an MMV program. However, nearshore environments are acoustically complex with different soundscape components that can disproportionately dominate ambient noise levels, potentially masking acoustic signatures of bubbles used to quantify seabed gas seeps. Therefore, a robust baseline describing ambient noise variability across the range of frequencies associated with acoustic emissions of gas seeps is required, from which changes can be detected and monitored. In this context, hydrophone measurements from a proposed nearshore CCS site in Australia are analyzed to establish a temporally resolved baseline, identifying key drivers causing overall ambient noise variability. These results are compared with acoustic bubble spectrum features and flow rate estimates from a controlled *in situ*gas release experiment to understand the likelihood of detecting bubbles and quantifying flow rate at the proposed CCS site. Despite the complexities of nearshore environment, the evaluation highlights that passive acoustic methods can provide a practical solution to complement quantification component of operational MMV programs.

9:40

2aAO3. Transient sounds of hydrothermal vents. Brendan Smith (Oceanogr., Dalhousie Univ., Halifax, NS, Canada, brendan.smith@dal.ca) and David Barclay (Oceanogr., Dalhousie Univ., Halifax, NS, Canada)

Transient acoustic signals were recently detected at the Main Endeavour Hydrothermal Vent Field which are believed to be generated by both geological and biological sources, including vent chimney collapse, impulsive geological signals, fish grunts, and snapping. These signals provide an opportunity for long-term passive acoustic monitoring of hydrothermal vent activity and ecology. This method offers advantages of longevity and robustness compared with other monitoring techniques, as the sensor can remain a safe distance away from the high-temperature, caustic vent fluid. Utilizing recordings from a bottom-mounted hydrophone on Ocean Networks Canada's NEPTUNE observatory, a detector was implemented to identify and classify these signals in more than one year of acoustic data after 2016. While only a single hydrophone is available at this site, an array of three seismic accelerometers also on the NEPTUNE observatory was used to localize transient events when sufficient signal-to-noise ratio was available to provide confidence in the location estimate. Correlation of the transient sounds with seismic activity at the vent field was also evaluated, suggesting that passive acoustic monitoring can augment seismic records to provide additional information regarding geological activity at hydrothermal vent sites.

10:00–10:20 Break

10:20

2aAO4. Direct measurements of gas flux at seep sites using a broadband split-beam echosounder. Elizabeth Weidner (Marine Physical Lab., Scripps Inst. of Oceanogr., 7835 Trade St., Ste. 121, San Diego, CA 92121, ereedweidner@ucsd.edu), Larry Mayer, and Thomas C. Weber (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, Durham, NH)

Measurement of methane gas flux from oceanic seep sites is challenging, given the ephemeral nature and the heterogeneous spatial distribution of seeps. Acoustic systems offer a solution, as they make synoptic measurements of the water column, and the gas bubbles are strong acoustic scatterers. Here, we present a method to directly estimate gas flux using a calibrated broadband split-beam echosounder. The vertical range resolution of the broadband system facilitates discrimination of individual bubbles in the acoustic record. By comparing measurements of bubble target strength to an analytical scattering model, bubble radii can be directly measured from the acoustic data. Concurrently, split-aperture processing allows for the precise tracking of bubbles as they rise from the seafloor for measurement of bubble rise velocity. Together, the observations of bubble radius and rise velocity offer a measure of gas flux, requiring nothing more than a vessel transiting over a seep site. Application of this method to seep data collected on the East Siberian Arctic Shelf shows good agreement between resulting measurements of bubble radii (0.68–8.40 mm) with those made using optical sensors, and bubble rise velocities (4–36 cm/s) are consistent with published measurements and models. Extrapolating from single bubbles measurements, our estimates of regional methane flux (2.9×10^4 kg/year) suggest that carbon emissions in this region may be lower than previously believed.

Invited Papers

10:40

2aAO5. Defining that *je ne sais QUOI* of seep sites: An overview of the Quantitative Ocean-Column Imaging (QUOI) voyage to offshore New Zealand. Elizabeth Weidner (Marine Physical Lab., Scripps Inst. of Oceanogr., 7835 Trade St., Ste. 121, San Diego, CA 92121, ereedweidner@ucsd.edu), Yoann Ladroit (Ocean Sci., Kongsberg Discovery, Horten, Norway), Vanessa Lucieer, Amy Nau (Inst. for Marine and Antarctic Studies, Univ. of Tasmania, Hobart, Tasmania, Australia), Sally Watson (National Inst. of Water and Atmospheric Res., Auckland, New Zealand), Geoffroy Lamarche (School of Environ. Sci., Univ. of Auckland, Auckland, New Zealand), Yves Le Gonidec (National Ctr. for Sci. Res., Univ. of Rennes, Rennes, France), Thomas C. Weber (Ctr. for Coastal and Ocean Mapping, Durham, NH), Peter Urban (Ghent Univ., Ghent, Belgium), Erin Heffron (Ocean Mapping Service, Portsmouth, NH), and Erica Spain (National Inst. of Water and Atmospheric Res., Auckland, New Zealand)

Detecting and characterizing oceanic seep sites is of considerable interest for the geoscience community due to their influence on global climate, ecological significance, connection to resources of high value, and, in many locations including New Zealand, their cultural importance. Modern acoustic systems provide the means for quantitative analysis of seep systems by collection of backscattered energy by gas bubbles and associated water column phenomena and surrounding seafloor. The July 2018 international, multi-institution Quantitative Ocean-Column Imaging (QUOI) voyage aimed to develop protocols and methodologies for identifying and quantifying seafloor and water-column backscatter data associated with bubble seep sites utilizing a large range of active acoustic systems and direct sampling methods. Here, we will provide an overview of QUOI voyage goals and research activities at several locations offshore of New Zealand, including the Calypso hydrothermal vent field in the Bay of Plenty. Gas bubble and hydrothermal fluid emission were captured in acoustic water column data, video transects, and direct sampling operations. More than 3000 individual seep bases were identified in the Calypso hydrothermal vent field, occupying approximately 9 of 115 km^2 of mapped seafloor. Efforts are underway to characterize the bubble size distribution of seep sites and quantify carbon flux to the atmosphere using a combination of broadband acoustic methods and *in-situ* measurements.

11:00

2aAO6. Multibeam sonar water column data simulation for improved and automated detection of gas seeps. Amy Nau (National Collections and Marine Infrastructure, CSIRO, Hobart, Tasmania, Australia, Amy.Nau@csiro.au), Vanessa Lucieer (Inst. for Marine and Antarctic Studies, Univ. of Tasmania, Hobart, Tasmania, Australia), Yoann Ladroit (Ocean Sci., Kongsberg Discovery, Horten, Norway), Haris Kunnath (Environment, CSIRO, Hobart, Tasmania, Australia), and Tara Martin (Antarctic Tasmania, Tasmanian Dept. of State Growth, Hobart, Tasmania, Australia)

Water column data (WCD) collected by multibeam echosounders (MBES) provide valuable datasets for the detection of gas bubbles. However, the presence of noise patterns, primarily caused by sidelobe interference from seafloor reverberation, limits the usability of WCD. We present a method for simulating characteristic noise patterns in WCD, particularly in Mills Cross configurations, by simulating predicted seafloor reverberation and transducer artefacts. The simulation is based on a reference pattern created from a subset of data and then applied as a correction based on the depth and seafloor backscatter of each ping within the dataset. Noise patterns can be removed by subtracting the simulated data from the original data, allowing for improved analysis of near-benthic water column features such as gas seeps. This method requires no prior knowledge of specific transducer characteristics, making it applicable to a wide range of MBES systems and acoustic targets. This method was applied to two MBES datasets collected simultaneously over a seep field in the Bay of Plenty, New Zealand, as part of the Quantitative Ocean-Column Imaging (QUOI) voyage in July 2018. We demonstrate the utility of applying a reference pattern for detecting the spatial extent of gas seep bases mapped using two different frequencies.

Contributed Paper

11:20

2aAO7. Estimating mixed gas fluxes with geochemically informed broadband acoustic methods: What the flux? Yoann Ladroit (Ocean Sci., Kongsberg Discovery, 23 Rue d'Anjou, Paris 75008, France, yoann.ladroit@km.kongsberg.com), Sally Watson, Sarah Seabrook (National Inst. of Water and Atmospheric Res., Wellington, New Zealand), Elizabeth Weidner (Marine Physical Lab., Scripps Inst. of Oceanogr., San Diego, CA), Thomas C. Weber (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, Durham, NH), Amy Nau (Inst. for Marine and Antarctic Studies, Univ. of Tasmania, Hobart, Tasmania, Australia), Vanessa Lucieer (Inst. for Marine and Antarctic Studies, Univ. of Tasmania, Hobart, Tasmania, Australia), and Geoffroy Lamarche (Parliamentary Commissioner for the Environment, Auckland, New Zealand)

Over the past few years, improvements in broadband split-beam echosounders have made it possible to obtain precise measurements of bubble-size distribution and density of gas seeps. Here, we model acoustic volume backscattering of a group of bubbles with a multi-modal size distribution that we matched to real-time acoustical measurement over a gas seep. The results are profiles of bubble size distribution descriptors (e.g., distribution type and parameters), density and ratio of the various components (i.e., gas composition). To test the ability of this methodology to differentiate gas flux composition and estimate gas flux magnitudes, we applied it in a semi-automated fashion on broadband data (12 to 250 kHz) acquired in the Bay of Plenty (NZ) over mixed gas seeps (e.g., CO_2, CH_4). Model outputs were compared with real-time dissolved gas measurements to constrain gas composition, magnitude, and ground-truth methodologies. The method can be applied to a variety of marine seeps to produce regional flux estimations, improving oceanic carbon budgets and our understanding of downstream feedbacks, such as localized deoxygenation or ocean acidification.

Session 2aBA

Biomedical Acoustics and Physical Acoustics: Biomedical Acoustics in Ophthalmology I

Jonathan Mamou, Cochair
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Tadashi Yamaguchi, Cochair
Chiba Univ., 1-33 Yayoicho, Inage, Chiba, 2638522, Japan

Invited Papers

9:00

2aBA1. Optical coherence elastography and Brillouin spectroscopy for tissue biomechanics. Kirill Larin (Univ. of Houston, 4800 Calhoun Rd., 3605 Cullen Blvd., Rm. 2028, Houston, TX 77204, klarin@uh.edu)

Quantifying the biomechanical properties of the different tissues can provide crucial information for disease detection and guiding precision therapeutic interventions. Optical coherence elastography (OCE) is an emerging technique to assess the mechanical properties of tissues completely noninvasively. Typically, elastography involves an external mechanical excitation of the tissue. We recently adopted the air-coupled acoustic radiation force method to generate elastic waves in tissues. This method shows several clear advantages other than the traditionally used air-puff excitation. Also, there is increased interest in passive elastography, where the mechanical response to natural physiological forces like heartbeat and respiration is measured. I will demonstrate the capability of OCE to measure tissue biomechanics in response to spatially varying fluctuations in IOP due to the heartbeat—thus, without any external force. Finally, I will demonstrate a few examples of the application of Brillouin spectroscopy to quantify tissue mechanical properties during Neural Tube Closure in mammalian embryos.

9:20

2aBA2. Flexion pulse wave in the retina: Toward à mechanical characterization of blood vessels? Gabrielle Laloy-Borgna (Univ. of Lyon, LabTAU, INSERM, Lyon, France), Stefan Catheline (Univ. of Lyon, LabTAU, INSERM, 151 Cours Albert Thomas, Lyon 69003, France, stefan.catheline@inserm.fr), Léo Puyo (Inst. of Biomedical Optics, Lubeck, Germany), Hidero Nishino (Dept. Sci. and Technol., Tokushima, Japan), and Michael Atlan (Service Ophtalmologie, Hopital des Quinze-Vingts, Paris, France)

The risk of cardiovascular events is linked to arterial elasticity that can be estimated from the pulse wave velocity. This symmetric wave velocity is related to the wall elasticity through the Moens–Korteweg equation. However, ultrasound imaging techniques need improved accuracy, and optical measurements on retinal arteries produce inconsistent results. After a quick historical review, it will be shown that the observation of an antisymmetric pulse wave, namely, the flexural pulse wave is possible. An optical system performs *in vivo* wave velocity measurements on retinal arteries and veins. Velocity estimation ranges between 1 and 10 μm per second. The theory of guided waves confirms the existence of this wave mode and its low velocity. Natural flexural waves can also be detected at the bigger scale of a carotid artery using ultrafast ultrasound imaging. This second natural pulse wave has great potential of becoming a biomarker of blood vessel aging.

9:40

2aBA3. Toward ultrasound-mediated elasticity modulation and monitoring in the crystalline lens. Maxime Lafond (Univ. of Lyon, LabTAU, INSERM, 151, cours Albert Thomas, Lyon 69424, France, maxime.lafond@inserm.fr), Alice Ganeau (Univ. of Lyon, LabTAU, INSERM, Lyon, France), Clément Foullounoux (Univ. of Lyon, LabTAU, INSERM, Lyon, France), François Legrand (Univ. of Lyon, LabTAU, INSERM, Lyon, France), Frédéric mascarelli (Biology, Eng., and Imaging in Ophtalmology (BiiO), Saint-Etienne, France), Philippe Gain (Biology, Eng., and Imaging in Ophtalmology (BiiO), Saint-Etienne, France), Gilles Thuret (Biology, Eng., and Imaging in Ophtalmology (BiiO), Saint-Etienne, France), Stefan Catheline, and Cyril Lafon (Univ. of Lyon, LabTAU, INSERM, Lyon, France)

Presbyopia is the age-related stiffening of the crystalline lens, reducing near vision. We propose ultrasonic cavitation to interact with the lens structure and restore flexibility by disrupting the disulfide bonds responsible for lens stiffening. Peak negative pressure of 25 MPa for 6-μs pulses at 2 MHz with a 100 Hz pulse repetition frequency allowed cavitation nucleation in the lens nucleus of both 6-months and 3-years old porcine eyes. Disulfide bonds disruption was evaluated in both ovoalbumin gels and *ex vivo* lens using Raman spectroscopy. We explored the measurement of surface wave dispersion over a 0.1 to 2 kHz frequency range to evaluate viscoelastic properties of the lens samples. We identified two regions in the dispersion curves: a sharp decline at low frequency due to guided waves and a gradual slope at high frequency, attributed to viscoelastic dispersion of a Scholte wave. By fitting the dispersion curves with a Kelvin–Voigt model, we found that while the impact of ultrasound remained uncertain, elastography allowed appreciating the age-dependent shear and viscosity moduli.

10:00–10:20 Break

Contributed Paper

10:20

2aBA4. Depth-resolved reconstruction of anisotropic elastic moduli in a crosslinked cornea with wave-based optical coherence elastography. Gabriel Regnault, Agathe Marmin, Ruikang K. Wang (Bioengineering, Univ. of Washington, Seattle, WA), Tueng T. Shen (Eye Inst. Harborview, Seattle, WA), Matthew O'Donnell, and Ivan Pelivanov (Bioengineering, Univ. of Washington, 616 NE Northlake Pl, Benjamin Hall Bldg., Rm. 363, Seattle, WA 98105, ivanp3@uw.edu)

Ectatic changes in the cornea are typically marked by corneal thinning with increased corneal deformation and curvature, often leading to high levels of myopia and irregular astigmatism. The most common form of ectasia is keratoconus. Corneal collagen crosslinking (CXL) is commonly used to prevent or treat keratoconus. Limited penetration of CXL into the cornea may lead to a demarcation between treated and untreated regions, suggesting a two-layer structure after the treatment. Although corneal topography can be used to monitor corneal curvature changes pre- and post-surgery, it cannot predict surgical outcomes without mapping corneal elasticity. Recent studies in non-contact dynamic optical coherence elastography (OCE) demonstrated promise in quantifying changes in corneal anisotropic stiffness induced by CXL. However, depth dependent reconstruction of corneal elastic moduli following CXL remains challenging. Here, we show that the thickness of the crosslinked corneal layer can be determined from structural OCT, whereas its mechanical moduli can be reconstructed from acoustic micro-tapping (AuT) OCE using an analytical two-layer model of guided wave propagation. In addition, we discuss how the elastic moduli of partially CXL-treated cornea layers reflect the effective engineering stiffness of the entire cornea to properly quantify corneal deformation.

Invited Papers

10:40

2aBA5. Acoustical imaging and myopia. Sally A. McFadden (The Univ. of Newcastle, University Dr., Callaghan, New South Wales 2308, Australia, sallyannemcfadden@gmail.com)

A new blindness epidemic has emerged, and it is predicted that by 2050, half the world population will have low vision from myopia (where images are blurred because the eyeball grows too long for its optical power) and 1 Billion people will be blind from pathologies associated with high myopia for which there are no viable treatments. We will discuss how high frequency ultrasonography was used in early studies to prove across species, including humans, that this excessive eye growth was caused by visual input related to our modern environments. Furthermore, aberrant visual exposure causes signals in the retina that induce the outer skin of the eye (sclera) to remodel, becoming excessively thin and weakened. High frequency ultrasound has also been essential to demonstrate that the biomechanics of the sclera are disturbed in complex ways during the development of myopia. For people who are at risk of developing the pathologies of high myopia, new treatments that aim to strengthen the sclera are in development. Due to its depth of penetration, we will introduce how acoustical imaging combined with other imaging modalities may pave the way for guiding more precise scleral treatments for high myopia.

11:00

2aBA6. Biomechanical and microstructural changes in posterior myopic Guinea pig. Sayantan Dutta (Dept. of Radiology, Weill Cornell Medicine, New York, New York, NY), Sally McFadden (Vision Sci., School of Psychol. Sci., College of Eng., Sci., and Environment, The Univ. of Newcastle, Callaghan, New South Wales, Australia), Quan V. Hoang (Singapore National Eye Ctr., Singapore Eye Res. Inst., Singapore, Singapore), and Jonathan Mamou (Dept. of Radiology, Weill Cornell Medicine, New York, 416 East 55th St., B1, New York, NY 1022, jom4032@med.cornell.edu)

The biomechanical and microstructural properties of the posterior sclera are drivers of myopia progression to high/pathologic myopia, a leading cause of blindness, affecting over 2.3 billion people worldwide. We investigated the effect of myopia on mechanical properties of the posterior sclera in guinea-pig (GP) eyes using scanning-acoustic-microscopy (SAM) with 3-μm spatial resolution. Form-deprivation (FD) was used to induce myopia in the right eyes of six GPs between 4 and 12 days of age, and 6-μm-thick scleral cryosections were scanned using a custom-made SAM. Bulk modulus (K) parameter-maps were estimated from SAM radiofrequency echo signals. We studied the effects of myopia on SAM maps in the nasal versus temporal regions of the posterior sclera. In the induced myopia eyes, K-values in the nasal region were 0.126 GPa lower than that in the temporal region ($p = 0.009$), whereas in the control eye, K did not significantly differ between regions ($p > 0.07$). Scleral biomechanical properties changed depending on eye region and induced myopia level. These results demonstrate that SAM-maps are important to elucidate myopia pathogenesis, providing insight to the location and timing of scleral stiffness reduction, as reflected in lower K-values in myopic eyes. [Work supported by the NIH grant R01GM143388 (J.M.), UON grant G2100649 (S.M.), and NMRC CIRG19nov-0030 (Q.V.H.).]

11:20

2aBA7. High-frequency point-of-care quantitative ultrasound to assess myopia-induced microstructural changes in the anterior sclera *in vivo*. Cameron Hoerig (Radiology, Weill Cornell Medicine, 416 E 55th St., MR-007, New York, NY 10022, cah4016@med.cornell.edu), Quan V. Hoang (Singapore Eye Res. Inst., Singapore, Singapore), and Jonathan Mamou (Radiology, Weill Cornell Medicine, New York, NY)

The progression of myopia to advanced stages is a leading cause of blindness worldwide. Standard ophthalmic measurements can evaluate myopia level but cannot predict disease progression. Recent evidence suggests the microstructural properties of the anterior sclera may be altered by myopia. We have developed a high-frequency (80MHz) point-of-care (POC) ultrasound device to collect

radiofrequency echo data from the anterior sclera and infer the microstructural and mechanical properties using quantitative ultrasound (QUS) techniques and passive elastography (PE), respectively. We hypothesized that QUS parameters derived from measurements of the backscatter coefficient (BSC) and shear-wave speed (SWS) estimated through PE correlate with myopia severity. Both eyes of 85 patients exhibiting varying levels of myopia were scanned with the POC device. Multilinear regression between the QUS parameters with refractive error or axial length demonstrated moderate correlation ($p < 0.001$). A logistic regression classifier trained using a subset of the QUS parameters to differentiate eyes based on myopia level achieved satisfactory performance as measured by area under the ROC curve (AUROC = 0.7). Results of this study suggest high-frequency QUS can detect myopia-induced microstructural changes of the anterior sclera and may pave the way toward a method to predict myopia progression *in vivo*.

11:40

2aBA8. Quantitative collagen fiber assessment in myopic guinea pig eye sclera using a cylindrical Gaussian form factor at 80 MHz. Kazuyo Ito (Tokyo Univ. of Agriculture and Technol., 2-24-16 Nakacho, Koganei, Tokyo 184-8588, Japan, itokazuyo@go.tuat.ac.jp), Quan V. Hoang (Singapore National Eye Ctr., Singapore Eye Res. Inst., Singapore, Singapore), Cameron Hoerig (Radiology, Weill Cornell Medicine, New York, NY), Sally McFadden (Univ. of Newcastle, Callaghan, New South Wales, Australia), and Jonathan Mamou (Radiology, Weill Cornell Medicine, New York, NY)

Myopia (also called near-sightedness) is an irreversible ocular abnormality that occurs when the eye is excessively elongated, resulting in defocused images and vision. When myopia progresses to pathological myopia, the possibility of permanent vision loss is extremely high. Therefore, it is important to identify early myopic eyes likely to progress to pathologic myopia and vision loss to start treatment and intervene as soon as possible. As myopia progresses, posterior eye elongation is caused by changes in the microstructural properties of the sclera including decreased collagen density and fibril diameters. High-frequency quantitative ultrasound (QUS) allows measuring biomechanical parameters associated with changes in tissue microstructure. This study investigated microstructural changes occurring in the posterior sclera in myopic guinea pig eyes using two independent QUS approaches at 80-MHz: envelope statistics quantification using the Homodyned-K distribution and backscatter coefficient quantification using a cylindrical Gaussian form factor. QUS parameters suggest changes in collagen fiber size occur in myopic sclera and is associated with microstructural changes in the scleral collagen fiber homogeneity and density. In conclusion, high-frequency QUS has the potential to quantitatively characterize microstructural changes that occur in the posterior sclera during myopia development and/or progression.

Session 2aCA

Computational Acoustics and Physical Acoustics: Data-Driven Methods in Acoustics and Vibration I

Marcus Maeder, Cochair
Technical Univ. of Munich, Boltzmannstrasse 15, Garching 85748, Germany

Johannes D. Schmid, Cochair
Chair of Vibroacoustics of Vehicles and Machines, Technical Univ. of Munich, Boltzmannstraße 15, Garching 85748, Germany

Chair's Introduction—7:55

Invited Papers

8:00

2aCA1. Maximum-entropy principle in knowledge-enhanced machine learning in acoustics. Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, xiangn@rpi.edu) and ZIQI CHEN (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY)

Recent acoustics investigations increasingly apply model-based methods, they purposely empower learning machines to analyze intrinsic parameters encapsulated in well-established models from experimental data. These data-driven analyses incorporate the well-understood /established models to the machine learning process as an important part of prior information, in addition to another part of prior information on the model parameters of interest. A data-driven machine learning method represents a learning process from data to update our knowledge, namely, learning from the data. This consistently represents the core of probabilistic inference within Bayesian framework on how one's prior knowledge is improved upon the experimental data. This paper discusses the data-driven machine learning methods in a way of thinking like a Bayesian. It emphasizes that knowledge-enhanced incorporation of the well-understood models in many learning processes can rigorously resort to the principle of maximum entropy, including Gaussian process due to their Bayesian nature. Benefits of the data-driven machine learning methods are highlighted when incorporating well understood/established models. This paper discusses physics-informed, knowledge-enhanced methods based on recent acoustic investigations, such as estimation of acoustic boundary conditions in wave-based numerical simulations and diffusion-equation-informed reverberation analysis.

8:20

2aCA2. Physics-informed neural network for predicting unmeasured ocean acoustic pressure field. Seunghyun Yoon (Seoul National Univ., 1, Gwanak-ro, Gwanak-gu, Seoul, Republic of Korea, 34-305, Seoul 08826, Republic of Korea, justin1128@snu.ac.kr), Yongsung Park, Peter Gerstoft (Scripps Inst. of Oceanogr., San Diego, CA), and Woojae Seong (Seoul National Univ., Seoul, Republic of Korea)

This study employs a physics-informed neural network in an ocean waveguide to predict the unmeasured acoustic pressure field, leveraging partially measured data. The method addresses a scenario where an acoustic source transmits signals across different ranges and is measured by multiple receivers. The acoustic pressure field in ocean waveguides exhibits rapid spatial variations over kilometer-range scales. The fully connected neural networks encounter challenges when approximating high-frequency functions, known as spectral bias. To mitigate this problem, the measured pressure field is transformed into a low-frequency function for training the neural network. We propose two methods sharing the same neural network architecture but utilizing different information. The first method uses a complex value of the pressure field (i.e., both magnitude and phase), while the second method uses only magnitude. We validate the proposed methods using simulations and experimental data from the SWellEx-96 environment. Results demonstrate that the first method exhibits superior performance with sparse data, while the second method works better in real-world scenarios.

Contributed Papers

8:40

2aCA3. Achieving stable convergence of neural networks for estimating acoustic field in uniform ducts. Veerababu Dharanalakota (Dept. of Elec. Eng., Indian Inst. of Sci. Bengaluru, 315, SPIRE Lab, CV Raman Rd., Bengaluru, Karnataka 560012, India, vrujntukkd@gmail.com), Ashwin R. Raikar, and Prasanta K. Ghosh (Elec. Eng., Indian Inst. of Sci. Bengaluru, Bengaluru, Karnataka, India)

The ability of the neural networks as function approximators can be exploited to solve several governing differential equations. In this work, 1-D Helmholz equation is solved to predict the acoustic pressure in a uniform duct. Solving the Helmholtz equation across a range of frequencies, especially at higher frequencies is challenging as the loss function destabilizes the training process, thereby preventing it from converging to the true solution with the desired accuracy. To overcome this issue, a dynamic learning rate technique is proposed that helps to stabilize the training process and improve overall accuracy of the network. The efficiency of the method is demonstrated by comparing the results with a static learning rate method and the analytical solutions. A good agreement is observed between the predicted solution with dynamic learning rate and the analytical solution up to 2000 Hz. Without dynamic learning rate, the relative errors are observed to be 2% and 58% at 500 and 2000 Hz, respectively, whereas they reduced to 0.6% and 0.1%, respectively, with the dynamic learning rate at the same frequencies. The proposed dynamic learning rate method is found to be effective for different types of boundary conditions.

9:00

2aCA4. Loss-based optimizer switching to solve 1-D Helmholtz equation using neural networks. Veerababu Dharanalakota (Dept. of Elec. Eng., Indian Inst. of Sci. Bengaluru, 315, SPIRE Lab, CV Raman Rd., Bengaluru, Karnataka 560012, India, vrujntukkd@gmail.com), Pavan K. J, and Prasanta K. Ghosh (Elec. Eng., Indian Inst. of Sci. Bengaluru, Bengaluru, Karnataka, India)

In this paper, 1-D Helmholtz equation is solved using physics-informed neural networks. In general, either the L-BFGS or ADAM algorithm is used to perform the optimization procedure. Unlike ADAM, L-BFGS is structured to reduce the loss function at each iteration. However, it becomes stagnant and fails to reach the global minimum at higher frequencies due to lack of momentum in the direction of the global minimum. On the other hand, ADAM has the advantage of added momentum. However, it requires a manual tuning of hyperparameters at each frequency to converge to the global minimum. Hence, one optimizer alone is inefficient to predict acoustic field at higher frequencies. This work proposes an algorithm called loss-based optimizer switching (LOS). This approach intelligently switches between L-BFGS and ADAM based on the specific criteria on the loss value to leverage the strengths of both optimizers. The performance of the LOS is evaluated by comparing the relative error between the predicted solution and the exact solution up to 3500 Hz. At 500 Hz, the relative errors with all three algorithms lie below 1.5×10^{-3}. However, at 3500 Hz, the relative errors are observed to be 6.6×10^{-3}, 0.92, and 0.28 with LOS, ADAM and L-BFGS, respectively.

Invited Papers

9:20

2aCA5. Simulation-based inference for the *in situ* estimation of the acoustic boundary admittance. Jonas M. Schmid (Chair of Vibroacoustics of Vehicles and Machines, Tech. Univ. of Munich, Munich, Germany, jonas.m.schmid@tum.de), Johannes D. Schmid, and Steffen Marburg (Chair of Vibroacoustics of Vehicles and Machines, Tech. Univ. of Munich, Garching, Germany)

Achieving accurate results in interior acoustic simulations relies on precise knowledge of the boundary properties of all interacting surfaces. Typically, the boundary admittance, which fully characterizes the acoustic properties of a surface, is determined under laboratory conditions such as the impedance tube. Yet, this approach has limitations, motivating the exploration of *in situ* methods to characterize materials in real-world conditions. In this work, we present a Bayesian approach to determine the acoustic boundary admittance *in situ* based on a limited number of measurement points. The method utilizes simulation-based inference, where a neural network is trained to approximate the posterior probability distributions of the unknown boundary admittances. The core of the approach is a finite element model used to generate sound pressure data, which also acts as the forward model during the inference process. Consequently, this technique is especially well-suited for applications involving pre-existing geometrical models, such as digital twin applications or model updating. By adopting simulation-based inference, we gain advantages over sampling-based Bayesian approaches, as it effectively handles complex and computationally expensive forward models.

9:40–10:00 Break

10:00

2aCA6. Advancing sound field analysis with physics-informed neural networks. Xenofon Karakonstantis (Elec. and Photonics Eng., Tech. Univ. of Denmark, Ørsteds Plads, Bldg. 352, Kgs. Lyngby 2800, Denmark, xenoka@elektro.dtu.dk) and Efren Fernandez-Grande (Elec. and Photonics Eng., Tech. Univ. of Denmark, Kgs. Lyngby, Denmark)

This work introduces a method that employs physics-informed neural networks to reconstruct sound fields in diverse rooms, including both typical acoustically damped meeting rooms and more spaces of cultural significance, such as concert halls or theatres. The neural network is trained using a limited set of room impulse responses, integrating the expressive capacity of neural networks with the fundamental physics of sound propagation governed by the wave equation. Consequently, the network accurately represents sound fields within an aperture without requiring extensive measurements, regardless of the complexity of the sound field. Notably, our approach extends beyond sound pressure estimation and includes valuable vectorial quantities, such as particle velocity and intensity, resembling classical holography methods. Experimental results confirm the efficacy of the proposed approach, underscoring its reconstruction accuracy and computational efficiency. Moreover, by enabling the acquisition of sound field quantities in the time domain, which were previously challenging to obtain from measurements, our method opens up new frontiers for the analysis and comprehension of sound propagation phenomena in rooms.

10:20

2aCA7. Deep-learning driven reverberation-aware microphone array speech enhancement. Feng Tong (Xiping Bldg., Xiamen Univ., Xiangan Campus, Xiamen, Fujian 361102, China, ftong@xmu.edu.cn), Wei He, Jiayang Zhang (Xiamen Univ., Xiamen, Fujian, China), and Yuanxun Kang (YeaLink, Xiamen, China)

Traditional microphone array speech enhancement algorithms suffer from significant performance degradation in complex and diverse speech application scenarios. The adoption of data-driven microphone array optimization models based on machine learning (deep learning) is capable of achieve significant performance improvements. However, such methods exhibit data dependency. A deep learning based reverberation-aware microphone array speech enhancement algorithm is proposed. The proposed algorithm first obtains the reverberation pattern through reverberation-aware environment probing, and then extracts the main component features to train reverberation-aware network (RAN). This operation enables the RAN to have the ability to decouple and generate adaptive de-reverberation beamforming coefficients based on reverberation features. Simulation and experiments are provided to verify the effectiveness and robustness of the proposed method in different reverbrant environments.

10:40

2aCA8. Impact on quality and diversity from integrating a reconstruction loss into neural audio synthesis. Yunyi Liu (Elec. and Information Eng., Univ. of Sydney, Maze Cres, Darlington, Rm. 840, J03, Eng. Bldg., Sydney, New South Wales 2008, Australia, yunyi.liu@sydney.edu.au) and Craig Jin (School of Elec. and Information Eng., Univ. of Sydney, Sydney, New South Wales, Australia)

In digital media or games, sound effects are typically recorded or synthesized. While there are a great many digital synthesis tools, the synthesized audio quality is generally not on par with sound recordings. Nonetheless, sound synthesis techniques provide a popular means to generate new sound variations. In this research, we study sound effects synthesis using generative models that are inspired by the models used for high-quality speech and music synthesis. In particular, we explore the trade-off between synthesis quality and variation. With regard to quality, we integrate a reconstruction loss into the original training objective to penalize imperfect audio reconstruction and compare it with neural vocoders and traditional spectrogram inversion methods. We use a Wasserstein GAN (WGAN) as an example model to explore the synthesis quality of generated sound effects, such as footsteps, birds, guns, rain, and engine sounds. In addition to synthesis quality, we also consider the range of sound variation that is possible with our generative model. We report on the trade-off that we obtain with our model regarding the quality and diversity of synthesized sound effects.

11:00

2aCA9. Physics-informed neural networks for characterization of structural dynamic boundary conditions. Johannes D. Schmid (Chair of Vibroacoustics of Vehicles and Machines, Tech. Univ. of Munich, Boltzmannstraße 15, Garching 85748, Germany, johannes.d.schmid@tum.de), Philipp Bauerschmidt, Caglar Gurbuz, and Steffen Marburg (Chair of Vibroacoustics of Vehicles and Machines, Tech. Univ. of Munich, Garching, Germany)

Structural dynamics simulations are often faced with challenges arising from unknown boundary conditions, leading to considerable prediction uncertainties. Direct measurement of these boundary conditions can be impractical for certain mounting scenarios, such as joints or screw connections. In addition, conventional inverse methods face limitations in integrating measured data and solving inverse problems when the forward model is computationally expensive. In this study, we explore the potential of physics-informed neural networks that incorporate the residual of a partial differential equation into the loss function of a neural network to ensure physically consistent predictions. We train the neural network using noisy boundary displacement data of a structure from a finite element reference solution. The network learns to predict the displacement field within the structure while satisfying the Navier–Lamé equations in the frequency domain. Our results show that physics-informed neural networks accurately predict the displacement field within a three-dimensional structure using only boundary training data. Additionally, differentiating the trained network allows precise characterization of previously unknown boundary conditions and facilitates the assessment of non-measurable quantities, such as the stress tensor.

11:20

2aCA10. Learning a surrogate Pekeris waveguide model. Jay C. Spendlove (Dept. of Phys. and Astronomy, Brigham Young Univ., C341 ESC, Provo, UT 84602, jayclark24@gmail.com), Mark K. Transtrum, and Tracianne B. Neilsen (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Underwater sound propagation models predict transmission loss (TL) given environmental parameters, such as seafloor sediment parameters. Also of interest is the reverse: Can seafloor parameters, such as sediment density, sound speed, and attenuation, be inferred from ocean acoustic data? This inverse problem can be addressed using information geometry tools for parameter identifiability analysis and model reduction, where we identify parameters that can be removed from a model without sacrificing accuracy. Critical to these methods for model reduction is the ability to evaluate derivatives of the model's TL predictions with respect to model parameters. Automatic differentiation (AD) allows for rapid evaluation of a model's derivatives but may face some challenges for implementation with sound propagation models, including models being written in "legacy code" and having non-differentiable points in the function space. Our solution is to train a surrogate machine learning (ML) model which can circumvent these challenges, and which we can apply AD to. We demonstrate this method using the Pekeris waveguide model of the ocean. We compare the surrogate and original model outputs and demonstrate the process of model reduction using a ML surrogate model. [Work supported by Office of Naval Research.]

Session 2aED

Education in Acoustics and Architectural Acoustics: Teaching Acoustics Across Trans-Disciplinary Areas

Lucky Tsaih, Cochair
Architecture, National Taiwan Univ. Sci. Technol., 43 Keelung Rd., Sec. 4, Taipei 106, Taiwan

Marion Burgess, Cochair
Univ. New South Wales, Australia, UNSW, Canberra 2610, Australia

Chair's Introduction—7:55

Invited Papers

8:00

2aED1. Challenges for flexible distance learning professional acoustics education. Marion Burgess (Univ. New South Wales, Australia, UNSW, Canberra, Australian Capital Territory 2610, Australia, m.burgess@adfa.edu.au)

A fully flexible, distance learning program comprising a number of modules has been available for those entering the acoustics consulting industry for over a decade. The flexibility requested by the association for the Australasian Acoustical Consultants (AAACs), means each of the modules can be commenced at any time and there are no firm deadlines for completion. The program has been very successful with satisfied students and employers. Clearly there is a great benefit for such flexibility as the industry does not recruit new staff at times that fit the University semester system. Hence, other options such as a microcredential approach are not always suitable. Also the students are able to adjust their progress in accordance with the varying work pressures of consultancies. However, the open approach means that there are challenges for the management of the program. These include always being available for processing new applications, supporting and guiding current students, undertaking the marking as well as updating the content, assignments and tests. This paper will discuss the challenges and the changes that have had to be made and those that will need to be introduced to ensure the long term viability of the program.

8:20

2aED2. What does learning sound like? A STEM outreach experience. Matthew Ottley (Marshall Day Acoust., C14 372 Wattle St., Ultimo, New South Wales 2007, Australia, mottley@marshallday.com) and Fiona Young (Hayball, Surry Hills, New South Wales, Australia)

What an acoustic engineer does is generally not well understood by high school students, and this likely applies to the broader architecture and design fields. Acoustic engineers and architects who are engaged in the design of high schools and classrooms also may not have been in a class of students since they left school. This presentation outlines a Problem Based Learning experience run for high school students in Australia which provided insight into the above gaps for both students and practitioners. The student-centred challenge has been run in two formats. The first was run weekly over three 100 minute sessions, while a condensed format was later run as an 80 minute hands-on design challenge. Both formats involved an architect(s) and an acoustician facilitating the challenge. The students considered the use, acoustic requirements and affordances of design in a classroom. Students were given a classroom with poor acoustics and challenged to develop and present a design solution which improved the area for learning, including room acoustics. The challenge involved mathematics (Sabine equation), measurement (including scale and reverberation time), and architecture.

8:40

2aED3. Architectural acoustics for non-acousticians. K. Anthony Hoover (McKay Conant Hoover, 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362, thoover@mchinc.com)

The great majority of professionals who design our everyday environments, as well as our special spaces, have a very limited understanding of acoustics. Those who may be interested in learning more about acoustics must overcome the vast oceans of misinformation available on the web, and often feel "allergic" to math and physics. This presentation will focus on the methods and approaches for providing useful information that they can use and remember, in a meaningful and organized manner, as based on decades of "brown bag" lunches, meetings, professional society presentations, webinars, short courses, and 3-credit classes.

9:00

2aED4. Prediction of psychological impacts of noise from ventilation systems. Kuen Wai Ma (The Hong Kong Polytechnic Univ., 11 Yuk Choi Rd., Hung Hom, Hong Kong, kuen-wai.ma@polyu.edu.hk), Cheuk Ming Mak, Fu Lai Korris Chung, and Hai Ming Wong (The Univ. of Hong Kong, Hong Kong, Hong Kong)

Air-conditioning ventilation systems are essential for maintaining good indoor environmental quality. However, the noise generated by these systems can have negative psychological impacts on occupants, such as dissatisfaction, discomfort, disturbance, and unacceptability. This problem cannot be solved simply by reducing the noise level. With the help of a valid, reliable, and applicable psychometric tool called the psychoacoustics perception scale (PPS), the psychological impacts of the noise on human general judgment (Evaluation, E), sensitivity to the magnitude (Potency, P), and sensation of the temporal and spectral compositions (Activity, A) of sounds can be quantified. A holistic sound quality assessment was proposed to cover the objective assessments of traditional indoor criteria and acoustic metrics, as well as subjective assessment using the PPS. The correlations found between the objective characteristics of the noise from ventilation systems, the PPS scores, and the occupants' cognitive performance can extend traditional noise level prediction to the prediction of psychological impacts of air-conditioned environments. This advanced knowledge of noise prediction will help acoustic professionals in the design of teaching courses in acoustics, noise, and vibration control, as well as in building services engineering, and throughout the built environment.

9:20

2aED5. Soundscape exploring through choral performance and music creation. Wei-Chun Wang (Dept. of Humanities and Social Sci., National Taiwan Univ. of Sci. and Technol., No. 43, Sec. 4, Keelung Rd., Taipei 106335, Taiwan, vgnwang@mail.ntust.edu.tw)

With a profound concern for the local environment and culture in Taiwan, this project integrated collected soundscape clips and mobile devices to compose and perform a cross-domain audiovisual soundscape choral concert. Drawing inspiration from M. Schafer's soundscape concept and music education philosophy, as well as the "Three-S Model" of soundscape—sound, setting, and significance, the project aimed to develop university students' performance and music creation skills and enhance their perception of soundscapes. Employing the concept of choral theater, the concert creatively combined collected soundscape recordings and imagery from various sites, presenting them through a multidisciplinary theatrical framework. This approach led to a reinterpretation of soundscape choral repertoire, shedding light on their historical context and contemporary relevance. Through quantitative and qualitative analysis of feedback from students, it is found that the integration of visual and auditory elements allowed the audience to have a more enriching sensory experience. Moreover, university student performers gained a heightened sense of observation and analysis of their surroundings through their involvement in specialized creations and performances. This experience allowed them to authentically connect with the essence of world music and local folk songs, fostering a deeper connection with the cultural heritage.

9:40–10:00 Break

Contributed Papers

10:00

2aED6. Teaching acoustics across multi-disciplines. David A. Brown (ATMC/ECE, Univ. Massachusetts Dartmouth, 151 Martine St., Fall River, MA 02723, dbAcousticsdb@gmail.com)

Vibration, sound, and acoustics fundamentals appear in many disciplines and at all stages of life. This presentation covers acoustic concepts and foundations applicable in many professional disciplines that may be suitable for short professional courses, undergraduate, and graduate courses with a particular focus in engineering and underwater acoustics although the lessons may apply to music, noise, bioacoustics, audio, and electromagnetics. Topics include frequency, resonance, quality factor, damping, sensors, transducers, sensitivities, beam patterns, decibels, intensity, power, traveling and standing waves, speed, waveguides, dispersion and more.

10:20

2aED7. Ultrasonic testing for metal additive manufacturing: Experience with new course development. Joseph A. Turner (Mech. and Mater. Eng., Univ. Nebraska-Lincoln, Lincoln, NE), Luz D. Sotelo (Purdue Univ., 585 Purdue Mall, West Lafayette, IN 47907, lsotelo@purdue.edu), and Nathanial Matz (Mech. and Mater. Eng., Univ. of Nebraska-Lincoln, Lincoln, NE)

Metal additive manufacturing (AM) has brought about the need for nondestructive evaluation (NDE) methods to assess part quality. AM processes result in complex microstructures, porosity, material texture, and residual stresses all of which may vary spatially throughout the part. Thus, knowledge of ultrasonic NDE methods to characterize materials is relevant to the AM community. To address this challenge, a new graduate-level course entitled "Ultrasound for Metal Additive Manufacturing," was created at the University of Nebraska-Lincoln. The course followed a cognitive-situative blended learning approach to give students practical ultrasonics training regardless of prior acoustics background. The theory introduced key topics including: the general wave equation, plane wave solutions, waves in isotropic solids, impact of material anisotropy, role of material interfaces, reflection and transmission, surface waves, scattering, dispersion, material dissipation, and scattering attenuation. A series of laboratory experiments allowed students to learn: experimental setups, signal processing, transducer properties and selection, beam mapping, wave speed and attenuation measurements, and diffuse-field measurements. Finally, each student developed a project incorporating ultrasound measurements into their individual AM research. Their findings were included in research articles, conference presentations, and PhD dissertations. This presentation will describe the course design, successes, and recommendations for future implementations.

10:40

2aED8. A noise footprint calculator as a tool for education and practice. Simone Torresin (Dept. of Civil Environ. Mech. Eng., Univ. of Trento, via Mesiano 77, Trento, Trentino Alto Adige 38123, Italy, simone.torresin@unitn.it), Gianluca Maracchini, Rossano Albatici (Dept. of Civil Environ. Mech. Eng., Univ. of Trento, Trento, Trentino Alto Adige, Italy), and Francesco Aletta (Inst. for Environ. Design and Eng., Univ. College London, London, United Kingdom)

Although noise pollution is a major environmental risk factor for public health, the general understanding of the phenomenon and the attention paid to it is secondary to other relevant environmental issues. The carbon footprint has proved useful to quantify and visualize the impact of human activities on the environment, to compare actions, and to monitor progress towards reducing environmental emissions. Similarly, a "noise footprint" framework could be beneficial for measuring and understanding the impact of human activities and personal/collective choices on the environment in

terms of noise pollution. However, how can the noise footprint be reduced and what compensatory measures can be taken? One answer comes from the soundscape approach, which considers the acoustic environment as perceived by humans in context trough a trans-disciplinary approach. According to this perspective, sound stimuli are not necessarily a source of pollution (i.e., noise) but can also be a resource for healthier environments. Building on this knowledge, soundscape interventions can not only reduce the noise footprint, but even reverse the impact on the environment and society. The present contribution introduces the roadmap towards the development of a noise footprint calculator as a simple and accessible tool for education and practice, creating noise-aware citizens.

11:00

2aED9. Demonstration of colored noise on brainwave activity during reading tasks: An electroencephalogram study. Stirena R. Tamariska (Architecture, National Taiwan Univ. of Sci. and Technol., No. 43, Keelung Rd., Sec. 4, Daan District, Taipei City, Taipei 10607, Taiwan, stirena.tamariska@ar.itera.ac.id), Lucky Tsaih (Architecture, National Taiwan Univ. of Sci. and Technol., Taipei, Taiwan), and Martin Clinton T. Manullang (Dept. of Electron. and Comput. Eng., National Taiwan Univ. Sci. and Technol., Taipei City, Taipei, Taiwan)

This study investigates the effects of different types of colored noise on brainwave activity during reading task, with a unique intent to use this research as a practical demonstration for the participant to learn about acoustics. Electroencephalogram (EEG) data from 29 participants were obtained and analyzed to assess brainwave activity across all channels when exposed to these colored noises. After brainwave recording process, participants were invited to view their recorded individual brainwave activities and topo mapping, so that they could learn the different colored noises might affecting their brainwave responses. The study found that most of the participants are interested to learn more about how the colored noise affects their daily activity. Besides demonstration to get participants interested in learning noise effects on brainwave activity, our brainwave study revealed that pink noise significantly influenced theta, low alpha, high alpha, low beta, and high beta frequencies during the reading tasks. In comparison, white noise had a notable impact on low and high alpha frequencies, while brown noise yielded significant changes in theta brainwave activity. This research points out that colored noise, could alter brainwave activity. It is potentially affecting cognitive engagement such as alertness and focus. By involving different major students as participants, we demonstrated the tangible effects of acoustics on cognitive functions, fostering a more profound understanding of the subject matter.

Session 2aNSa

Noise and Physical Acoustics: Aeroacoustic Sources and Fields I

Danielle Moreau, Cochair
UNSW Sydney, Sydney, 2015, Australia

Con Doolan, Cochair
School of Mechanical and Manufacturing Engineering, UNSW Sydney, Sydney 2052, Australia

Chair's Introduction—7:55

Contributed Papers

8:00

2aNSa1. Design and characterization of a low-speed aeroacoustic wind-tunnel research facility at IIT Kanpur. Akhilesh Mimani (Mech. Eng., Indian Inst. of Technol. Kanpur, MED, IIT Kanpur, Kanpur, Uttar Pradesh 208016, India, amimani@iitk.ac.in), Shubham Kumar, and Randhir Kumar (Mech. Eng., Indian Inst. of Technol. Kanpur, Kanpur, India)

This work presents the design and characterization of a state-of-the-art low-speed aeroacoustic wind-tunnel at Indian Institute of Technology Kanpur (IITK), which is a newly commissioned research facility for the experimental study of flow-induced noise generated from bodies placed in a flow field. This open-circuit, open-jet suction facility which is unique to IITK, is also the first aeroacoustic wind-tunnel in India. The facility is powered by large centrifugal fan of 75 KW rating which is controlled through a variable frequency drive. To achieve a low background noise, carefully designed parallel baffle mufflers (PBMs) are placed at the fan inlet and outlet. The inlet silencer is connected to lined diffuser and a bell-mouth collector which opens in a fully anechoic chamber made from metallic wedges, and its internal dimensions are 3.5 m $\times$ 3.5 m $\times$ 3 m. The chamber can provide a reflection-free environment beyond 250 Hz. A smooth contraction-nozzle of ratio 10:1 whose profile follows a fourth-order spline, and a settling-chamber with honeycombs and flow-screens is placed on the other side of the anechoic chamber. The test-section of size 600 mm $\times$ 600 mm is attached to the exit plane of the nozzle, and can provide flow speeds up to 45 m/s, where the maximum turbulence intensity was less than 1%.

8:20

2aNSa2. Design and characterisation of a 3-D microphone array for windtunnel measurements. Angus Wills (School of Mech. and Manufacturing Eng., UNSW, UNSW, Kensington, New South Wales 2052, Australia, a.wills@unsw.edu.au), Manuj Awasthi, Charitha de Silva, Danielle Moreau, and Con Doolan (School of Mech. and Manufacturing Eng., UNSW, Sydney, New South Wales, Australia)

A three-dimensional (3-D) array of GRAS 40PH-10 CCP Free-field Array Microphones has been designed for use in the UNSW anechoic wind tunnel (UAT) for aeroacoustic measurements. The array consists of 192 microphones split into three identical planar arrays of 64 microphones which are placed on both sides and above the potential core of the open-jet that issues into the wind tunnel test section. The planar arrays feature an Underbrink spiral comprised of eight arms of eight microphones each. The spiral configuration was selected to maximize signal-to-noise ratio (SNR) and minimize the beamwidth given the geometrical restrictions of the UAT. A white noise speaker was placed at a range of locations within the testing domain to characterize the source localization abilities of the array as a function of position and frequency. Furthermore, a pair of independent white noise speakers were placed at various separation distances to determine the array's capability to identify multiple sources within the domain. A 3-D implementation of conventional cross-spectral beamforming and Clean-SC deconvolution algorithms was used, and the performance of each was compared. It was determined that Clean-SC was the preferred method for 3-D beamforming in terms of source localization accuracy, SNR, and beamwidth, despite the higher computational cost.

8:40

2aNSa3. When less is more: Reducing environmental noise from open cycle gas turbines with acoustically transparent stacks. Benjamin Cazzolato (School of Elec. and Mech. Eng., The Univ. of Adelaide, 70 Third Ave. Forestville, South Australia 5035, Australia, benjamin.cazzolato@adelaide.edu.au), Orddom Leav, and Carl Howard (School of Elec. and Mech. Eng., The Univ. of Adelaide, Adelaide, South Australia, Australia)

Sound radiation from large open cycle gas turbine power stations is strongly refracted by high temperature and velocity gradients in the near field of the exhaust stack outlet. Recent research by the authors has shown that in the presence of a cross-flow, the exhaust plume bends downwind and in doing so leads to significant increases in far-field sound pressure levels of up to 10dB compared to levels predicted from spherical spreading. This paper is the second in a series that explores a unique "acoustically transparent silencer," which limits sound refraction consequently reducing downwind SPLs by separating the sound from the bulk exhaust flow. The efficacy of the approach is demonstrated two ways; initial characterization studies conducted in the reverberation chambers at the newly renovated Acoustic and Vibration Laboratory, University of Adelaide, and subsequent trials undertaken on a laboratory scale of 250 kW gas turbine in the field. Numerous designs are explored to quantify the effective insertion loss, where it will be shown as the transmission loss of the stack walls is reduced, the effective insertion loss increases, thus demonstrating that less is more. Reductions in ground-plane SPLs of up to 10 dB are achieved with less material and weight compared to traditional hard-walled stacks.

9:00

2aNSa4. AeroFeathers: Feathered airfoils inspired by the quiet flight of owls. William Johnston (Mech. Eng., Mech., Michigan Technol. Univ., 1400 Townsend Dr., Houghton, MI 49931, wjohnsto@mtu.edu), Janith Godakawela (Mech. Eng., Mech., Michigan Technol. Univ., Houghton, MI), L P. Nobles, Amulya Lomte (Aerosp. Eng., Wichita State Univ., Wichita, KS), Maria Carrillo-Munoz, and Bhisham Sharma (Mech. Eng., Mech., Michigan Technolog. Univ., Houghton, MI)

Owls are well-known as the quietest flying birds. Recent studies show that they achieve noise-less flight because of their specialized feather design. Their feathers have three important components: a velvety surface

texture, comb-like hooks on their leading edge, and jagged fibrous fringes on their trailing edges. Some researchers have attempted to imitate these design features within airfoil designs to reduce aeroacoustic noise emitted by fixed wing and vertical lift vehicles; however, such features cannot be easily fabricated using traditional manufacturing processes. In this student-led project funded by NASA's University Student Research Challenge, we leverage our established success at additively manufacturing fibrous structures to introduce a new method to 3-D print velvety textures, fibrous fringes, and flexible edge serration using low-cost material extrusion-based printing methods. By altering the printer's G-code, we create customized feathered airfoils with individual design parameters, including serration thickness and fringe density. We conduct acoustic tests in an anechoic chamber and aeroacoustic tests in an acoustic wind tunnel to determine the effect of each parameter on the overall acoustic signature of the airfoil. We analyze the data and explain the role of owl feather-like features on the noise reduction performance of propeller blades and scaled fixed-wing airfoils.

9:20

2aNSa5. Simultaneous measurement of unsteady surface pressure and leading edge noise of a NACA0012 airfoil. Roman Kisler (School of Mech. and Manuf. Eng., UNSW Sydney, Sydney, New South Wales 2052, Australia, r.kisler@unsw.edu.au), Chaoyang Jiang, Charitha de Silva, Con Doolan (UNSW, Sydney, New South Wales, Australia), and Danielle Moreau (UNSW, Sydney, New South Wales, Australia)

Airfoil-turbulence interaction noise, which is created whenever turbulent flow encounters an airfoil, is a major source of unwanted noise emitted by aircraft and turbomachinery. Despite being a major research focus over the past decades, the accurate prediction and understanding of airfoil–turbulence interaction noise remain an open question. This experimental investigation involved simultaneous measurements of unsteady pressure fluctuations at 61 locations across the surface of a NACA0012 airfoil and far field sound pressure at a phased 64-microphone array. Various square-mesh turbulence grids were placed upstream of the airfoil to vary the isotropic free stream turbulence. Additionally, the influence of the airfoil's angle of attack as well as different mean flow speeds on the unsteady surface pressure and radiated far field noise was investigated. Ultimately, the links between the simultaneously acquired unsteady surface pressure on the airfoil and the microphone array sound pressure were studied. This provided insight into the mechanisms of leading edge noise source generation and their influence on the radiated far field noise.

9:40–10:00 Break

10:00

2aNSa6. An aeroengine fault diagnosis method using high-frequency acoustic coupling with rotating blade flow field. Sicong Liang (Peking Univ., Beijing, China) and Xun Huang (College of Eng., Peking Univ., Beijing 100871, China, huangxun@pku.edu.cn)

Utilizing high-frequency sound waves for aeroengine structure diagnosis has been established in previous works. However, exploring its potential in analyzing rotating blade flow fields remains an open area of research. In this study, we propose a non-invasive fault diagnosis method that couples high-frequency sound waves with rotating blade flow fields, performing mode detection of the coupled frequency. As a widely used noise detection technique, acoustic mode detection enables the extraction of spatial information from noise, providing essential insights for noise reduction design. Essentially, the flow-sound coupling arises from nonlinear interactions, wherein the characteristic low-frequency periodic flow is excited by the high-frequency external sound source, generating new coupled acoustic sources that radiate and propagate inside the duct. Mode detection aids in understanding the location and spatial phase information of the coupled acoustic sources. Furthermore, through both simulations and experiments, we have established the connection between fault information and mode spectrum. Finally, we endeavor to employ a machine learning approach to build a fault diagnosis model. In summary, this method ingeniously superimposes low-frequency periodic flow field information onto high-frequency sound waves, achieving high-precision diagnosis of the rotating blade flow field.

10:20

2aNSa7. Flow noise radiated from a cylinder above a plate. Jiawei Tan (School of Mech. Manuf. Eng., UNSW Sydney, New South Wales, Australia, jiawei.tan@unsw.edu.au), Danielle Moreau, Charitha de Silva, Jeoffrey Fischer, and Con Doolan (School of Mech. Manuf. Eng., UNSW Sydney, Sydney, New South Wales, Australia)

The flow noise of an infinite span circular cylinder has been well documented over the past decades. In reality, cylinder flows are rarely isolated from any nearby boundaries. The addition of a large flat plate to one side of the cylinder created an asymmetric configuration, termed as the cylinder above a plate (CAP). Examples include roof racks and rail pantographs. Flow diagnostics of a CAP have been lacking when compared to the baseline unbounded cylinders, while the corresponding noise measurements have been practically absent. Recently, the rigid CAP configuration was systematically tested in subsonic, air flow. Cylinders with various cross-sectional shapes were tested. Microphone measurements were used to characterize the noise generated by a CAP, while particle image velocimetry (PIV) was used to characterize the cylinder wake. When the gap between the cylinder and plate was reduced, the amplitude of the tonal noise radiated from the cylinder flow generally reduced, while acoustic directivity pattern was heavily modified. The correlated flow structures were weakened and pushed away from the plate, which strongly suggested a modulation of the fluctuating forces exerted on the cylinder. By Curle's acoustic analogy, this was related to the modified acoustic noise and directivity.

10:40

2aNSa8. Aeroacoustics of finite wall-mounted square cylinders in pressure gradient flows. Chaoyang Jiang (School of Mech. Manuf.Eng., UNSW Sydney, Sydney, New South Wales, Australia, chaoyang.jiang@unsw.edu.au), Con Doolan, Charitha de Silva, and Danielle Moreau (School of Mech. Manuf. Eng., UNSW Sydney, Sydney, New South Wales, Australia)

Flow-induced noise produced by finite wall-mounted cylinders (FWMCs) is a major noise source for aircraft landing gear, rail pantographs and submarine appendages. These applications often encounter flows with various pressure gradients, however, there is little information in the literature on the effects of pressure gradients. Our recent work has demonstrated that the presence of a pressure gradient can significantly affect the near-wake flow structures of a square FWMC with a low aspect ratio of 2.4, thereby suppressing/enhancing the vortex-shedding tones. The current study extends this work to square FWMCs with varying aspect ratios, focusing on the role that aspect ratio plays in the noise generation of square FWMCs in pressure gradient flows. Experiments were undertaken using the open-jet pressure-gradient test rig in the UNSW anechoic wind tunnel, where the square FWMC model was immersed in flows with favourable-, near-zero-, and adverse-pressure gradients at a width-based Reynolds number of 28 800. The square FWMC model was installed on a traversing system to realize the change in cylinder aspect ratio. Inside the test model, a series of channels and surface pressure taps were created to measure the unsteady surface pressure using a remote microphone technique. Far-field noise and unsteady surface pressure signals were simultaneously acquired to characterize the combined effects of aspect ratio and pressure gradient on the far-field noise production and reveal the noise generation mechanisms.

11:00

2aNSa9. Theoretical prediction of unsteady wall pressure during turbulence-airfoil-interaction. Con Doolan (School of Mech. Manuf. Eng., UNSW Sydney, UNSW Sydney, Sydney, New South Wales 2052, Australia, c.doolan@unsw.edu.au), Roman Kisler (School of Mech. Manuf. Eng., UNSW Sydney, Sydney, New South Wales, Australia), Chaoyang Jiang, Charitha de Silva, and Danielle Moreau (School of Mech. Manuf. Eng., UNSW Sydney, Sydney, New South Wales, Australia)

The turbulence-airfoil-interaction generates unsteady surface pressure, which is an unwanted source of noise in many aerospace and marine applications. Consequently, an accurate methodology is desired to calculate the wall-pressure-spectrum along the chord to create new technology such as quiet wind turbine and drone propeller blades. Theoretical approaches are ideal as they not only provide an efficient means of predicting wall pressure

spectra, but also give improved insights into the physical mechanisms controlling noise generation. This paper will explore the accuracy of current theoretical approaches to unsteady wall pressure prediction. It will do this by comparing experimental measurements of wall pressure spectra in a variety of turbulent flow fields with theory. The results show the limitations of common theoretical approaches and the importance of including the effects of turbulence distortion in the theoretical model.

11:20

2aNSa10. Broadband noise characteristics of fans with different blades under identical working conditions. Tianyu Xu (Dept. of Mech. Eng., The Univ. of Hong Kong, Pokfulam Hong Kong, Hong Kong SAR, China, tianyu30@connect.hku.hk), Bohua Huang, Ying Hu, and Lixi Huang (Dept. of Mech. Eng., The Univ. Hong Kong, Hong Kong, China)

Small axis-flow fans are widely used for air cooling and ventilation systems. A common small axis-flow fan has an unshrouded rotor, structs or stator vanes and a round casing. The fans used in this work are designed based on isolated-airfoil blade design mode and simple radial equilibrium theory supplemented by three-dimensional flow simulations with steady flow. The noise radiated from the fan can be decomposed to discrete tones and broadband noise, which is mainly due to interaction between blades and structs or stators vanes and turbulence, respectively. Under the working conditions of the same pressure rise and volume flow rate, the noise spectra from different fans are analyzed. It is found that with the addition of the fan blade number, the concentration of the broadband sound energy gradually shifts from low frequency to high frequency. And the fewer the number of blades, the less the corresponding total broadband energy. Besides, the calculation results also show the fan with more blades can be absorbed more sound by the same sound absorption materials with the same volume. The findings are useful for improving the acoustic design of small axis-flow fans.

Session 2aNSb

Noise and Physical Acoustics: Sonic Boom I

Alexandra Loubeau, Cochair
NASA Langley Research Ctr., 2 N Dryden St., Hampton, VA 23681

Victor W. Sparrow, Cochair
Grad. Prog. in Acoustics, Penn State Univ., 201 Applied Science Bldg., University Park, PA 16868

Chair's Introduction—10:35

Contributed Paper

10:40

2aNSb1. A vision of the next era of supersonic flight. Sandy R. Liu (Office of Environment & Energy, Noise Div., U.S. Federal Aviation Administration (FAA), 800 Independence Ave., SW, Washington, DC 20591, sandy.liu@faa.gov)

Concorde was decommissioned in 2003, ending civil supersonic aircraft operations in the world. However, today there is continued interest by proponents/developers of supersonic aircraft that believe by utilizing advanced technologies, new supersonic aircraft can be economically viable, safe, reliable, and significantly quieter than the Concorde. In 2004, the International Civil Aviation Organization (ICAO)'s formed a Supersonic (noise) Task Group to monitor whether supersonic research and developments could revive lost high-speed capabilities and support commercially viable, environmentally responsible global aviation operations. Since then, the United States has been focused on progressing work related to the need for a landing and takeoff noise standard for supersonic aircraft, and an en-route sonic boom standard. This presentation briefly highlights achievements to date that intend on opening the new era of supersonic flight ahead.

Invited Papers

11:00

2aNSb2. New reference day atmosphere for analyses of quiet supersonic overflight. Alexandra Loubeau (NASA Langley Res. Ctr.1 NASA Dr., MS 463, Hampton, VA23681, a.loubeau@nasa.gov), Victor W. Sparrow (Grad. Prog. in Acoust., Penn State Univ., University Park, PA), William Doebler, Sriram K. Rallabhandi (NASA Ames Res. Ctr., Moffett Field, CA), Stephane Lemaire, Pierre-Elie Normand (Dassault Aviation, Saint-Cloud, France), and Sandy R. Liu (Office of Environment & Energy, Federal Aviation Administration, Washington, DC)

New reference day atmospheric conditions have been defined for future supersonic aircraft en route noise certification procedures. As field measurements are acquired, in a variety of atmospheric conditions, it is important to relate those measurements back to a common reference atmosphere to provide uniformity in method across applicants for noise certification. Because of the long propagation distances involved between the aircraft flight altitude and the ground, en route noise certification measurements can be substantially affected by the atmosphere. Properties, such as pressure, density, temperature, humidity, and winds, vary with altitude. Therefore, the definition of reference atmospheric profiles as a function of altitude are needed instead of the coarse layered atmosphere and homogeneous humidity used in subsonic aircraft landing and takeoff noise certification. Ground loudness levels from overflight of a quiet supersonic demonstrator concept, calculated with atmospheric data from sites across the world, were compared with the results from several candidate reference atmospheres to identify a reference that would minimize the required adjustment to certification test measurements. The existing ICAO 7488/3 profiles for temperature and pressure, combined with a modified ISO 5878 humidity profile, were found to offer the best results.

11:20

2aNSb3. Simulations of X-59 sonic thumps and traditional sonic booms propagated around the world for three atmospheric models. William Doebler (NASA Langley, 1 NASA Dr., MS 463, Hampton, VA 23681, william.j.doebler@nasa.gov) and Victor W. Sparrow (Penn State, University Park, PA)

Propagation simulations of sonic booms from supersonic aircraft through atmospheric data over time at fixed locations provide the opportunity to assess noise exposure statistics for different climate regions. Knowledge of climate-based differences in sonic boom noise exposure statistics is important to ensure that future civil supersonic aircraft noise certification standards are globally applicable and effective. In this presentation, simulated sonic booms from the NASA X-59 Quesst quiet supersonic aircraft and conventional supersonic aircraft were propagated through atmospheric data at 100 locations across the world using PCBoom. Noise exposure statistics are compared for propagation results from three different atmospheric databases (NOAA Global Forecast System, NOAA Climate Forecast System Version 2, and the ECMWF Reanalysis Version 5). These atmospheric models were chosen due to their global coverage, popularity, and database availability. Preliminary statistical models are fit to assess the impact of several factors including flight direction, season, ground elevation, and climate on noise exposure size and loudness. Areas with prevalence of higher noise due to their climate are identified, which could help inform future supersonic aircraft noise standards.

11:40

2aNSb4. Robust sonic boom reduction in primary boom carpet. Atsushi Ueno (Japan Aerosp. Exploration Agency, 6-13-1 Osawa, Mitaka, Tokyo 181-0015, Japan, aueno@chofu.jaxa.jp), Hiroaki Ishikawa, Shinya Koganezawa, and Yoshikazu Makino (Japan Aerosp. Exploration Agency, Mitaka, Japan)

Japan Aerospace Exploration Agency has conducted a low-boom design for a small-sized (about 50 PAX) supersonic airliner. In this study, sonic boom loudness is evaluated in a primary boom carpet, while the focus boom is not considered. Two geometries having different robustness with respect to loudness distribution in a primary boom carpet are designed. One is designed considering only a cruise Mach number, and sonic boom loudness is reduced at both under-track and off-track positions. The other is designed considering both cruise and off-design Mach numbers, and the maximum sonic boom loudness in a primary boom carpet is minimized. In the presentation, design examples to achieve low-boom characteristics at both under-track and off-track positions are reported. In addition, a low-boom design concept using canard is proposed to minimize the maximum sonic boom loudness in a primary boom carpet. Finally, differences of loudness distribution in a primary boom carpet between two geometries are discussed.

Session 2aPA

Physical Acoustics and Biomedical Acoustics: Multiphase Flow and Acoustics I

John S. Allen, Cochair
Mechanical Engineering, Univ. of Hawaii, 2540 Dole St., Holmes Hall 302, Manoa, Honolulu, HI 96822

Richard Manasseh, Cochair
Mechanical and Product Design Engineering, Swinburne Univ. of Technol., John St., Hawthorn, Melbourne 3122, Australia

Joseph (Yeo Cheon) Kim, Cochair
Mechanical Engineering, UNSW, Sydney 2052, Australia

Chair's Introduction—8:55

Invited Papers

9:00

2aPA1. On the utilisation of the coupled-oscillator approach as a surrogate model for sound propagation in the vicinity of bubble groups. Andrew Ooi (Dept. of Mech. Eng., The Univ. of Melbourne, Melbourne, Victoria 3010, Australia, a.ooi@unimelb.edu.au), Shuang J. Zhu (Mech. and Product Design Eng., Swinburne Univ. of Technol., Hawthorn, Victoria, Australia), Alex Skvortsov (Maritime Div., Defence Sci. Technol., Melbourne, Victoria, Australia), and Richard Manasseh (Mech. and Product Design Eng., Swinburne Univ. of Technol., Melbourne, Victoria, Australia)

The presence of bubble curtains have been shown to reduce the level of underwater noise propagation. This is due to the change in density and acoustic-impedance between the group of bubbles and the water. In addition, sound waves can be absorbed and scattered when they interact with bubble curtains. In this paper, it is shown that a mathematical model based on the discrete bubble model (DBM) and utilising the self-consistent coupled-oscillator theory is able to correctly predict the bubble collective resonance frequencies of 1D line, 2-D planar, and 3-D complex bubble cloud configurations. This framework can also accommodate polydisperse-sized bubbles. We use the DBM and conduct modal analysis on common 2-D and 3-D bubble curtain configurations and show that the fundamental modes in both configurations dominated the vibration of bubble curtains when subjected to an external acoustic source. In addition, it is demonstrated that the performance of the bubble curtain is most affected by the changes in the bubble size whereas the variations in inter-bubble spacing had no significant influence.

9:20

2aPA2. The acoustic emissions of bubbles released from melting glacier ice. Grant B. Deane (Marine Physical Lab., Scripps Inst. of Oceanogr., UC San Diego, UC San Diego, 9500 Gilman Dr. #0206, La Jolla, CA 92093-0206, gdeane@ucsd.edu)

The transfer of heat and salt across the near-ice (proximal) boundary layer of tidewater glacier termini are important controls on submarine ablation. Glacier ice typically contains pressurized air bubbles, formed at the base of the firn layer and transported over centuries or millennia to the ocean. These bubbles are released during ice melting into the proximal boundary layer. The combination of ice, water, and air makes the proximal boundary layer a complex, multiphase flow. Bubble release is accompanied by pulses of sound with peak pressures spanning a wide range of values that can exceed 100 Pa. Investigations into the origins of this signal and its application to monitoring glacier submarine ablation rates will be presented. [Work supported by the US Office of Naval Research Ocean Acoustics program (Grant No. N00014-21-1-2316).]

Contributed Paper

9:40

2aPA3. Analysis of sound emissions and physical properties of bubbles generated by breaking waves. Filippo Nelli (Swinburne Univ. of Technol., John St., Hawthorn, Victoria 3122, Australia, fnelli@swin.edu.au), Shuang J. Zhu, Chintaka Jacob (Swinburne Univ. of Technol., Hawthorn, Victoria, Australia), Andrew Ooi (The Univ. of Melbourne, Melbourne, Victoria, Australia), and Richard Manasseh (Swinburne Univ. of Technol., Melbourne, Victoria, Australia)

Although the relationship between bubble diameter and the corresponding resonant frequency has been known since 1930s, the link between the amplitude of the sound and the physical properties of the bubble is still unclear. Sound amplitude is a parameter inevitably measured when passive acoustic measuring techniques are employed, and its magnitude as measured is related to not only the size of the bubble but also its position in space relative to the measurement point. Here, we present a study where individual bubbles were first generated through a vertical plunging jet and analyzed to establish a link between their diameters and sound emissions. Subsequently, these data were used to predict bubble properties occurring in breaking waves generated in laboratory experiments. A high frequency camera was used to capture images of the bubbles and visually check the accuracy of the diameter and location predictions. Results show good agreement between plunging jet and breaking waves bubbles within the $+-$ 1.5 dB confidence interval; addressing with residual experimental uncertainties should offer further improvements.

Invited Papers

10:00

2aPA4. Utility of ocean wave parameters for improving predictions of ambient noise. William E. Rogers (Ocean Sci. Div., Naval Res. Lab., NRL Code 7322, Stennis Space Ctr., MS 39529, erick.rogers@nrlssc.navy.mil), Laurie T. Fialkowski, and Daniel J. Brooker (Acoust. Div., Naval Res. Lab., Washington, DC)

This study is concerned with prediction of the "wind noise" component of ambient noise (AN) in the ocean. It builds on the seminal paper by Felizardo and Melville (1995), in which the authors quantified the correlation between AN and individual wind/wave parameters. We utilize hydrophone observations deployed in the north and northeast Pacific Ocean. Wind/wave parameters are obtained from moored buoys and numerical models. We describe a procedure developed for this study which isolates the correlation of AN with wave parameters, independent of mutual correlation with wind speed (residual correlation). We then describe paired calibration/prediction experiments, whereby multiple wind/wave parameters are used simultaneously to estimate AN. We find that the improvement from inclusion of wave parameters is robust but modest. We interpret the latter outcome as suggesting that wave breaking responds to changes in local winds quickly, relative to, for example, total wave energy, which develops more slowly. This outcome is consistent with prior knowledge of the physics of wave breaking, e.g., Babanin (2011). We discuss this in context of the wave spectrum's time/space response to wind forcing.

10:20–10:40 Break

10:40

2aPA5. Disturbing bubbles in drop-on-demand piezo-acoustic inkjet printing. Michel Versluis (Phys. of Fluids group, Univ. of Twente, P.O. Box 217, Enschede 7500 AE, Netherlands, m.versluis@utwente.nl), Hans Reinten (Canon Production Printing Netherlands, Venlo, Netherlands), Arjan Fraters, Maaike Rump, Youssef Saade, Christian Diddens, Detlef Lohse (Phys. of Fluids Group, Univ. of Twente, Enschede, Netherlands), and Tim Segers (BIOS Lab-on-a-Chip Group, MESA+ Inst. for Nanotechnology, Enschede, Netherlands)

Drop-on-demand piezo-acoustic inkjet printing is the method of choice for high-frequency and high-precision droplet deposition. Microbubbles entrained in a piezo-driven silicon MEMS printhead disturb the acoustics of the microfluidic ink channel and, thereby, the jetting behavior. The strong deformations of the air-liquid interface at the nozzle exit may lead to an inward directed air jet with bubble pinch-off and the subsequent entrainment of an air bubble. Here, we use ultrafast x-ray phase-contrast imaging and direct numerical simulations to study the mechanisms underlying the bubble entrainment, where we demonstrate good agreement between experiments and numerics. The entrained bubbles were also visualized in the ink channel of the silicon printhead in a highly sensitive shortwave infrared (SWIR) imaging setup exploiting the optical window at 1550 nm. The infrared recordings show the rich phenomena of acoustically driven bubble dynamics inside the printhead. We also study the resonance behavior of the ink channel as a function of the microbubble size and number of bubbles through theoretical modeling and experiments. The system is modeled as a set of two coupled harmonic oscillators, one for the compliant ink channel and one for the microbubble and where we find excellent agreement between the predicted and measured eigenfrequencies.

11:00

2aPA6. Passive acoustic measurement of bubble size and number from a bubble chain. Mohammed M. Roshid (Chittagong Univ. of Eng. and Technol., Chittagong, Bangladesh) and Richard Manasseh (Mech. and Product Design Eng., Swinburne Univ. of Technol., John St., Hawthorn, Melbourne, Victoria 3122, Australia, rmanasseh@swin.edu.au)

Experiments and matching simulations are presented on the passive-acoustic emissions of a chain of bubbles, with the aim of deducing both the bubble size and number of bubbles from the emitted sound. It is well known that bubble-acoustic interactions cause shifts in frequency from those predicted for single bubbles, complicating the goal of measuring bubble size by measuring frequency, and also affecting the collective resonances of bubbles driven by ultrasound. A horizontal line of identical bubbles was held fixed in space, eliminating uncertainties in earlier studies due to the variable shape and location of freely rising bubbles. At one end of the line, a first, identical bubble naturally emitted a pulse of sound on formation from an underwater nozzle, exciting the rest of the bubbles into a coupled acoustic emission. It was found that a theoretically predicted relation between frequency and the number of bubbles, which can be derived from the coupled equations of motion, permitted an accurate fit to the data. Measurements of two independent spectral peaks unambiguously determined the two unknowns of bubble size and bubble number. However, experiments diverge from theory and simulations for greater than six bubbles, which are speculated may be explicable by network theory.

11:20

2aPA7. Acoustical characterization of concentrated bubbly media. Camille Cohen (Saint-Gobain Res. Paris, 39 quai Lucien-Lefranc, Aubervilliers Cedex 93303, France, Camille.Cohen@saint-gobain.com), Tony Valier-Brasier (Sorbonne Univ., Paris, France), Yann Desailly (Saint-Gobain Res. Paris, Aubervilliers, France), and Valentin Leroy (CNRS/Paris-Cité Univ., Paris, France)

Bubbles are known to be strongly resonant acoustic scatterers. As such, their presence in a fluid, even for amounts of gas as low as 1%, highly impacts the propagation of acoustic waves. Ultrasounds are, thus, a great tool for bubble distribution measurements in opaque fluids. Additionally, diluted media models have been shown to have a good agreement with experiments at volume fraction below 1%. One of the aims of our study is to determine the limit volume fraction beyond which the models start to fail. To achieve that, we measure relfection and transmission of ultrasounds through bubbly media ranging from 5% to 60 % of gas volume fraction. Samples are polydisperse thin slabs with median radii ranging between 30 and 180 μm. Alongside experimental measurements, numerical simulations are carried out with monodisperse distributions. Up to 10%, we find a good agreement between the independent scattering approximation and our experimental results.

11:40

2aPA8. Acoustic attenuation in mixtures of air and water droplets. Orddom Leav (Defence Sci. Technol. Group, DSTG, Fishermans Bend, Victoria 3207, Australia, orddom.leav@defence.gov.au) and Stephen Moore (Defence Sci. Technol. Group, Fishermans Bend, Victoria, Australia)

Sound propagation through mixtures of air and water-droplets occurs in many natural systems such as in fogs and clouds, where it has been observed that acoustic waves are attenuated to a greater degree than in the absence of water droplets. This effect has been exploited at space shuttle launches with the sound suppression system, where water sprayed around the shuttle significantly reduces the acoustic power radiated from the rocket engines. Theoretical studies have investigated various mechanisms causing acoustic attenuation in air-water droplet mixtures: heat transfer, mass transfer and momentum transfer. More recent studies have incorporated other mechanisms into analytical models so that they are more representative of sound propagation through fogs or clouds. The aim of this paper is to review the different mechanisms causing attenuation of acoustic waves, and to present a comparison of the different analytical models to illustrate the differences in predicted acoustic attenuation and dispersion. Some limitations of the different models are identified, and proposals for experimental work to validate analytical results are discussed.

Session 2aPPa

Psychological and Physiological Acoustics: Understanding Hearing in a Dynamic World (Poster Session)

Alan Kan, Cochair
School of Engineering, Macquarie Univ., 50 Waterloo Road, Macquarie Park, 2113, Australia

Valeriy Shafiro, Cochair
Communication Disorders & Sciences, Medical Ctr., Rush Univ., 600 S. Paulina St., AAC 1015, Chicago, IL 60612

Raymond Goldsworthy, Cochair
Biomedical Engineering, Univ. of Southern California, Los Angeles, CA

All posters will be on display and all authors will be at their posters from 8:00 a.m. to 9:20 a.m.

Contributed Papers

2aPPa1. Is the crying of my toddler detrimental to my hearing? An acoustic assessment of a toddler's cry and the estimated impacts on primary caregivers. Victoria H. Rastelli (Acoust., Norman Disney & Young Ltd, Level 1, AON Ctr., 29 Customs St. West, Auckland 1010, New Zealand, v.rastelli@ndy.com)

Toddlers' caregivers receive a fair amount of noise regularly, they are typically exposed to a first year of high intensity, high pitch crying noise from newborns and sometimes as a full-time activity during parental leave. It was estimated that the partial noise exposure during 8 hours of this stage for caregivers was between 78.5 and 81.5 dB, which is not negligible. After this initial year, toddlers' crying intensity, patterns, and proximity of the caregivers' changes; however. the surveyed population still considers that their hearing experiences a detrimental effect. Moreover, research is pointing towards the exposure to medium intensity noise, previously assumed "safe," that may cause changes in the structure and function of the auditory system, possibly contributing to increasing the individual's hearing threshold or aggravating the range and extent of age-related hearing loss, among other impacts. This study continues a survey addressed to caregivers, interviews medical experts and estimates noise exposure scenarios for toddlers' caregivers, to advise safe noise mitigation practices, raise awareness and promoting a healthier, nurturing environment for all, child, parents and caregivers.

2aPPa2. Variability of speech spectra in offices by room types and communication methods. Rewan Toubar (Dept. of Bldg., Civil and Environ. Eng., Concordia Univ., 1515 Saint-Catherine St. W, EV-0S3.412, Montreal, QC H3G 1S6, Canada, rewantoubar@yahoo.com), Roderick Mackenzie (Soft dB, Montreal, QC, Canada), and Joonhee Lee (Dept. of Bldg., Civil and Environ. Eng., Concordia Univ., Montreal, QC, Canada)

The acoustic characteristics of room environments can significantly influence speech levels and spectra, an aspect often overlooked in the current method for predicting speech privacy. This paper presents an investigation into the variability of speech spectra in various office contexts, with a specific focus on the influence of room type, communication medium, and language. The study involved over 70 workers in different office room types in Quebec, Canada, who participated in measuring speech spectra within those spaces. Two communication methods, in-person and video-conferencing scenarios, were utilized with participants using either English or French languages. The real-world scenarios shed light on the significant impact of various parameters on speech characteristics within office settings. Comparing the results with standardized speech levels and spectra from ASTM and ISO standards, derived from controlled environments like anechoic chambers, the study reveals significant differences when applying the standard speech spectra to actual office environments. Based on these findings, the study proposes potential modifications to the existing standards, aiming to enhance the accuracy of speech privacy and intelligibility predictions.

2aPPa3. Head movement during natural group conversation and inter-annotator agreement on manual annotation. Angkana Lertpoompunya (Commun. Sci. and Disord., Mahidol Univ., 71/10 Petkasaem Rd., Soi 81/1, Nongkangpu, Bangkok, Nongkaem 10160, Thailand, angkana.leo@mahidol.edu), Nathan C. Higgins, Erol J. Ozmeral, and David A. Eddins (Commun. Sci. and Disord., Univ. of Central Florida, Orlando, FL)

During speech communication and conversational turn-taking, listeners direct their head and eyes to receive meaningful auditory and visual cues. Features of these behaviors may convey listener intent. This study designed a test environment, data collection protocol and procedures, and investigated head movement behaviors during self-driven conversations among multiple partners. Nine participants were tested in cohorts of three. Participants wore a headset with sensors tracked by an infrared camera system. Participants watched an audio-video clip, followed by a 5-min undirected discussion. The entire session was video recorded for annotation purposes. Two annotators independently coded the video files using the EUDICO Linguistic Annotator software application. Annotations were then co-registered with the head tracking data in post processing. Inter-annotator agreement demonstrated the desired reliability, thereby validating the procedures designed. Movement trajectories showed that there were individual differences on the head yaw distribution. The combination of objective measures of head movement and manual annotation of conversation behaviors provides a rich data set for characterizing natural conversations in ecologically valid settings. The measurement procedures and coding system developed here is a first step towards characterizing head movements during conversations needed to predict listening intent and to create actions based on those predictions.

2aPPa4. Classification of head movements during multi-person conversation. Nathan C. Higgins (Commun. Sci. and Disord., Univ. of South Florida, Tampa, FL, higgins1@usf.edu), Erol J. Ozmeral (Commun. Sci. and Disord., Univ. of South Florida, Tampa, FL), Angkana Lertpoompunya (Commun. Sci. & Disord., Mahidol Univ., Bangkok, Nongkaem, Thailand), and David A. Eddins (Univ. of Central Florida, Orlando, FL)

Hearing-aid processing strategies could be greatly improved by real-time knowledge of the users' listening intentions, information that may be encoded by characteristic head movements during conversational turn-taking with multiple talkers, and measured by accelerometers in the hearing devices. First, however, it must be determined to what degree head movements are stereotypic across listeners, or alternatively, unique to each individual, and whether an automatic classifier can identify specific movements. Three cohorts of three young, normal-hearing individuals participated in a semi-structured conversation for 50 minutes. Participants wore hearing aids with accelerometers and were video-recorded. Video recordings were then used to track head movements using zFace (Jeni *et al*. 2017) and manually annotate head movement and communication activities. Accelerometer data was segmented into 1-s windows and labeled by the annotated activity identified in that window. Features based on temporal and frequency characteristics were used to train multi-class models with across-subject or within-subject design. Significantly better than chance classification was observed for all activities using both model designs, with higher classification accuracy for the models trained on the individuals' own-data. In summary, head movement behaviors can be classified with short-scale time resolution, and individual differences are a key factor for accurate classification.

2aPPa5. Lack of a clear head movement benefit in a spatial release form masking task. Erol J. Ozmeral (Commun. Sci. and Disord., Univ. of South Florida, Tampa, FL, eozmeral@usf.edu), Nathan C. Higgins, and Hoonki Chun (Commun. Sci. and Disord., Univ. of South Florida, Tampa, FL)

The study of spatial benefit in complex listening environments often requires listeners to maintain a fixed head position so that controls are in place for understanding binaural function. In natural listening environments, however, listeners may use head movements (e.g., turns and tilts) to achieve their maximum speech reception. In a previous study, we developed a spatial release from masking task that adapts the location of the maskers to determine the angular threshold for a predetermined masking release (in dB), and in the present study, we modified this task to allow for the listeners to move their heads as they wished to maximize performance on the task. Twenty young normal hearing listeners were tested on the head-fixed and head-free conditions. Head tracking was collected for posthoc analyses. Overall, there was no significant difference in behavioral thresholds between conditions despite wide ranges of head movements in the head-free condition. Interestingly, large individual differences in head movement translated to a narrowing of the variance in behavioral thresholds relative to the head-fixed condition, suggesting that listeners were indeed optimizing their speech reception. Further analyses will dissect the types of head movements and strategic differences among the participants, with implications for listeners with hearing loss.

2aPPa6. The interaction between talker head orientation, spatial separation, and extended high frequencies for speech-in-speech recognition. Rohit M. Ananthanarayana (Speech and Hearing Sci., Univ. of Illinois Urbana-Champaign, 901 S Sixth St., Champaign, IL 61820, rohitma2@illinois.edu), Vahid Delaram, Allison Trine (Speech and Hearing Sci., Univ. of Illinois Urbana-Champaign, Champaign, IL), Margaret Miller (Boys Town National Res. Hospital, Omaha, NE), Chris Stecker (Boys Town National Res. Hospital, Omaha, NE), Emily Buss (Otolaryngol./HNS, Univ. of North Carolina at Chapel Hill, Chapel Hill, NC), and Brian B. Monson (Speech and Hearing Sci., Univ. of Illinois Urbana-Champaign, Champaign, IL)

Speech-in-speech recognition experiments generally involve presentation of stimuli as if target and masker talkers are facing the listener. This is because speech signals are recorded with microphones located in front of the talkers. In real-world situations, masker talkers are often rotated away from the listener, facing their own conversational partners. This head orientation mismatch between the target and masker talkers provides cues that enhance speech recognition, including cues at extended high frequencies (EHFs; >8 kHz) due to the directional nature of EHFs in speech radiation. However, it is unclear how head orientation and EHF cues affect speech recognition when target and masker speech come from different spatial locations, as is the case in realistic multi-talker environments. In the present study, we investigated the EHF benefit in a complex auditory scene involving differences in target and masker head orientation and spatial location. Masker head orientation was either 0° (facing) or 90° (non-facing). Target speech location was 0° azimuth, while masker speech location was either 0° or ±45° azimuth. Preliminary analyses suggest spatial cues provide greater benefit than head orientation cues, and that the EHF benefit is diminished when talkers are spatially separated. [Work supported by NIH grant R01DC019745.]

2aPPa7. Do individual perceptual abilities affect speech recognition with spatial hearing aid processing? Varsha H. Rallapalli (Commun. Sci. & Disord., Northwestern Univ., 2240 Campus Dr., Evanston, IL 60208, varsha.rallapalli@northwestern.edu) and Erol J. Ozmeral (Univ. of South Florida, Tampa, FL)

Individuals with hearing loss experience persistent communication difficulties in noisy conditions, even with specialized hearing aid (HA) processing. Little is known about how HA processing interacts with auditory perceptual abilities required for speech-in-noise listening, possibly leading to mismatched parameters for certain listeners. We investigated how speech recognition with spatial HA processing—intended to reduce signals from off-axis locations—is affected by individual auditory perceptual abilities. 16 listeners with hearing loss completed a sentence recognition in noise task with wearable HAs in omnidirectional processing or binaural beamforming settings. On- (0°) or off-axis (+90°) target sentences were mixed with gender-matched two-talker maskers that were either spatially separated or co-located. The Portable Automated Rapid Testing program was used to obtain individual binaural (Spatial Release from Masking task) and monaural processing (Spectrotemporal Modulation Detection task) measures. Data show positive associations between perceptual abilities and speech recognition with omnidirectional processing across target and noise locations and negative associations with beamforming for off-axis targets in spatially separated noise. Relative associations of binaural and monaural processing abilities varied by test condition. The study has implications for individualizing spatial processing parameters in clinical HA fittings. [Work supported by NIH-K01DC018324.]

2aPPa8. Effects of vowel coloration on perceived directional offsets between single-syllable speech sounds. William Martens (National Acoust. Labs., Macquarie University, Sydney, New South Wales 2109, Australia, bill.martens@mq.edu.au) and Shuichi Sakamoto (Res. Inst. of Elec. Commun., Tohoku Univ., Sendai, Miyagi, Japan)

The ability of normal hearing listeners to spatially localize single-syllable speech sounds was examined via a two-interval, two alternative forced choice (2AFC) spatial offset discrimination task. To examine directional biases in horizontal sound localization, this task required listeners to observe the direction of an unseen second talker's voice immediately after hearing a first talker's voice in an alternative direction. Listeners reported only the relative frontward versus rearward spatial offset between the two voices heard during each trial. Headphone-based virtual acoustic simulation enabled blind listening tests that were completed without feedback to the listener, revealing the ease with which listeners judged the relative spatial offset of the second speech sound, although those judgments were strongly biased by differences in the source spectra. The single-syllable speech-sound stimuli were processed using individually measured head-related transfer functions (HRTFs) of 15 listeners for whom individually measured earphone transfer functions (ETFs) were also employed. While analysis of the 2AFC response data via receiver operating characteristic (ROC) curves showed a generally high level of directional sensitivity, systematic biases were observed that could be explained via the differences in vowel coloration between the first and second source presented during each trial.

2aPPa9. Hearing aid processing affects acoustic features important for emotional responses to sounds. Erin Picou (Dept. of Hearing and Speech Sci., Vanderbilt Univ., 1215 21st Ave S, Rm. 8310, Nashville, TN 37232, erin.picou@vanderbilt.edu) and William Martens (National Acoust. Labs., Sydney, New South Wales, Australia)

A wide range of emotional responses is desirable and supports overall quality of life. However, adults with hearing loss exhibit a reduced range of emotional responses to non-speech sounds compared to their peers with normal hearing. Recent work demonstrates that audibility of low- and high-frequency cues supports emotion perception of non-speech sounds, as does modulation energy and roughness in the sound. The purpose of this study was to quantify the effects of hearing aid processing on acoustic parameters that have been related to emotional responses to non-speech sounds. Twenty-three adults with bilateral hearing loss rated valence of non-speech sounds, with and without hearing aids. The non-speech sounds were then recorded through an acoustic manikin with and without the research hearing aids with each participant's settings. To examine the effects of hearing aid processing, the acoustic parameters previously related to emotional responses were analyzed using a combination of cluster analysis and linear regression. Results indicate that some acoustic parameters are affected by hearing aid processing (e.g., less amplitude modulation at 1270 Hz), which partially explain the reduced range of subjective emotional responses seen in adults with hearing loss. The results of this study have implications for the refinement of hearing aid processing algorithms.

2aPPa10. Auditory reality and evaluation of hearing-aid function. Florian Wolters, Petra Herrlin, Josefina Larsson (ORCA Europe, WS Audiol., Stockholm, Sweden), and Karolina Smeds (ORCA Europe, WS Audiol., Bjorns Tradgardsgrand 1, Stockholm SE-116 21, Sweden, karolina.smeds@orca-eu.info)

Knowledge about people's auditory reality (AR), i.e., the variety of listening demands and environments experienced in everyday life, is crucial when developing and evaluating hearing-aid function. We have previously investigated AR of older people with impaired hearing and generally seen an "easy" AR (many reports of passive listening and focused listening to TV, mainly in quiet). In the current study, we included participants with both normal and impaired hearing to investigate if participants with impaired hearing are *avoiding* difficult situations. The EMA study ran for a week with six daily prompts. In addition to the EMA survey items from our previous study, sound levels were recorded, and avoidance was investigated in an exit interview and by analyzing differences in AR between the groups. Preliminary analyses show that there were differences between the two groups in terms of *where* they were when they responded. The differences in *listening task* were small, but the participants with impaired hearing rated the reported situations to be more difficult than the normal-hearing participants. Our new results will be presented together with a discussion about the consequences of our auditory reality data on the way we evaluate hearing-aid function.

Session 2aPPb

Psychological and Physiological Acoustics: Auditory Cognition in Interactive Virtual Environments I

Janina Fels, Cochair
Institute for Hearing Technology and Acoustics (IHTA), RWTH Aachen Univ., Kopernikusstr. 5, Aachen 52074, Germany

Joerg M. Buchholz, Cochair
Linguistics - Audiology, Macquarie Univ., Australian Hearing Hub, Level 1, 16 University Ave., 2109, Australia

Chair's Introduction—7:55

Invited Papers

8:00

2aPPb1. How does realistic classroom noise affect auditory selective attention? Carolin Breuer (Inst. for Hearing Technol. and Acoust. (IHTA), RWTH Aachen Univ., Aachen, Germany), Larissa Leist (Cognit. and Developmental Psych., Rheinland-Pfälzische Technischen Universität Kaiserslautern-Landau, Kaiserslautern, Germany), Stephan Fremerey, Alexander Raake (Audiovisual Technol. Group, Technische Universität Ilmenau, Ilmenau, Germany), Maria Klatte (Cognit. and Developmental Psych., Rheinland-Pfälzische Technischen Universität Kaiserslautern-Landau, Kaiserslautern, Germany), and Janina Fels (Inst. for Hearing Technol. and Acoust. (IHTA), RWTH Aachen Univ., Kopernikusstr. 5, Aachen 52074, Germany, Janina.Fels@akustik.rwth-aachen.de)

In the past years, more and more effort has been made to bring the reality in the lab to investigate cognitive performance in a close-to real life manner and, thus, increase the validity of current research. A first step towards being ecologically valid is using a dynamic binaural reproduction, which offers an accurate representation and allows the participants to engage in the acoustic scene. In a study on noise effects regarding the auditory selective attention of children, Loh *et al*. 2021 found that children were impaired by noise, while adults remained unaffected. However, they used white noise at a comparably low signal-to-noise ratio. To investigate the impact of a realistic scenario, the current work investigated a fluctuating realistic classroom noise using typical noise sources such as writing and the movement of chairs. Additionally, semantically meaningful speech was added as an attention capturing noise source. To get first insights into the effects of this realistic noise, a pilot study was conducted with adult participants in a virtual classroom scene.

8:20

2aPPb2. Realistic interaural level differences help listeners suppress auditory distractors. Barbara Shinn-Cunningham (Neurosci. Inst., Carnegie Mellon Univ., 5000 Forbes Ave., Pittsburgh, PA 15213, bgsc@andrew.cmu.edu), Wusheng Liang (Elec. and Comput. Eng., Carnegie Mellon Univ., Pittsburgh, PA), and Christopher A. Brown (Commun. Sci. and Disord., Univ. of Pittsburgh, Pittsburgh, PA)

Past research hints that realistic auditory spatial simulations not only "sound better," but better engage brain mechanisms controlling spatial auditory attention. We sought to replicate this finding by comparing behavior and neural responses for simulations using: (1) individualized head-related transfer functions (HRTFs), (2) filters preserving individualized, frequency-specific interaural level differences (ILDs) but removing interaural time differences (ITDs), and (3) filters preserving ITDs but removing ILDs. Listeners were asked to report back a random stream of consonant-vowel syllables (tokens spoken by a male talker) from left or right while ignoring a different random, temporally inter-digitated consonant-vowel stream from the opposite direction. To extend previous findings, we tested a listener's ability to ignore a salient distinct sound (a cat MEOW) that occurred in the middle of some randomly selected trials. ITD-only simulations led to worse performance, especially on cat-interrupted trials. Simultaneous electroencephalography showed that in ITD-only simulations, attention evoked no significant lateralized alpha oscillations (a signature of spatially directed attention) and the to-be-ignored cat elicited larger neural responses than for other simulations. These results highlight how different auditory virtual environment simulations can influence perceptual and neural outcomes and suggest that simulations including natural ILDs enhance a listener's ability to focus spatial attention.

8:40

2aPPb3. Development and perceptual evaluation of hearing aids using interactive virtual environments. Ralph Peter Derleth (SONOVA AG, Laubisruetistrasse 28, Staefa 8340, Switzerland, peter.derleth@sonova.com), Hannes Wuethrich, Laurent Simon, and Stefan Klockgether (SONOVA AG, Staefa, Switzerland)

In hearing research, it has been shown that the realism of the test conditions has an influence on speech intelligibility thresholds, the subject's behavior, learning abilities and listening effort. This talk presents two multimodal experiment setups that aim to bridge the differences between the laboratory and reality, focusing on the development and perceptual evaluation of hearing devices. The first laboratory uses 360° video and higher order ambisonics stimuli, a head-mounted display, and a VR perceptual evaluation interface to asses the effect of hearing devices on noise intrusiveness, speech quality, and naturalness. The second laboratory is a large loudspeaker array inside of which subjects movements and behavior can be tracked and videos can be projected on walls. This talk will introduce several challenges encountered in the audio reproduction in these two laboratories and will also present a broad overview of solutions used and developed to ensure the validity of the experimental conditions.

9:00

2aPPb4. Speech intelligibility and hearing aid benefit in a living room: Comparison of real and simulated acoustics. Julia Schütze (Medical Phys. and Cluster of Excellence Hearing4all, Carl von Ossietzky Universität Oldenburg, Ammerländer Heerstraße 114-118, Oldenburg 26129, Germany, julia.schuetze@uol.de), Stephan D. Ewert, Christoph Kirsch, and Birger Kollmeier (Medical Phys. and Cluster of Excellence Hearing4all, Carl von Ossietzky Universität Oldenburg, Oldenburg, Germany)

For hearing-impaired patients, a disparity between hearing aid benefit determined by conventional audiological assessment and observed in everyday life has been reported. Hence, ecologically valid testing methods to better reflect performance in real-world scenarios are required. Hereby, the living room is a highly relevant (home) environment for speech communication involving various target and interfering sources. This study examines speech intelligibility in an average German living room with a connected kitchen and acoustic reproductions of the real room using loudspeakers. Speech recognition thresholds were measured using the Oldenburg Sentence Test (OLSA) and a male-transformed version of the ISTS (International Speech Test Signal) as a masker. Aided and unaided hearing-impaired listeners, as well as a normal-hearing control group, participated. Three target positions were investigated, with the most challenging target position in the adjacent kitchen with an obstructed direct sound path. Interactive room acoustics simulation (allowing for head movements) was used to replicate the living room via a small-scale (four) loudspeaker array, as applicable in the clinical environment, and a large-scale, three-dimensional array with 86 loudspeakers. The study discusses the impact of room acoustics on speech intelligibility at different target positions and the relationship between real and simulated environments regarding hearing aid benefit.

9:20

2aPPb5. Using simple virtual environments to study the differentiation and integration of sensory cues in complex scenes. G. Christopher Stecker (Ctr. for Hearing Res., Boys Town National Res. Hospital, 555 N 30th St., Omaha, NE 68131, cstecker@spatial-hearing.org)

The promise of virtual environments for studying sensory perception and cognition cannot be overstated. Because such environments can approximate aspects of the real world while affording a high degree of experimental and stimulus control, they promise to "bring the real world into the lab" and vice-versa [Stecker 2019, *Hear J*. **72**(6):20-23], helping to make research and clinical testing more valid and relevant to real-world situations. Such benefits accrue to all sorts of investigations, and in this presentation we focus on relatively low-level aspects of auditory cognition—specifically the ability of listeners to differentiate sensory cues belonging to different objects in a complex scene (e.g., talkers) and to integrate information across disparate partially informative cues. Across multiple studies, we investigate the potential for virtual environments to enhance this work. A particular focus on simplified environments aims to elucidate a minimum feature set supporting different aspects of reality-like performance. [Work supported by US NIH R01-DC016643.]

9:40

2aPPb6. Association between interaural phase sensitivity and performance on the trail making cognitive test. Pavel Zahorik (Dept. of Otolaryngol. and Communicative Disord., Dept. of Psychol. and Brain Sci., Univ. of Louisville, Louisville, KY 40292, pavel.zahorik@louisville.edu) and Olaf Strelcyk (Sonova U.S. Corporate Services, Aurora, IL)

Sensitivity to interaural differences in time or phase critically supports human abilities to spatially localize sounds in the horizontal plane. The trail making test (TMT) is a non-auditory cognitive test that provides information about visual search speed, visual/spatial attention, and executive function. Surprisingly, we have shown strong positive relationships between interaural phase difference (IPD) sensitivity and TMT performance, both in a sample ($n = 20$) of older adults (48–85 years) with very similar sensorineural hearing loss profiles [Strelcyk *et al*., **23**, 2331216519864499 (2019)], and in a sample ($n = 41$) of young to middle-aged adults (21–54 years) with normal audiograms [Shehorn *et al*., Hear. Res. **392**, 107982 (2020)]. The fact that associations between IPD sensitivity and TMT performance were observed in both samples suggests that the associations were not related to audiometric sensitivity. Alternative explanations for the associations will be discussed, including potential higher-order relationships with spatial processing abilities and/or cognitive processing speed.

10:00

2aPPb7. Live evaluation of auditory preference for group conversations. Petra Herrlin (ORCA Europe, WS Audiol., Stockholm, Sweden), Frédéric Marmel, Florian Wolters (ORCA Europe, WS Audiol., Stockholm, Sweden), and Karolina Smeds (ORCA Europe, WS Audiol., Bjorns Tradgardsgrand 1, Stockholm SE-116 21, Sweden, karolina.smeds@orca-eu.info)

Improving hearing-impaired people's ability to participate in conversations is important. To date, few outcome measures let us relate conversation success to hearing impairment and hearing-aid function. In a previous study, we developed the LEAP test (Smeds *et al*. 2021), where a test leader guided one participant through a set of listening scenarios, in which the participant compared two hearing-aid settings and reported the preferred setting. The current study extends the LEAP test to unsupervised group conversations. Nine groups of three participants held conversations sparked by so-called consensus questions. Two test scenarios were used: "business meeting" (where a projector created low-level background noise) and "dinner party" (where an ambisonics recording reproduced in 2-D was used as background). Two test paradigms were evaluated: *direct comparisons*, where participants toggled between the two hearing-aid settings during a conversation and then selected their preferred setting, and *indirect comparisons*, where one hearing-aid setting was used at the time and participants after each conversation rated how well the hearing aids worked in the current setting, and the preferred setting was determined by comparing the two ratings. The reliability of the two test paradigms was evaluated. At the time of the submission of this abstract, analyses are ongoing.

Session 2aSA

Structural Acoustics and Vibration, Engineering Acoustics, and Physical Acoustics: Acoustic Metamaterials I

Christina Naify, Cochair
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Chair's Introduction—9:55

Invited Papers

10:00

2aSA1. Acoustic cloaking via a hybrid passive–active technique. William J. Parnell (Dept. of Mathematics, Univ. of Manchester, Oxford Rd., Manchester, Greater Manchester M13 9PL, United Kingdom, william.parnell@manchester.ac.uk), Cikai Lin, and Nicole Kessissoglou (School of Mech. and Manufacturing Eng., UNSW Sydney, Kensington, New South Wales, Australia)

Both passive and active methods have been employed extensively in recent times to create the effect of acoustic cloaking, with obstacles being made near-invisible over specific frequency ranges. Passive methods typically rely on metamaterials with complex microstructures, whereas active methods require an external power supply. Furthermore, a large number of active sources and sensors are typically required for active control. In this work, a hybrid method that combines passive and active approaches is developed. The passive acoustic cloak is designed theoretically via layered shells with alternating, functionally graded properties. The layer properties are informed by coupling transformation acoustics with homogenization theory and require unphysical values on the inner cloak wall. As such, some scattering occurs from the obstacle to be cloaked as well as due to the discontinuity in material properties across the shell boundaries. Using distributed monopole control sources, active control is then employed to nullify the non-zero scattered field outside the passive cloak domain. Effective cloaking is achieved using the hybrid passive–active control system.

10:20

2aSA2. Design of acoustic coatings with symmetric and asymmetric resonant inclusions. Alexander McIntosh (School of Mech. and Manufacturing Eng., UNSW Sydney, Ainsworth Bldg., Eng. Rd., Kensington, Kensington, New South Wales 2052, Australia, alexander.mcintosh@unsw.edu.au), Gyani S. Sharma, Alex Skvortsov, Ian MacGillivray (Platforms Div., Defence Sci. and Technol. Group, Melbourne, Victoria, Australia), and Nicole Kessissoglou (School of Mech. and Manufacturing Eng., UNSW Sydney, Kensington, New South Wales, Australia)

An acoustic coating for underwater applications, designed using periodically distributed cavities and hard spherical particles embedded in a soft matrix, is herein modelled as monopolar and dipolar scatterers positioned along the centre of a sound hard duct. Modified equations of motion for the scatterers account for resonance phenomena as well as multiple scattering of waves by a lattice. Combinations of monopolar and dipolar scatterers in the direction of the incident sound field are considered. The acoustic performance of the various coating designs composed of symmetric and asymmetric resonant inclusions are compared. The influence of asymmetry in coating designs, arising from variation in scatterer size and material distribution, on effective material properties and Willing coupling are reported.

10:40

2aSA3. Wave transmission through acoustic lossy barrier by using non-Hermitian complementary metamaterial. Ki Yong Lee (Mech. Eng., Korea Adv. Inst. of Sci. and Technol., KAIST, 291 Daehak-ro, Yuseong-gu, Daejeon 34141, Republic of Korea, Daejeon 34141, Korea (the Republic of), moonkross@kaist.ac.kr) and Wonju Jeon (Mech. Eng., Korea Adv. Inst. of Sci. and Technol., Daejeon, Republic of Korea)

Complementary metamaterials (CMMs), characterized by negative effective mass density and moduli within a specific frequency range, hold the promise of significantly enhancing wave energy transmission across acoustic barriers. These materials operate by acoustically counteracting the barrier, thus offering an efficient conduit for wave propagation. However, when the intrinsic loss factor of the acoustic barrier is considered, the transmission performance of CMMs experiences a marked decrease. This decrement is notably severe, surpassing the theoretical expectations based solely on acoustic dissipation within the barrier. Delving into the depth of this issue, our research scrutinizes the far-reaching implications of the loss factor on wave transmission and takes a critical step forward by extending the effective properties of the complementary metamaterials from real to complex domain, an approach embodying the concept of non-Hermitian physics. Subsequently, we introduce a non-Hermitian CMM, a groundbreaking solution that maintains high transmittance despite the barrier's loss. It adeptly reconciles the impedance mismatch between the lossy barrier and the background medium, ultimately enabling perfect transmission.

11:00

2aSA4. Reciprocity, passivity, and causality in fully coupled acousto-electrodynamic media. Chirag A. Gokani (Walker Dept. of Mech. Eng. and the Appl. Res. Labs., Univ. of Texas at Austin, 10000 Burnet Rd., Austin, Texas 78758, Austin, TX, chiragokani@gmail.com), Samuel P. Wallen, and Michael R. Haberman (Walker Dept. of Mech. Eng. and the Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX)

Multiple-scattering homogenization procedures in acoustics and electromagnetics have shown that subwavelength asymmetries and lattice effects in metamaterials give rise to *bianisotropy*, the tensorial coupling of two constitutive relations. Asymmetric piezoelectric scatters in the quasi-electrostatic regime have recently been shown to couple the electric field to the bianisotropic elastodynamic relations, yielding an additional, emergent property termed *electro-momentum coupling*. To facilitate the study of scatterers responsible for electro-momentum coupling, Wallen *et al.* [Proc. Mtgs. Acoust. **46**, 065002 (2022)] and Lee *et al.* [J. Appl. Phys. **132**, 125108 (2022)] abandoned the quasi-electrostatic approximation in favor of a simultaneously acoustic and electrodynamic scatterer. We previously used a multiple-scattering homogenization procedure to show that a collection of such scatterers gives rise to constitutive relations that fully couple acoustics and electrodynamics. Now, we derive bounds on these fully coupled relations due to reciprocity, passivity, and causality. These results recover known bianisotropic and piezoelectric bounds and reveal that the imaginary parts of the additional electro-momentum and magneto-momentum coupling coefficients vanish in lossless, reciprocal, and passive media. The bounds also verify that the constitutive relations derived by our homogenization procedure satisfy reciprocity, passivity, and causality. [Work supported by DARPA and ARL:UT McKinney Fellowship in Acoustics.]

11:20

2aSA5. Homogenization and analysis of planar piezoelectric metamaterials: A comprehensive study. Guosheng JI (Dept. of Eng. Sci., the Univ. of Oxford, Trinity College, Broad St., Oxford OX1 3BH, United Kingdom, guosheng.ji@eng.ox.ac.uk) and John Huber (Dept. of Eng. Sci., the Univ. of Oxford, Oxford, United Kingdom)

This article presents a comprehensive investigation into the homogenization and analysis of planar piezoelectric metamaterials. The classical transfer matrix method is extended by using homogenization methods to evaluate the sound transmission of piezoelectric metamaterials in both normal and oblique incidences. This has been validated numerically using the finite element method and experimentally in the solid medium propagation in normal incidence. Several vibro-acoustic analytical models are compared in oblique incidence, including the Kirchhoff thin plate theory, the Reissner-Mindlin thick plate theory, and the theory of wave propagation in elastic solids. These theories are used to determine the dispersion relation, coincidence, and transition frequency of thin and thick plate theories in analyzing piezoelectric metamaterials. Additionally, acoustic properties of piezoelectric layers connected to external circuits are parametrically studied, exploring both dimensional and dimensionless variables. Results indicate that significant control over the resonance frequency and sound transmission can be enabled by adjusting the external electrical impedance, flat-layer structures, and piezoelectric materials. This demonstrates excellent tunability and compactness of planar piezoelectric metamaterials for space-sensitive applications. The study indicates a straightforward and powerful analytical approach for the optimization of acoustic insulation using planar piezoelectric metamaterials.

11:40

2aSA6. Non-reciprocal selective frequency transmission with nonlinear bi-layer cylinder chains. Beomseok Oh (Mech. Eng., Pohang Univ. of Sci. and Technol., Bldg. 5, Pohang 37673, Republic of Korea, bs.oh@postech.ac.kr), Yeongtae Jang, and Junsuk Rho (Mech. Eng., Pohang Univ. of Sci. and Technol., Pohang, Republic of Korea)

Recent advancements in wave physics have led to the development of systems that break the principle of wave transmission reciprocity. In this study, we present selective frequency transmission with asymmetry in bi-layer phononic crystals, which are composed of two layers of cylinder contact chains. We demonstrate significant non-reciprocal harmonic transmission effects, as well as self-demodulated displacement amplification under excitation from different sides. These effects are achieved through a combination of contact nonlinearity and cascaded bandgaps formed by adjusting the chain lengths in both layers. We find that the evanescent waves generated through nonlinear frequency conversion in the stopband are converted into propagating waves at the interface, subsequently leading to the transmission. Our findings have implications for programmable asymmetric acoustic devices, providing a novel framework for energy mitigation, conversion, and harvesting.

Session 2aSC

Speech Communication: Listening Challenges in Different Learning Environments

Kiri Mealings, Cochair
Macquarie Univ., Level 1 Australian Hearing Hub, 16 University Ave., 2109, Australia

Benjamin V. Tucker, Cochair
Communication Sciences and Disorders, Northern Arizona Univ., 208 E. Pine Knoll Dr., PO Box 15045, Flagstaff, AZ 86011

Contributed Paper

8:00

2aSC1. Enhancing understanding of real-world listening experiences: Insights from ecological momentary assessments with assistive listening technologies. Jorge Mejia (Signal Processing, National Acoust. Labs. (NAL), Hearing Australia, Macquarie Univ. NSW, Level 4, Australian Hearing Hub,16 University Ave., Sydney, New South Wales 2109, Australia, jorge.mejia@nal.gov.au), David Meng, and ARUN SEBASTIAN (Signal Processing, National Acoust. Labs. (NAL), Hearing Australia, Macquarie Park, Sydney, New South Wales, Australia)

Comprehending the daily listening experiences of people who use assistive listening technologies is difficult due to individual differences and the variety of listening environments they encounter. Consequently, a lack of comprehensive insight into real-world experiences often leads to misconstrued interpretations of research outcomes and future technological directions. Addressing this issue, the National Acoustic Laboratories (NAL) in Sydney, Australia, has been actively investigating the applications of ecological momentary assessments (EMA) since 2010. By employing EMA, NAL aims to comprehend the impact of assistive listening technologies in real-world conditions. To achieve this, we invested data from four distinct studies, encompassing more than 3000 survey entries. Utilizing an EMA app specifically developed by NAL, participants reported various challenges they faced in their listening experiences. These challenges include difficulties in understanding conversation when listening to a single speaker or in multitasking scenarios, as well as frustration with the loudness and annoyance of background noise. The analysis of this extensive dataset involved a combination of conventional statistical methods and advanced machine learning models. By employing these techniques, key insights were extracted from the data, offering valuable guidance for the design of future research conducted in real-world settings.

Invited Papers

8:20

2aSC2. Improved acoustics puts students on the autism spectrum in a better position to learn, but they still have to learn. Wayne J. Wilson (Audiol., Discipline of Audiol., School of Health and Rehabilitation Sci., The Univ. of Queensland, Queensland 4072, Australia, w.wilson@uq.edu.au), Keely Harper-Hill (Queensland Univ. of Technol., Queensland, Australia), Rebecca Armstrong, Cerys Downing (The Univ. of Queensland, Queensland, Australia), Kelsey Perrykkad (Monash Univ., Victoria, Australia), Mary Rafter (The Univ. of Queensland, Queensland, Australia), and Jill Ashburner (Autism Queensland, Queensland, Australia)

Over the past decade, researchers have increasingly argued that improved acoustics can lead to improved classroom performance for students on the autism spectrum. This paper reviews these arguments to find stronger evidence of proximate benefits for acquiring skills lower on the listening hierarchy (such as sound detection and phonemic awareness) and weaker evidence of distant benefits for acquiring skills higher on the listening hierarchy (such as language and learning). These findings suggest that improved acoustics can put students on the spectrum in a better position to learn, but learning must still take place over time. These findings warrant case-by-case consideration for improving acoustics to benefit students on the spectrum, with realistic expectations of the benefits that can be expected from improved acoustics alone.

8:40

2aSC3. The Listen to Learn for Life (L3) assessment framework. Kiri Mealings (Macquarie Univ., Level 1 Australian Hearing Hub, 16 University Ave., New South Wales 2109, Australia, kiri.mealings@mq.edu.au), Kelly Miles, Alan Kan, Ronny Ibrahim, and Joerg M. Buchholz (Macquarie Univ., Macquarie Park, New South Wales, Australia)

Listening is the primary gateway for children to learn in the mainstream classroom, but the dynamics and noise of modern classrooms can make listening challenging. This is especially true for children with hearing loss, language and communication difficulties, attention deficits, autism, other learning needs, and/or those communicating in a language other than their native language. It is, therefore, critical for researchers to realistically assess how children listen to learn in the classroom and to understand how listening can be improved to enhance children's learning and wellbeing. The Listen to Learn for Life (L^3) assessment framework is a tool to use when conducting this research which holistically incorporates frameworks from health, speech and hearing sciences, and education sectors.

The L^3 assessment framework has three main components: characterisation of activity (perceptual setting of lecture, group work, or individual learning activity), functioning assessment (hearing, listening, comprehending, or communicating), and impact (learning and wellbeing). These are affected by External and Internal Influences. Here, we present selected examples of how to apply the framework to assess children's listening, learning, and wellbeing during different classroom activities as well as determine the effectiveness of a chosen intervention.

9:00

2aSC4. Voice quality and its effects on university students' listening effort in a virtual seminar room. Isabel S. Schiller (Work and Eng. Psych., RWTH Aachen Univ., Jaegerstrasse 17-19, Aachen 52066, Germany, isabel.schiller@psych.rwth-aachen.de), Lukas Aspöck, Carolin Breuer (Inst. for Hearing Technol. and Acoust., RWTH Aachen Univ., Aachen, Germany), Jonathan Ehret, Andrea Bönsch (Visual Computing Inst., RWTH Aachen Univ., Aachen, Germany), Janina Fels (Inst. for Hearing Technol. and Acoust., RWTH Aachen Univ., Aachen, Germany), Torsten W Kuhlen (Visual Computing Inst., RWTH Aachen Univ., Aachen, Germany), and Sabine J Schlittmeier (Work and Eng. Psych., RWTH Aachen Univ., Aachen, Germany)

A teacher's poor voice quality may increase listening effort in pupils, but it is unclear whether this effect persists in adult listeners. Thus, the goal of this study is to examine the impact of vocal hoarseness on university students' listening effort in a virtual seminar room. An audio-visual immersive virtual reality environment is utilized to simulate a typical seminar room with common background sounds and fellow students represented as wooden mannequins. Participants wear a head-mounted display and are equipped with two controllers to engage in a dual-task paradigm. The primary task is to listen to a virtual professor reading short texts and retain relevant content information to be recalled later. The texts are presented either in a normal or an imitated hoarse voice. In parallel, participants perform a secondary task which is responding to tactile vibration patterns via the controllers. It is hypothesized that listening to the hoarse voice induces listening effort, resulting in more cognitive resources needed for primary task performance while secondary task performance is hindered. Results are presented and discussed in light of students' cognitive performance and listening challenges in higher education learning environments.

9:20

2aSC5. Language processing speed in school students with hearing loss: A review. Rebecca Holt (Macquarie Univ., 16 University Ave., New South Wales 2109, Australia, rebecca.holt@mq.edu.au), Trudy Smith, and Greg Leigh (NextSense Inst., Sydney, New South Wales, Australia)

For effective learning in classroom settings, it is critical that students comprehend spoken and written language rapidly. If language processing is slow, children may miss content in lessons, struggle to participate effectively in group work, and be disadvantaged in high-stakes assessment tasks. Rapid language processing may be challenging for students with hearing loss using hearing aids or cochlear implants. They may process language more slowly than peers with normal hearing due to either contemporaneous signal degradation from their hearing devices, or "fuzzy" mental representation of language resulting from degraded input during language acquisition. A better understanding of language processing speed in students with hearing loss is urgently required to provide the basis for appropriate classroom support strategies and provisions in assessment tasks. This review of the current state of knowledge on language processing speed in school-age children with hearing loss will demonstrate that while some research does find slow language processing at the word and sentence level in this population, this may be modulated by factors including task difficulty and degree of hearing loss. Priority areas for future research, including processing speed at the paragraph/discourse level and the relationship between spoken and written language processing speed, will also be highlighted.

Contributed Papers

9:40

2aSC6. Speech communication in outdoor learning environments. Christian Vossart (GHD NZ, 27 Napier St., Level 3, GHD Ctr., Auckland 1011, New Zealand, christian.vossart@ghd.com)

Newton Central School (NCS) celebrates its 100th birthday this year. Its noise environment has evolved significantly over that time and is now dominated by road-traffic noise from the adjacent State Highway 16. Master planning for the redevelopment of NCS, to enable the school roll to be more than doubled, necessitates consideration of how noise emissions to the site can best be minimised and managed, particularly for outdoor learning areas. The transmission of road traffic noise across the site (up to 2043) is predicted and the methodology is detailed. The use of the local topography, locations, and arrangement of buildings and open space, and zoning of different types of outdoor activity to both mitigate and manage transmission is explored. The resultant levels of ambient noise achieved through different mitigation options are discussed via comparison with different sources of published design criteria for outdoor learning and other activities within open spaces. The results are also considered in terms of a signal-to-noise relationship. The relative benefits of reductions in noise for the end user/stakeholder are clarified with reference to the likely subjective perception of comparative changes in noise level. The application of lessons learned and the potential for further study are investigated.

10:00

2aSC7. Cortical processing of Mandarin lexical tone in children with different language backgrounds. Samantha Cajas, Rigel Baron, Andres Diaz, Sharry Guo, Maria Bianchi (St. John's Univ., Queens, NY), and Yan H. Yu (Commun. Sci. & Disord., St. John's Univ., 4631 216 St., Bayside, NY 11361, yanhyu@gmail.com)

Bilingual experience modulates cognitive skills, particularly executive control. However, it is not clear if such modulation can be observed in automatic speech processing. Furthermore, it is unknown if bi-dialectal experience will enhance speech processing. This study aims to investigate whether and how early bilingual experience influences cortical sensitivity for automatic cortical processing of speech, specifically in the context of tonal and non-tonal languages. A passive listening oddball paradigm with two lexical tone contrasts were used and event-related potentials were recorded. An easy contrast (tone 1 versus tone 3) and a hard contrast (tone 2 versus tone 3) were used as the stimuli. Children between 5 and 10 years of age were tested. Preliminary results indicate that monolingual mainstream English, African-American English, and bilingual Mandarin-English learners all showed similar brain responses to the easy lexical tone contrast. However, there was an age effect across language groups. Such results suggest that biological factors such as age are stronger predictors of brain responses to lexical tone contrasts comparing to the factor of language background.

Session 2aSP

Signal Processing in Acoustics, Noise and Physical Acoustics: Signal Processing for Active Sound and Vibration Control I

Sipei Zhao, Cochair
Ctr. for Audio, Acoust. and Vib., Univ. of Technology Sydney, 32-34 Lord St., UTS Tech Lab, Botany 2019, Australia

Yijing Chu, Cochair
State Key Lab. of Subtropical Building and Urban Science, South China Univ. of Technol., Guangzhou 510641, China

Chair's Introduction—8:35

Contributed Paper

8:40

2aSP1. An active noise control algorithm based on an improved convex combination loop. Xu Yang (School of Mech. and Automotive Eng., Qingdao Univ. of Technol., Qingdao, China) and Tian Ran Lin (School of Mech. and Automotive Eng., Qingdao Univ. of Technol., 777 Jialingjiang Rd., West Coast New Distr., Qingdao, Shandong 266520, China, trlin888@163.com)

The conflict between a faster convergence using a Filtered-X Least Mean Square (FXLMS) algorithm and the requirement of a small steady-state error of an adaptive active control (ANC) system which limits its performance in terms of computation efficiency and the control error. The problem can be solved by using two paralleling adaptive filters to form a convex combination loop in the control system. However, the exponential weight function used in the original convex combination loop design can consume a large computer power which has limited its application. As a result, a new weight function is proposed to replace the original exponential function in the original convex combination loop to enhance the convergence rate of the adaptive ANC system. Furthermore, a MRFXLMS algorithm is used to replace the FXLMS algorithm in the ANC system so that it can be used for impulse noise control. The result shows that the proposed algorithm performs well in the control of input impulse noise with different intensity.

Invited Papers

9:00

2aSP2. Robust improved multiband-structured subband adaptive filter for active noise control with impulsive interference. Somanath Pradhan (Elec. Eng., Indian Inst. of Technol. Patna, Office Room: 03/122, First Fl., Dept. of Elec. Eng., Block-III, Patna, Bihar 801106, India, pradhan.somnath@gmail.com), Guoqiang Zhang, Xiaojun Qiu, J. C. Ji, and Jeffrey Parnell (Ctr. for Audio, Acoust. and Vib., Univ. of Technol. Sydney, Sydney, New South Wales, Australia)

The feedforward filtered-x least mean square algorithm is extensively implemented for active control of broadband noise. However, the control performance is substantially deteriorated due to colored noise and the presence of uncorrelated disturbances near the reference and error sensors. To tackle these issues, in this paper, a robust improved multiband-structured subband adaptive filter based on logarithmic and total least squares method is proposed for active control. Unlike the conventional method of total least squares, the proposed algorithm adopts logarithmic and Rayleigh quotient functions. The closed loop implementation of the improved multiband-structured subband adaptive filter is adopted to meet the delayless requirement. The proposed algorithm is well-suited for environments where the reference signal is highly correlated and the residual noise is contaminated by impulsive noise. Furthermore, an affine combination of the proposed algorithm is developed to meet the complementary requirements of faster convergence and improved noise reduction. Eventually, the stability and computational complexity are studied. Simulation results using measured acoustic paths in an anechoic chamber and a normal room illustrate the effectiveness of the proposed algorithm for controlling broadband noise with impulsive interference. In addition, the tracking control performance is evaluated in a time-varying acoustic environment.

9:20

2aSP3. Analysis of distributed active noise control systems. Sipei Zhao (Ctr. for Audio, Acoust. and Vib., Univ. of Technol. Sydney, 32-34 Lord St., UTS Tech Lab, Botany, New South Wales 2019, Australia, Sipei.Zhao@uts.edu.au)

Active noise control technology has witnessed a great success in the past decades for local noise control in headsets and similar devices. However, many challenges are yet to be tackled for active control of noise over a large spatial region, where multichannel systems are needed. One of the significant issues is the high computational complexity of multichannel adaptive algorithms running on a centralized processor. To overcome this problem, distributed and decentralized adaptation strategies have been proposed to spread the computational burden over multiple processors recently. This paper will review existing distributed and decentralized active noise control systems based on the filtered-reference least mean square algorithms and analyze their convergence behavior under a unified framework. The convergence conditions in the mean and mean-square sense are studied and the transient and stead-state mean square errors are also analyzed. Simulations are performed to verify the proposed framework and analysis. This work will provide insights into current distributed active noise control systema and shed light on developing new algorithm design in the future.

Contributed Papers

9:40

2aSP4. Updating the secondary path using deep learning models to enhance the performance of active road noise control. Jun Young Oh (Mech. Eng., Seoul National Univ., 1, Gwanak-ro, Gwanak-gu, Seoul 08791, Republic of Korea, eric6518@snu.ac.kr), Hyun Woo Jung, Yeon June Kang (Mech. Eng., Seoul National Univ., Seoul, Republic of Korea), Myung Han Lee, and Kyoung Hoon Lee (Hyundai Motor Co., Hwaseong-si, Republic of Korea)

This research investigates the importance of accurately updating the secondary path estimation to enhance the performance of active road noise control (ARNC), particularly in response to variable conditions within a vehicle. The study analyzes the impact of four spatial conditions on the secondary path and introduces prediction models using Delaunay triangulation-based interpolation and deep learning methods, using multi-layer perceptron (MLP) networks. MLP networks were constructed based on the representation domain chosen for the secondary path. The deep learning methods demonstrated higher prediction accuracy and smaller data storage requirements compared to the interpolation method. Among these deep learning methods, the approach that stood out as the best performer was representing the secondary path through principal component analysis (PCA) and learning the weight of each basis vector. Through this technique, it was experimentally confirmed that the performance of ARNC under variable conditions in the vehicle improved through the accurate update of real-time secondary path estimation. These advantages are particularly prominent when dealing with high frequencies, which are likely to be the foundational technology for the frequency expansion of ARNC.

10:00–10:20 Break

10:20

2aSP5. Dynamic neural network switching for active noise control of nonlinear systems. Xander Pike (ISVR, Univ. of Southampton, University Rd., Southampton, Hampshire SO17 1BJ, United Kingdom, X.Pike@soton.ac.uk) and Jordan Cheer (ISVR, Univ. of Southampton, Southampton, United Kingdom)

Advancements in digital technologies have allowed for the development of increasingly complex active noise and vibration control solutions, with a wide range of applications. Such control systems are commonly designed using linear digital filters which cannot fully capture the dynamics of nonlinear systems. To overcome this, it has previously been shown that replacing linear controllers with neural networks (NNs) can improve control performance in the presence of nonlinearities in both the system plant and primary path. Inferring the outputs of NN controllers can, however, be computationally expensive, limiting the practicality and real-time performance of such control systems. It has been demonstrated previously that, with nonlinearity in the primary path, the range of nonlinearity over which a single NN controller can achieve significant levels of control is limited by the size of the NN. In this paper, a method of dynamically switching between multiple smaller NN controllers is presented. In a simple time-discrete simulation, the performance and computational cost of this approach is compared to that of individual fixed NN controllers, demonstrating improved computational efficiency and performance over a range of nonlinearity as the system disturbance changes over time.

10:40

2aSP6. A virtual sensing method based on plane-wave assumptions for active noise control headrests. Toma Yoshimatsu (The Univ. of Electro-Communications, 1-5-1, Chohugaoka, Chohu ci, Tokyo 182-8585, Japan, y.toma.0613@gmail.com), Naoki Shinobu (The Univ. of Electro-Communications, Tokyo, Japan), Hiroaki Itou, Shihori Kozuka, Noriyoshi Kamado (Nippon Telegraph and Telephone Corp., Tokyo, Japan), and Yoichi Haneda (The Univ. of Electro-Communications, Tokyo, Japan)

Active noise control (ANC) headrests have been studied to suppress cabin noise and achieve a comfortable sound environment in public transportation mobility. To improve the suppression performance at the ear, this system requires virtual sensing technology to estimate sound pressure at the ear from microphone observations on the headrest. In this study, we propose a virtual sensing method for ANC headrests based on the assumption that noise arrives as plane-wave. The proposed method assumes that the cross-correlation between the observed signals of two adjacent microphones located on a grid point in space is constant and independent of absolute position. Based on this assumption, the cross-correlation estimated from observations between two microphones far from the ear is used to predict the noise propagation from the microphones near the ear to the ear. When noise reaches the ear first, the estimated cross-correlation becomes non-causal. To address this issue, we propose a method for predicting noise at the ear that involved removing pure delays in the impulse response of primary path estimated from observations of a reference microphone placed near the noise source and a microphone placed far from the ear. The effectiveness of the proposed method is demonstrated by computer simulation.

Invited Paper

11:00

2aSP7. Local active control of road noise with door loudspeakers in a car cabin. Shuping Wang (Key Lab. of Modern Acoust. and Inst. of Acoust., Nanjing Univ., No. 22 Hankou Rd., Nanjing, Jiangsu 210093, China, shuping.wang@nju.edu.cn), Xinfa Xiao, Jiancheng Tao (Key Lab. of Modern Acoust. and Inst. of Acoust., Nanjing Univ., Nanjing, Jiangsu, China), and Xiaojun Qiu (Key Lab. of Modern Acoust. and Inst. of Acoust., Nanjing Univ., Sydney, New South Wales, Australia)

Active control of road noise in car cabins has received much attention in recent years and it has been well known that local quiet zones can be created with headrest loudspeakers. The local control performance achieved with door loudspeakers is investigated in this manuscript based on the measurements in a car cabin, and it is compared with that achieved with two headrest speakers. The noise reduction performance at error microphones, the upper limit frequency of effective control, as well as the size of the quiet zone are investigated in detail. The difficulties and challenges of using the door loudspeakers for local control are discussed.

Contributed Paper

11:20

2aSP8. Time domain virtual sensing method based on a rigid-sphere transfer function for active noise control headrests. Naoki Shinobu (The Univ. of Electro-Communications, 1-5-1 Chofugaoka, Chofu, Tokyo 182-8585, Japan, s2230068@edu.cc.uec.ac.jp), Toma Yoshimatsu (The Univ. of Electro-Communications, Tokyo, Japan), Hiroaki Itou, Shihori Kozuka, Noriyoshi Kamado (Nippon Telegraph and Telephone Corp., Tokyo, Japan), and Yoichi Haneda (The Univ. of Electro-Communications, Tokyo, Japan)

Active noise control (ANC) is expected to be used in public transportation such as airplanes and trains. While using these applications, passengers prefer not to wear earphones or ear microphones, if possible, to reduce ear strain. However, conventional ANC systems require a physical microphone, which acts as an error sensor, at the ear position. To address this issue, a virtual sensing technique has been studied that uses a microphone mounted on a seat's headrest to estimate the sound pressure at the ear position. In this study, we propose a virtual sensing technique that interpolates the sound pressure at the ear position using a spherical microphone array surrounding the head. This method improves robustness to variations in noise arrival direction using spherical harmonic coefficients without directional dependence. In general, the interpolation formula using spherical harmonic expansion is obtained in the frequency domain. By contrast, the proposed method designs the interpolation process as an IIR filter by considering the minimum phase and estimates the noise signal directly in the time domain, achieving the real-time processing required for ANC. The effectiveness of the proposed method in reducing noise was investigated for various noise arrival directions.

Invited Paper

11:40

2aSP9. A hat-shape error microphone array for user-friendly spatial active noise control. Huiyuan Sun (School of Elec. and Information Eng., Univ. of Sydney, Sydney, New South Wales, Australia, huiyuan.sun@sydney.edu.au), Craig Jin (School of Elec. and Information Eng., Univ. of Sydney, Sydney, New South Wales, Australia), Jihui (Aimee) Zhang (Inst. of Sound and Vib., Univ. of Southampton, Southampton, United Kingdom), Prasanga N. Samarasinghe, and Thushara Abhayapala (School of Eng., The Australian National Univ., Canberra, Australian Capital Territory, Australia)

Spatial active noise control (ANC) has been developed in the past decades to reduce unwanted acoustic noise over a desired region by generating an anti-noise field with secondary loudspeakers. Unlike conventional multi-channel ANC systems, the integration of spherical harmonic-based sound field analysis allows the ANC system to reduce noise over a continuous region surrounding the user's head rather than discrete points. However, the use of a spherical error microphone array on the boundary of the region for spherical harmonic analysis poses a limitation as it obstructs user access to the controlled area. In this work, we introduce a hat-shaped error microphone array design which can be placed above the listener's head. The proposed error microphone array can achieve spherical harmonic-based residual noise field analysis and is more user-friendly, allowing unrestricted user head movement. We construct the hat-shaped error microphone array and present the results of ANC experiments conducted using the array.

Session 2aUW

Underwater Acoustics and Acoustical Oceanography: Effects of Shear Waves on Propagation and Scattering of Underwater Sound I

Oleg A. Godin, Cochair
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Alex Skvortsov, Cochair
Platforms Div., Defence Sci. and Technology Group, Melbourne, Australia

Chair's Introduction—8:15

Invited Papers

8:20

2aUW1. Shear speed in Arctic ice and underwater acoustic reflection. Nicholas P. Chotiros (The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, chotiros@utexas.edu), Gaye Bayrakci (National Oceanogr. Ctr., Southampton, United Kingdom), Oliver Sanford, Timothy Clarke (Defence Sci. and Technol. Lab., Porton Down, United Kingdom), and Angus Best (National Oceanogr. Ctr., Southampton, United Kingdom)

The underwater reflected sound wave under a sheet of Arctic ice contains information about the ice including thickness and mechanical properties. The ideal case of a perfectly flat ice floe floating above the water may be modeled using the OASES wavenumber integration code. A spectrogram of the acoustic response contains features related to the compressional and shear waves. The resonance associated with the shear wave speed in the ice is particularly distinctive. In the real world, the ice-water interface is not perfectly smooth. The finite element code SPECFEM2D is used to simulate the response of ice with a rough ice–water interface. It is particularly well suited to this problem since it accounts for all orders of multiple reflections. It shows the effects of fine-scale roughness on the acoustic response. Depending on the severity of the roughness, it may enhance or diminish the acoustic features related to the properties of the ice. [Work supported by the U. S. Office of Naval Research, Code 32 Grant N00014-20-1-2041 (N. Chotiros). G. Bayrakci and A. Best were funded by the UK Defence and Security Accelerator, Grant ACC2016927. The Texas Advanced Computing Center provided the high-performance computing resources.]

8:40

2aUW2. Sea ice mechanical property inference using cryophones and coherent signals. D. Benjamin Reeder (Oceanogr., Naval Postgrad. School, 833 Dyer Rd., Bldg. 232, SP-311B, Monterey, CA 93943, dbreeder@nps.edu), John Joseph, and S. Kyle Wheeler (Oceanogr., Naval Postgrad. School, Monterey, CA)

Mechanical properties of Arctic sea ice can be inferred by observation of the speeds of compressional, shear and flexural waves generated through in-ice conversion of impulsive energy. In prior work, the impulsive signal was generated by a lead ball or sledge hammer dropped onto the top of the sea ice, and the inversion process required meticulous, manual extraction of signal amplitudes. The work presented here makes use of (a) coherent sources with which broadband signals can be generated to replace the manually generated hammer-drop signals and improve accuracy via matched filter, as well as (b) recent observations and modeling of ice sheet compressional resonances from which ice thickness can be more easily inferred. Analysis and modeling of observations from a recent field experiment in the Beaufort Sea are shown that demonstrate the potential capability of remote monitoring of sea ice mechanical properties.

9:00

2aUW3. Sound absorption by a soft medium embedded with complex-shaped hard inclusions. Gyani Shankar Sharma (Platforms Div., Defence Sci. and Technol. Group, Sydney, New South Wales, Australia), Alex Skvortsov (Platforms Div., Defence Sci. and Technol. Group, Melbourne, Victoria 3207, Australia, alexei.skvortsov@defence.gov.au), Ian MacGillivray (Platforms Div., Defence Sci. and Technol. Group, Melbourne, Victoria, Australia), and Nicole Kessissoglou (School of Mech. and Manufacturing Eng., UNSW Sydney, Kensington, New South Wales, Australia)

Acoustic coatings for marine applications are commonly designed using a soft elastic matrix embedded with an array of resonant inclusions. In this work, a simple analytical approach to predict the acoustic performance of a soft elastic matrix embedded with hard inclusions of complex shape is presented. Utilizing analogies from fluid dynamics and electrostatics, a small number of well-known lumped parameters such as the added mass and capacitance are employed to relate the dominant dipole component of scattering of a complex-shaped hard inclusion to that of an equivalent sphere. Effective acoustic properties are then derived using a homogenization. The absorption coefficient of the coating for a range of complex shapes is compared with numerical simulations.

9:20

2aUW4. Predicting the acoustic response of a cylindrical shell immersed in an underwater waveguide using a substructuring technique. Jamie Kha (Ctr. for Audio, Acoust. and Vib., Univ. of Technol. Sydney, 32/34 Lord St., Botany, New South Wales 2019, Australia, jamie.kha@student.uts.edu.au), Florent Dumortier (Laboratoire Vibrations Acoustique, INSA-Lyon, Univ. Lyon, Lyon, France), Mahmoud Karimi (Ctr. for Audio, Acoust. and Vib., Univ. of Technol. Sydney, Sydney, New South Wales, Australia), Laurent Maxit (Laboratoire Vibrations Acoustique, INSA-Lyon, Univ. Lyon, Villeurbanne, France), and Valentin Meyer (Naval Group, Ollioules, France)

Sound radiation from structures in an underwater waveguide presents a challenging underwater acoustics problem typically resolved with numerical methods. This can be computationally inefficient and limited to low-frequency analysis. Hybrid approaches improve on these limitations by combining numerical methods with analytical solutions. This study utilises a substructuring technique called the condensed transfer function approach (CTF) and a decoupling technique called the reverse condensed transfer function approach (rCTF). The hybrid CTF-rCTF approach builds systems by coupling and decoupling simpler subsystems. Each subsystem is represented at the junction interfaces, described by condensed transfer functions (CTFs) that can be analytically or numerically determined. These CTFs are decomposed into orthonormal condensation functions (CFs), for example, gate or exponential functions. To demonstrate the potential of CTF-rCTF, the acoustic response of an infinitely long cylindrical shell under line-distributed load and immersed in an underwater waveguide composed of an upper free surface and lower rigid floor is predicted. This system is built from an underwater waveguide decoupled with a fluid volume and coupled with an excited shell that takes place of the decoupled fluid. Acoustic pressures received in the fluid domain are predicted and verified against an analytical solution. It is particularly shown that using exponential functions as the CFs improves the prediction quality compared to gate functions.

9:40

2aUW5. Internal sound pressure and power radiation from a submerged cylindrical enclosure with interior acoustic excitation. Xia Pan (Platforms Div., Defence Sci. and Technol. Group, 506 Lorimer St., Fishermans Bend, Melbourne, Victoria 3207, Australia, xia.pan@defence.gov.au), Stephen Moore, James Forrest, and Ian MacGillivray (Platforms Div., Defence Sci. and Technol. Group, Melbourne, Victoria, Australia)

This paper presents the modelling and analysis of the effect of an interior acoustic excitation on a submerged cylindrical enclosure. The enclosure has ring stiffeners and acoustic absorption material on the internal wall. The acoustic excitation generates an incident and scattered sound field inside the enclosure. The incident and scattered sound forms the internal pressure which induces a structural response and, thus, the radiated sound. The previous work (James 1985) with an internal noise source located on the vertical axis is expanded to allow forces and interior acoustic sources at arbitrary locations. The formulation of the internal pressure is developed by solving the unknown coupling term between the internal fluid and the hull. The coupling term was neglected by James (1985). However, this term is significant in magnitude for the internal pressure and must, therefore, be included. The analytical results are compared with numerical finite element/boundary element models with good agreement.

10:00–10:20 Break

10:20

2aUW6. On the evaluation of the target strength for comparison between monostatic simulations and measurements. Laurent Fillinger (Acoust. and Sonar, TNO, Oude Waalsdorperweg 63, The Hague 2597AK, Netherlands, laurent.fillinger@tno.nl)

The target strength (TS) quantifies the acoustic return on an active sonar from a submerged object. It is a far-field quantity that varies with frequency and orientation of the object with respect to the sonar source and receiver. Numerical modelling allows to perform simulations in the far-field at an arbitrary frequency. Experimentally, a sonar pulse covers a finite bandwidth, the presence of the water surface and other aspects of the environments complicate the propagation, and the presence of noise and reverberation limit the ability to measure the weakest components of the acoustic return. A numerical investigation is conducted to support the comparison and understanding of the differences between measurements and simulations in monostatic scenario (source and receiver in the same direction). Elastic spheres (for which an analytic solution is available) and an elongated object (simulated with Kirchhoff diffraction approximation) are considered. These objects are simulated in the far-field and in the near-field, at single frequencies and with pulses of several duration and bandwidth. Various TS metrics, including peak TS and integrated TS are evaluated. The comparison of these metrics with the single frequency TS is discussed, as well as the impact of measurement noise and reverberation on these metrics.

10:40

2aUW7. Scattering enhancement of partially buried sphere at the water-sand interfaces due to reflected subsonic Rayleigh waves. Bing Li (School of Naval Architecture, Ocean & Civil Eng., Shanghai Jiao Tong Univ., 800 Dongchuan Rd., Minhang District, Shanghai, Shanghai 200240, China, leebing_w@163.com), Jun Fan, Liwen Tan, Zihao Liu, and Fulin Zhou (School of Naval Architecture, Ocean & Civil Eng., Shanghai Jiao Tong Univ., Shanghai, China)

In view of the development needs of maritime security, the detecting of buried targets over long distances at low frequencies needs to be solved urgently. Through well designed experiments at low frequency (ka = 5–10), this paper presents a study on the scattering echoes from polymethylmethacrylate (PMMA) spheres in partially buried cases (50%, 75% and 100% buried), and the propagation characteristics of subsonic Rayleigh surface waves on the surface of PMMA spheres at the water-sand interface can be clearly observed. It is shown that the fine sand medium with larger impedance brings about smaller circumferential phase and group velocities of subsonic Rayleigh waves on the surface of the PMMA sphere; the resonance peak caused by subsonic Rayleigh waves is shifted toward the low frequency with the deepening of the burial depth, and the offset can be predicted by the circumferential phase velocity of the subsonic Rayleigh wave in different media and burial depth of PMMA sphere, and the experimental and finite element simulation results validate the shift of the Rayleigh resonance peak. The research results of this paper can provide certain reference value for the detection of low frequency buried targets.

11:00

2aUW8. Dynamics and acoustics of bubbly plumes. Alex Skvortsov (Platforms Div., Defence Sci. and Technol. Group, Melbourne, Victoria 3207, Australia, alexei.skvortsov@defence.gov.au), Shuang J. Zhu, Richard Manasseh (Mech. and Product Design Eng., Swinburne Univ. of Technol., Melbourne, Victoria, Australia), Andrew Ooi (Dept. of Mech. Eng., The Univ. of Melbourne, Melbourne, Victoria, Australia), and Martin Kocan (Platforms Div., Defence Sci. and Technol. Group, Port Melbourne, Victoria, Australia)

Transient bubbly plume dynamics including their effects on the free surface were studied numerically using a simple numerical model based on integral theory [1, 2] and Eulerian–Eulerian coupled large Eddy simulation (EE-LES) method. The EE-LES model was validated with the interfacial closures evaluated for drag, lift and virtual mass forces. The simulation data showed good agreement with both numerical predictions and experimental measurements in the literature. The spectrum of the surface pressure fluctuations and acoustical pressure induced by a bubbly plume were extracted from numerical simulations [3]. The effect of interaction of two plumes is discussed. [1] B.R. Morton, G.I. Taylor, J.S. Turner, Turbulent gravitational convection from maintained and instantaneous sources, *Proc. R. Soc. Lond.* **A234**, 1–23 (1956) [2] A. Skvortsov, T. DuBois, M. Jamriska, M. Kocan, Scaling laws for extremely strong thermals, *Phys. Rev. Fluids* **6**, 053501 (2021) [3] S.J. Zhu, A. Ooi, R. Manasseh, A. Skvortsov, Prediction of gas holdup in partially aerated bubble columns using an EE-LES coupled model, *J. Chem. Eng. Science* **217**, 115492 (2020)

11:20

2aUW9. Shear wave effects in hydroacoustic recordings of the Comprehensive Nuclear-Test-Ban Treaty International Monitoring System. Mario Zampolli (Int. Monitoring System Div., Comprehensive Nuclear-Test-Ban Treaty Organization (CTBTO), Wagramerstrasse 5, Vienna 1400, Austria, mario.zampolli@ctbto.org), Georgios Haralabus, Dirk Metz (Int. Monitoring System Div., Comprehensive Nuclear-Test-Ban Treaty Organization (CTBTO), Vienna, Austria), Tiago Castro Alves Oliveira, and Mark K. Prior (Int. Data Ctr. Div., Comprehensive Nuclear-Test-Ban Treaty Organization (CTBTO), Vienna, Austria)

The hydroacoustic (HA) component of the Comprehensive Nuclear-Test-Ban Treaty Organization's International Monitoring System (IMS) comprises six stations with triplets of moored hydrophones suspended in the SOFAR channel and five near-shore seismometer stations for T-phase detection. Understanding the conversion between compressional and elastic waves at the crust-ocean boundary or at large bodies of ice can contribute to better identification and association of arrivals. In this talk, examples recorded by the IMS HA network are presented with a view to stimulating discussion and further analysis: (i) T-phases from the September 2017 announced DPRK underground nuclear test recorded by the hydrophones of HA11 (Wake Island); (ii) a "precursor" southerly arrival at HA04 (Crozet Islands), associated with the hydroacoustic anomaly related to the loss of the Argentine submarine ARA San Juan, which is *inter alia* compatible with a Lamb wave propagating through the Antarctic ice-sheet; (iii) T-phases from earthquakes along the Nazca subduction zone scattered horizontally by the Juan Fernandez sea-mounts and recorded by HA03 (Chile); and (iv) conversion of signals from distant underwater explosions at the crust-ocean boundary near IMS T-stations. Benchmark studies to assess the accuracy of spectral element model setups previously used for scenarios (iv) would be particularly desirable.

11:40

2aUW10. Geoacoustic inversion of wide-angle reflection-coefficient data to estimate sediment shear properties. Stan Dosso (School of Earth and Ocean Sci., Univ. of Victoria, BC V8W 2Y2, Canada, sdosso@uvic.ca), Charles W. Holland (Dept. of Elec. and Comput. Eng., Portland State Univ., Portland, OR), Jan Dettmer (Dept. of Geoscience, Univ. of Calgary, Calgary, AB, Canada), and Yong-Min Jiang (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada)

This paper considers Bayesian geoacoustic inversion of broadband, wide-angle reflection-coefficient data including shear parameters in the seabed model to investigate the ability to estimate these parameters as well as effects on the estimation of other geoacoustic parameters, particularly compressional-wave attenuation. The seabed parameterization is based on the viscous grain-shearing (VGS) sediment-acoustic model, including the grain-to-grain shear modulus as an unknown parameter. VGS sediment parameters are transformed to density and frequency-dependent compressional- and shear-wave speeds and attenuations. Data prediction involves spherical-wave reflection-coefficient calculations. Trans-dimensional inversion, which samples probabilistically over the number of layers in the seabed model, is applied to combine quantitative model selection with parameter/uncertainty estimation. The inversion is applied to reflection-coefficient data sets collected on the New England Mud Patch, and inversion results are compared to those obtained under the common assumption of negligible shear effects (i.e., inverting for a fluid sediment model).

Session 2pAA

Architectural Acoustics and Noise: Airborne and Impact Noise in Buildings II

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

John L. Davy, Cochair
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Wilson Byrick, Cochair
Pliteq Inc., 131 Royal Group Crescent, Vaughan, ON L4H 1X9, Canada

Contributed Papers

1:00

2pAA1. Improvement of sound insulation on the modular housing. Wonhak Lee (Acoust. Environment Ctr., Korea Conformity Labs., 73, Yangcheong 3-gil, Ochang-eup, Cheongju-si, Chungbuk 28115, Republic of Korea, whlee@kcl.re.kr), Soyoung Kim, and Sungjae Han (Acoust. Environment Ctr., Korea Conformity Labs., Cheongju, Chungbuk, Republic of Korea)

The modular construction method is an architectural technique in which buildings are produced in factories and then assembled on construction sites to complete the construction process. This method has two main advantages. First, since the construction takes place in factories, it can reduce construction waste. Second, it can shorten the construction period on the site. So there is considerable interest in using the modular construction method for the construction of residential buildings. In this study, the purpose of the modular construction method in the case of residential buildings is to improve the indoor sound insulation. For this purpose, first, the sound insulation performance of the walls and floor structures has been predicted through simulations at the design stage. Second, the sound insulation performance of the walls and floor structures has been measured using mock-ups in the factory and compared with the simulation results. Third, the sound insulation performance of the walls and floor structures in the actual completed building has been measured and compared with the performance of the simulations and mock-ups. As a result, we have compared the performance of modular construction from the design stage to the completion of the building. Through this process, we have presented procedures for improving the sound insulation performance of the building and quality management, from the design phase to the post-construction stage.

1:20

2pAA2. Experimental study on the sound transmission loss suite at the University of Technology Sydney. Qiaoxi Zhu (Ctr. for Audio, Acoust. and Vib., Univ. of Technol. Sydney, UTS Tech Lab, 32/34 Lord St., Botany NSW 2019, Australia, qiaoxi.zhu@uts.edu.au) and Paul Williams (Ctr. for Audio, Acoust. and Vib., Univ. of Technol. Sydney, Sydney, New South Wales, Australia)

Sound transmission loss suites are essential testing facilities for measuring the sound insulation properties of building elements and assessing noise attenuation. However, inconsistencies in test results can arise due to variations in the size, shape, and construction of test rooms across different laboratories, with biases introduced by the room acoustics or acoustical environment of the facility itself. To evaluate the reliability of such testing, we conducted an experimental study on the sound transmission loss suite at the University of Technology Sydney. Our investigation focused on three key factors: estimating the maximum achievable sound reduction using a heavyweight wall installed at the test aperture on the source room side, testing the effectiveness of vibration isolation between reverberation rooms, and assessing the decoupling of the sound field within the reverberation rooms.

1:40

2pAA3. Enhancing acoustic comfort with window coverings: Reducing sound transmission and reverberation times with a single product. Benjamin Cazzolato (The Univ. of Adelaide, 70 Third Ave., Forestville, South Australia 5035, Australia, benjamin.cazzolato@adelaide.edu.au), Cameron West (Acoust. Blinds and Curtains, Sydney, New South Wales, Australia), Tyler Schembri (The Univ. of Adelaide, Adelaide, South Australia, Australia), Peter Watkins (Acoust. Blinds and Curtains, Sydney, New South Wales, Australia), and Will Robertson (The Univ. of Adelaide, Adelaide, South Australia, Australia)

Window coverings have traditionally been used for design and light control only. However, they also present a great opportunity for a complete acoustic treatment. This is for a number of reasons: (1) They are installed over glazing which represents the typical weak point through which sound enters the space; (2) they have a large surface area for absorption; and (3) they are a natural fit in most modern spaces allowing architects, designers, and clients freedom in their design unconstrained by acoustics. By combining mass loaded vinyl to target sound transmission and absorptive surface linings to provide sound absorption, it is possible to significantly improve acoustic comfort with installation of acoustic curtains. This paper presents the results of extensive laboratory tests on such acoustic curtains, where the diffuse field sound reduction index and sound absorption coefficients were measured for four glazing conditions (open window, 4 mm glass, 6.38 mm glass, and 10.38 mm glass) with 15 different curtain configurations, totalling 76 tests. The results demonstrate significant sound level reductions with the addition of the curtains, with a weighted noise reduction index improvement as high as 13 dB, and a noise reduction coefficient of 0.65 for an additional 2 kg per square metre.

2:00

2pAA4. Quantifying acoustic performance and embodied carbon of acoustic solutions for mass timber buildings: Comparing common approaches around the globe. Aedan Callaghan (Pliteq Inc., 4211 Yonge St., Ste. 400, Ste. 404, Toronto, ON M2P 2A9, Canada, acallaghan@pliteq.com)

As mass timber continues to increase in popularity as a building method to shifting away from high embodied carbon concrete structures to utilize wood as a renewable and lower embodied material. As whole building life

cycle assessments (LCA) become more essential in meeting environmental emissions targets and qualifying for sustainability linked financing, quantifying the embodied carbon of both the structure and all other elements is necessary. The acoustic challenges of mass timber have been well documented and studied in previous works with a range of solutions now commonly utilized. Like most elements of construction, regionality results in different techniques being more popular and practical in different parts of the world. This work will compare the airborne and structure borne sound transmission and associated embodied carbon of four common acoustic designs for mass timber construction: floating concrete toppings, isolated board and batten floor systems, isolated dry floor linings, and raised access floors.

2:20

2pAA5. New draft DEGA guideline103-1 for sound protection classes. Christian Nocke (Akustikbuero Oldenburg, Sophienstr. 7, Oldenburg 26121, Germany, nocke@akustikbuero-oldenburg.de)

The DEGA recommendation 103 (DEGA-Empfehlung–Schallschutz im Wohnungsbau und Schallschutzausweis) has been introduced in the year 2009 and includes a classification scheme for dwellings. Seven classes are defined. The classification is based on different criteria for air borne noise, impact noise and other quantities. In 2018, a revised version has been published. Both version relied on classical quantities such as air born sound insulation R'w and impact sound insulation L'n,w. A new version has recently published as a draft DEGA guideline 103-1 (Entwurf zur DEGA-Richtlinie 103-1 "Schallschutz im Wohnungsbau, Teil 1"). In this new draft, the idea of seven classes hae been taken over. The classification not only is based on traditional values such as R'w and L'n,w but also offers the use of the weighed sound level differences DnT,w and weighed standardizes impact sound pressure level L'nT,w. It is suggested to use the later quantities. This dual track approach allows a better design and will lead to a higher acceptance among acousticians and other users. The new draft is presented and will be discussed in relation to other approaches for sound protection in dwellings.

2:40–3:00 Break

3:00

2pAA6. Optimising acoustic performance, cost efficiency, and embodied carbon in internal partitions: A holistic approach for sustainable building design. Alex Foster (Australian Acoust. Society (VIC), Level 4, 23 Peel St., Adelaide, South Australia 5000, Australia, alex.foster@resonate-consultants.com), Deb James, Even Fung, Bilal Khan, and Deborah Davidson (Adelaide, South Australia, Australia)

The successful design of a building is a complex endeavor that requires careful consideration of multiple factors and disciplines, ultimately aiming for a holistic outcome throughout the entire process—from design and delivery to the lifespan of the building. To achieve this outcome, various aspects, such as architectural design, structural integrity, functionality, environmental sustainability, cost efficiency, and occupant comfort, must be balanced and integrated. This paper explores the relationship between acoustic performance, cost implications, and embodied carbon in internal partitions within the context of building design. It emphasises the importance of a holistic approach in considering these factors collectively and highlights the early integration of acoustic considerations, cost-effective materials and techniques, and environmental sustainability. By making informed decisions, stakeholders can optimize building outcomes by creating superior acoustic environments while minimizing costs and environmental impact. This paper aims to provide a guide for professionals, underlining the significance of a comprehensive approach in achieving optimal project results.

3:20

2pAA7. Floor impact sound reduction and impact absorption performance of floor mat. Gukgon Song (Korea Conformity Labs., Ochang-eup, Cheongwon-gu, 73, Yangcheong 3-gil, Cheongju-si, Chungbuk 28115, Republic of Korea, gsong@kcl.re.kr), Hansol Song, and Yongjin Yoon (Korea Conformity Labs., Cheongju-si, Republic of Korea)

In Korea, the post evaluation system of floor impact sound performance was implemented in August 2022. Therefore, it is necessary to check the floor impact sound performance after the completion of the apartment. If a floor impact sound performance does not meet the regulation limit, compensationm and follow-up renovations are required. In this study, in order to develop a floor mat that can reduce heavy weight floor impact sound, the floor impact sound reduction performance and impact absorption performance of each condition of the sample were confirmed. Floor impact sound reduction performance was measured using the test methods of KS F 2865 and ISO 10140-3, and impact absorption performance was measured according to the test method of JIS A 6519. As a result of the measurement, the performance of reducing floor impact sound showed a clear difference in performance according to thickness rather than material type. Impact absorption performance was found to be affected by surface hardness and material thickness. In conclusion, it is considered that the selection, thickness, and surface hardness of materials are important design factors when developing floor mats.

3:40

2pAA8. A study on the evaluation of impact sound reduction performance of floor mats for reducing floor impact sound in apartment buildings. Junoh Yeon (Acoustic&Shock, KOMERI (Korea Marine Equipment Res. Institute), 24-20, Noksansandan 335-ro, Gangseo-gu, Busan 46754, Republic of Korea, joyeon@komeri.re.kr), Himo Goo, and Soonseong Moon (Acoustic&Shock, KOMERI (Korea Marine Equipment Res. Institute), Busan, Republic of Korea)

This is a study on the performance evaluation of floor mats used to reduce floor impact sound in apartment houses in South of Korea. A total of 8 types of floor mats and 5 or more test rooms with different floor structures were used to measure and evaluate impact sound reduction performance. The impact source is a Bang machine (heavy-weight impact sound 1), a rubber ball (drop of height 100 cm_{Heavy}-weight impact sound 2) tapping machine (light-weight impact sound), and a rubber ball drop height of 40 cm that simulates the actual impact force. As a result of the test, the impact sound reduction performance for the light-weight impact sound was evaluated as excellent. It was evaluated at the level of ambient background noise. Also, the reduction performance of the bang machine was somewhat different depending on the floor structure. In addition, there are cases where it is rather amplified in the low frequency band (63 Hz). In the case of rubber ball (drop of height 100 cm), the reduction performance was different depending on the floor structure. However, the impact sound reduction performance was evaluated to be sufficiently reduced. The impact noise reduction performance at the rubber ball drop height of 40 cm was found to be slightly higher than that at the drop height of 100 cm.

4:00

2pAA9. Comparative analysis of the heavy-weight floor impact noises using various shapes of EVA resilient materials. Jakin LEE (Architectural Eng., Chungbuk National Univ., Seowon-gu chungdea-ro, Chungju, Chungcheongbuk-do, Chungju 28644, Republic of Korea, rnjsdhsua@naver.com) and Chan-Hoon Haan (Architectural Eng., Chungbuk National Univ., Cheongju, Republic of Korea)

Social issues concerning the floor impact noises are increasing as apartments are being increased in Korea. The present study aims to improve the floor impact noises using various shapes of EVA resilient materials in composite floor structures. In order to this, three different shapes of EVA resilient materials were used including flat plate type, deck plate type, and cavity type of EVA. boards. All the composite structure consisted of PP panel, PET sound absorption materials and EVA resilient materials with a total depth of 40mm. Also, each shape of EVA has two different types according to the use of foothold of EVA materials. For the mount type of structure, the used numbers of EVA foothold was 25, 56, and 41 per 1 m2 of area. All the floor impact noise measurements were done at the recognized testing authority using impact ball as the standard heavy-weight floor impact sound source. The values of L'i, Fmax, and AW were deduced from every measurements. As a result, regardless of the EVA shapes, better performances were drawn when EVA footholds were used, i. e., mount type. It was shown that the average L'i, Fmax, and AW value of mount types was 41–42 dB while average value of 47–48 dB for deck plate type. 50–55 dB for flat plate and cavity type were measured. It can be recognized that the good performance of mount type EVA materials is caused by the air layer gap around the EVA footholds as well as the low dynamic elastic modulus of EVA.

4:20

2pAA10. Round-robin tests of the different types of sensors for floor impact sounds and vibrations. Jakin LEE (Architectural Eng., Chungbuk National Univ., Seowon-gu chungdea-ro, Chungju, Chungcheongbuk-do, Chungju 28644, Republic of Korea, rnjsdhsua@naver.com) and Chan-Hoon Haan (Architectural Eng., Chungbuk National Univ., Cheongju, Republic of Korea)

Recently, floor impact noise monitoring system was introduced for residents to control making noises spontaneously using sound and vibration sensors. The present study aims to investigate the performance and characteristics of various current using sound and vibration sensors which can be eventually compatible with the floor impact noise monitoring system. For measurements of sound sensors, 7 sound microphones were used including 2 condenser types, 4 dynamic types, and 1 electro condenser type. Also, 6 vibration meters were used including 2 MEMS types, 2 Piezo types, and 2 spring types. Sound levels were measured in a semi-anechoic chamber using white noise in the frequency range from 63 Hz to 12.5 kHz. Vibrations were measured using high density wooden panel with an impact ball in the frequency range from 16 Hz to 4 KHz. Round-robin tests were done to confirm the exact performances of all the sensors under same condition. As a result, it was found that dynamic type microphones show the most similar response characterization in comparison with the reference condenser microphone especially in the air-borne sound frequency range from 250 Hz to 2 KHz. Also, it was found that spring type vibration meters show the most appropriate characterization at 63 Hz, most critical structure-borne sound frequency. Thus, the results drawn from the round-robin tests could be used for floor impact noise monitoring systems with the benefit of both economy and performance.

4:40

2pAA11. Changes in heavy-weight impact sound isolation by construction stage in new residential buildings. Changyeon Yun (RD Ctr., DOOSAN, Doosan Bldg., 726, Eonju-ro, Doosan E&c, Seoul, Gangnam 06057, Republic of Korea, cyyun@doosanenc.com), Jungho Kim, Eeuichul Hwang, and Heejin Kim (RD Ctr., DOOSAN, Seoul, Republic of Korea)

In Korea, 63% of the population lives in residential buildings. Due to the prevalence of residential buildings, there are constant disputes and complaints related to floor impact sound. To address the floor impact sound problems, the Ministry of Land, Infrastructure, and Transport (MOLIT) announced strong legal regulations from 2022. The law holds construction companies liable for compensation if they fail to meet the minimum sound insulation standard. The existing construction methods for floor structures were unable to meet the required minimum sound insulation standards, leading to the need for improvements. The evaluation process was enhanced by changing the source from bang-machine to impact ball, leading to more accurate assessments of floor impact sound insulation. In response to the challenges faced with existing methods, the research paper developed and applied a composite floor structure using a new material with increased area density. The performance of each construction stage in the new composite floor structure was measured and evaluated according to the ISO 16283-2 (testing) and ISO 717-2 (evaluation) standards. The implementation of the new composite floor structure with increased area density resulted in a noise reduction of up to 10 compared to the bare slab. [Work supported by the Industrial Technology Innovation Program (20023556, Development of personalized noise mitigation technology and service based on noise simulation to solve the noise problem in personal space) funded By MOTIE, Korea.]

5:00

2pAA12. Further development and testing of a rainfall test rig. Michael D. Latimer (176 Hazeldean Rd., Addington, Christchurch, Canterbury 8024, New Zealand, mike@acoustic-testing.co.nz) and Sebastian Yeoman (Christchurch, Canterbury, New Zealand)

Changes made recently by The Ministry of Health New Zealand, to the internal noise design requirements for Preschools, has required that the design of the roof reduces rainfall noise to Ministry of Education, Design Quality Learning Spaces Acoustics (DQLS) Version 3.0, for rainfall noise. Previously Canterbury Acoustic Testing Services (CATS) had developed a rainfall noise test rig, in accordance with ISO 10140-1: 2016 [3] and ISO 10140-5:2010/Amd 1:2014 and carried out testing on a series of warm roof/ceiling constructions. Preschools tend to use cold roof systems where the thermal insulation sits on the ceiling, rather than in a warm roof, where the insulation is part of the roof structure. So there has been no tested performance data for rainfall noise acoustic design purpose, using a cold roof construction. As part of the testing program, there was a need to recommission the rig, to verify rain drop size and rainfall rate. A testing program was established, to include the base roof structure (roofing iron only) to a full buildup of roof and ceiling, to get a better understanding of how each component in the system contributes to the control of the rainfall noise.

Session 2pAB

Animal Bioacoustics and Underwater Acoustics: Session in Honor of Douglas H. Cato II

John Buck, Cochair
ECE, Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., Dartmouth, MA 02747

Michael Noad, Cochair
School of Veterinary Science, The Univ. of Queensland, Gatton 4343, Australia

Chair's Introduction—12:55

Invited Papers

1:00

2pAB1. Dr Doug Cato's contributions to the understanding, measurement, and characterization of Australia's ambient acoustic sea noise environment. Brian G. Ferguson (DSTG, 13 Garden St., Eveleigh, New South Wales 2015, Australia, brian.ferguson@defence.gov.au)

In 1966, Dr Douglas H. Cato began his investigation of ambient noise in Australian waters as part of the Royal Australian Navy Research Laboratory's Project BARRABOOL. His research involved identifying and understanding the various ambient noise mechanisms that limit the performance of operational sonar systems in Australian waters. He is responsible for the "Cato Curves" of ocean noise, which have been used extensively for predicting sonar performance. Similarly, his understanding of the ocean environment and the spatial and temporal variability of ambient undersea noise is exploited by the Royal Australian Navy to ensure effective sonar and submarine performance. Dr Cato is a Fellow of the Acoustical Society of America and his research achievements are published in international peer-reviewed journals including the Journal of the Acoustical Society of America and as book chapters in reference texts in underwater acoustics. In 2012, he was the recipient of the Minister's Achievement Award in Defence Science for his outstanding contributions in underwater acoustics and the effect of sound on marine mammals. This presentation provides examples of Doug's achievements in defence science and their significance to the Commonwealth of Australia, Department of Defence.

1:20

2pAB2. A review of sound radiated by breaking ocean waves. Grant B. Deane (Marine Physical Lab., Scripps Inst. of Oceanogr., UC San Diego, UC San Diego, 9500 Gilman Dr. #0206, La Jolla, CA 92093-0206, gdeane@ucsd.edu)

"Physical Mechanisms of Noise Generation by Breaking Waves—A Laboratory Study" was published over 35 years ago by Mike Banner and Doug Cato. This seminal work describes a laboratory experiment demonstrating that the dominant source of breaking wave noise above a few hundred Hz is associated with the formation of bubbles and coalescing or splitting bubbles. This observation opened the door to explaining the properties of ambient sound under wind-driven seas and remotely monitoring air entrainment by breaking waves. Progress on these topics since Banner and Catos' experiment will be reviewed and outstanding issues discussed. [Work supported by the US Office of Naval Research Ocean Acoustics program (Grant No. N00014-21-1-2316).]

1:40

2pAB3. Measurements of a 20, 440, and 3130 cubic inch air gun or array off Peregian Beach Queensland and Dongara Western Australia highlight small and large scale inhomogeneous sound propagation environments. Robert McCauley (Ctr. Marine Sci. and Technol., Curtin Univ., P.O. U1987, Perth, Western Australia 6854, Australia, R.McCauley@curtin.edu.au), Douglas H. Cato (School of Geosciences, Univ. of Sydney, Sydney, New South Wales, Australia), Rebecca Dunlop, and Michael Noad (The Univ. of Queensland, Gatton, Queensland, Australia)

During the project "Behavioural Response of Australian Humpback whales to Seismic Surveys" three air gun configurations were used to quantify the response of southerly migrating humpback whales to a northerly travelling air gun. Off Peregian Beach, Queensland (26.5^o S latitude) a 20 cubic inch (cui) single air gun, 440 cui and 3130 cui arrays were operated while off Dongara, Western Australia (29.5^o S) the 440 cui array was operated. At a first glance, we assumed within the respective $\sim$200 km^2 experimental areas sound propagation was reasonably uniform but this was not the case. At Peregian Beach, flat supposedly sandy areas had patchy outcrops of soft "coffee" rock which dramatically increased sound propagation loss while in the north a large arc of hard reef or shallow sand over reef again increased sound propagation loss and acted as an "amphitheatre." Off Dongara different depths of sand or no sand over a limestone base created variable sound propagation. When comparing Dongara and Peregian Beach the shallow limestone off Dongara increased propagation loss compared to Peregian. This talk will explore some of the idiosyncrasies of air gun propagation and inhomogeneities which can be expected when planning sound exposure experiments.

2:00

2pAB4. Estimation of seafloor properties in shallow oceans using minimally processed vertical directivity of surface-generated ambient noise. Adrian D. Jones (Ocean Acoust. Assoc., P.O. Box 333, Edinburgh, South Australia 5111, Australia, bearjones3@bigpond.com)

Typically, data from an array with vertical aperture will be beam-formed to provide the level and variation of spectrally averaged ocean ambient noise as a function of vertical angle. This paper considers the potential for such measurements to provide information about the acoustical properties and layering of a seafloor in a shallow ocean. First, with the assumption of dipole surface sources, it is feasible to describe the ambient noise in the near-horizontal in terms of the Weston α parameter (Weston, J. Sound Vib. **18**, 271–287, 1971). Furthermore, measurements of ambient noise in the positive and negative vertical directions may be used to provide a spectrally averaged determination of bottom loss. With the presence of a surficial seafloor layer, each of the near-horizontal and near-vertical noise intensities will exhibit spectral variation. Techniques are described by which such relative variations in noise directivity, over a frequency span of some octaves, enable a reasonable estimation of, first, the thickness of a seafloor layer, and second, the acoustical properties of the layer and the seafloor basement.

2:20

2pAB5. Laboratory measurements of thermal noise. Mark Readhead (14 Emu St., Strathfield, New South Wales 2135, Australia, mark.readhead@gmail.com)

An experimental study was undertaken of the aspect of underwater ambient noise which normally receives only a cursory reference—namely, thermal noise. A brief survey of the theory and literature, and consideration of how to separate amplifier noise from hydrophone noise, is followed by laboratory measurements on piston and spherical hydrophones. Results are presented of impedance and power spectral density measurements undertaken in water and other fluids. Measurements with the piston hydrophones were also performed as a function of distance from tank and water surface boundaries or from another hydrophone. The cross power spectral densities are presented as a function of separation distance between pairs of hydrophones. All hydrophones had their free field receive sensitivities and beam patterns measured, but the spectral densities have not yet been converted to pressures as the more interesting features are seen near resonances, where phase calibrations will also be required.

2:40–3:00 Break

3:00

2pAB6. Evidence of oceanic and atmospheric tides in underwater ambient noise time records. Anthony I. Eller (Appl. Ocean Sci., Springfield, VA), Kevin D. Heaney (Appl. Ocean Sci., 5242 Port Royal Rd. #1032, Springfield, VA 22151, kevin.heaney@appliedoceansciences.com), and David L. Bradley (Dept. of Defense, Alexandria, VA)

Multiyear time records of low-frequency ocean ambient noise show clear evidence of a modulating influence by major semi-diurnal and diurnal gravitational tides. These are displayed by means of a variance spectral analysis of noise time records obtained from the United Nations Comprehensive Nuclear Test Ban Treaty Organization for hydrophone stations at Wake Island, Ascension Island and Diego Garcia. Narrow-band noise frequencies ranged from $\frac{1}{2}$ to 10 Hz. There is also evidence in the noise records of a diurnal atmospheric tide related to solar radiation and atmospheric loading. This same tidal presence is seen in wind speed records as well. Part of the challenge in the analysis is to distinguish between expected diurnal behavior in wind speed, with its accompanying effect on ambient noise, and the presence of actual tidal lines. A brief history of earlier observations of diurnal variation in underwater acoustics is presented, along with some suggestions of causal mechanisms.

3:20

2pAB7. Vector acoustic observations of underwater noise energetics over a 24 h period in Puget Sound, Washington, as influenced by ferry and ship traffic. Peter H. Dahl (Appl. Phys. Lab. and Mech. Eng., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, dahl@apl.washington.edu) and David R. Dall'Osto (Acoust., Appl. Phys. Lab. at Univ. of Washington, Seattle, WA)

Vector acoustic measurements of underwater noise made continually over a 24-h period within a small bay on the west side of Puget Sound, Washington State are discussed. The bay contains the Washington State ferry landing at Kingston with measurement site approximately 1 km distant from the landing, at water depth 42 m. Periodic, nearby ferry traffic, and episodic large-vessel shipping activity punctuate the observations which will be compared with those made in deeper waters. Potential and kinetic energies in key shipping decidecimal bands are equal within calibration uncertainty; a practical result towards the inference of kinematic properties from pressure-only measurements within inland waters complementing similar results obtained in off shore waters. An important exception exists when the noise field is dominated by a vessel passing directly over the sensor, as was demonstrated by the deployment vessel moving at speed 7.5 knots. Observed near the seafloor, broadband noise emissions from a vessel passing directly above exhibit frequency bands where potential acoustic energy is greater than kinetic energy (and vice versa in neighboring frequency bands). This phenomenon is also examined with a model based on multiple interfering reflections from the seabed, sea surface, and buried sediment layers.

3:40

2pAB8. Large-aperture wide-bandwidth densely populated coherent hydrophone array system for ocean acoustic monitoring. Max K. Radermacher, Matthew E. Schinault, Anthony Britton, Sai Geetha Seri (Elec. and Comput. Eng., Northeastern Univ., Boston, MA), Hamed Mohebbi-Kalkhoran (Mech. Eng., Massachusetts Inst. of Technol., 77 Massachusetts Ave., Cambridge, MA 02139, hmohebbi@mit.edu), Nicholas C. Makris (Mech. Eng., Massachusetts Inst. of Technol., Cambridge, MA), and Purnima Ratilal (Elec. and Comput. Eng., Northeastern Univ., Boston, MA)

Underwater acoustic monitoring with a large-aperture coherent hydrophone array is advantageous because it enhances signal-to-noise ratio of received signals, provides estimates of signal bearing, and enhances signal detection ranges by one to two orders of magnitude over that of a single hydrophone. A large aperture coherent hydrophone array system comprising >160 elements has been developed inhouse at Northeastern University. The overall acoustic aperture length is 192 m with array elements nested into multiple uniformly spaced or log spaced subapertures. Hydrophones with integrated broadband pre-amplifiers designed with a linear frequency response from 10 Hz to 50 kHz send differential pair amplified and filtered analog signals to multiple 24-bit, 32-channel analog-to-digital converters with sampling rate that is programmable up to 100 kHz per channel. Array internals are designed using field replaceable pressure tolerant components verified by pressure chamber testing. Forward and aft modules are equipped with non-acoustic sensor elements to provide depth, heading, pitch, roll and temperature measurements. Acoustic aperture telemetry is user datagram protocol (UDP) converted to single-mode fiber for transmission along 600 m of faired tow cable to a shipboard data acquisition system. Examples of passive acoustic data from array deployment in the US Northeast coast are presented illustrating array capabilities.

4:00

2pAB9. Passive ocean acoustic waveguide remote sensing of vocalization behavior and spatial distribution of diverse marine mammal species in the Norwegian and Barents Sea. Hamed Mohebbi-Kalkhoran (Mech. Eng., Massachusetts Inst. of Technol., 77 Massachusetts Ave. Cambridge, MA 02139, hmohebbi@mit.edu), Shourav Pednekar, Chenyang Zhu (Mech. Eng., Massachusetts Inst. of Technol., Cambridge, MA), Heidi Ahonen (The Norwegian Polar Inst., Tromsø, Norway), Olav Rune Godø (Inst. of Marine Res., Bergen, Norway), Purnima Ratilal (Elec. and Comput. Eng., Northeastern Univ., Boston, MA), and Nicholas C. Makris (Mech. Eng., Massachusetts Inst. of Technol., Cambridge, MA)

Leveraging data acquired using a 160-element coherent hydrophone array deployed in the Norwegian and Barents Seas during spring 2014, and the passive ocean acoustic waveguide remote sensing (POAWRS) technique is employed to enable instantaneous wide-area monitoring of marine mammal vocalizations over expanses exceeding 100 km in diameter. The vocalization behavior of diverse marine mammal species including Fin, Humpback, Minke, Sperm, and Beluga whales are analyzed, quantifying time-frequency characteristics and call patterns from their vocalization signals present in high-resolution beamformed power spectrograms. Previously developed automatic detection and machine learning algorithms are employed for clustering and classification of underwater acoustic events, facilitating the subsequent manual audio-visual inspection of whale calls. Bearing-time trajectories of repetitive species-specific vocalizations signals are utilized to estimate locations of the whale calls and hence their temporo-spatial distributions. Through comparative studies, we assess the presence and abundance of marine mammal species-specific vocalization signals in three distinct coastal regions off Norway, namely, Alesund, Lofoten, and Northern Finmark. In addition, we investigate correspondences of marine mammal distributions with concurrently imaged instantaneous wide-area fish distributions from distinct fish species to elucidate potential predator-prey interactions and habitat preferences.

4:20

2pAB10. Wind noise source level and Bio-Goose: Perspectives on Doug Cato's contributions in ocean ambient noise. Ross Chapman (School of Earth and Ocean Sci., Univ. of Victoria, 3800 Finnerty Rd., Victoria, BC V8W3V6, Canada, chapman@uvic.ca)

Doug Cato's research has provided insights that form the basis of our understanding of ambient noise in the ocean. This paper focuses on only two aspects of ambient noise that are strongly linked to Doug's work. One is source level of ambient sound due to local wind at the sea surface. Estimates of wind sound source level derived from noise measurements with vertical line hydrophone arrays are shown to be consistent with the levels for noise due to wind reported in the Cato curves—up to date relationships for levels of components of ambient noise over a wide frequency band. The second aspect revisits a curious sound that was recorded in waters around Australia and New Zealand some years ago. The sounds were believed to be generated by marine animals, but the type of creature was not identified at sea or in subsequent analysis. Owing to the limited bandwidth of the recordings, the sound was accordingly dubbed as Bio-Duck in New Zealand and Bio-Goose in Australia. Further characteristics of the sound are presented, along with evidence of a probable conversation between two speakers and an example of the Lombard effect in the communications.

Session 2pAO

Acoustical Oceanography: Topics in Acoustical Oceanography

Andone C. Lavery, Cochair
Applied Ocean Physics and Engineering, Woods Hole Oceanographic Inst., 98 Water St., Woods Hole, MA 02540

David Barclay, Cochair
Dept. of Oceanography, Dalhousie Univ., 1355 Oxford St., PO Box 15000, Halifax B3H 4R2, Canada

Contributed Papers

1:00

2pAO1. Forward looking sonar applications for unmanned vehicles. Peter D. Romaine (Acoust., NSWC, 110 Vernon Ave. Panama City, FL 32408, peter.romaine@navy.mil)

Unmanned vehicles have the potential to greatly expand our ability to observe dynamic ocean environments. Whereas distributed acoustic sensors can monitor discrete measurement points in time and space, autonomous maritime platforms provide increased flexibility. Acoustic principles are a key component of this functionality and forward looking sonar (FLS) systems can acquire acoustically derived data products while enabling better informed autonomy behaviors based on enhanced situational and environmental awareness. This paper explores applications of forward look sonar technology when integrated with underwater vehicles. Utility of both interferometric and two-dimensional acoustic data products are examined to illustrate the impact on technologies associated with navigational drift mitigation and ocean observation. Results from sonar platform integration and in-water acquisition for development of an underwater terrain-aided navigation (UTAN) particle filter, an acoustic enabled simultaneous localization and mapping (SLAM) algorithm, obstacle avoidance techniques, and seabed observations are presented.

1:20

2pAO2. Noise levels in a changing Arctic Ocean and its implications for security. Giacomo Giorli (NATO STO CMRE, Viale S. Bartolomeo, 400, La Spezia 19126, Italy, Giacomo.Giorli@cmre.nato.int), Aniello Russo, and Sandro Carniel (NATO STO CMRE, La Spezia, Italy)

Arctic Ocean is undergoing an “Atlantification” of its oceanographic properties. Sea ice retreat and reduction of sea ice age will affect its underwater soundscape, with anthropogenic noise expected to increase due to the exploitation of new maritime routes. The CMRE’s Environmental and Operational Effectiveness Programme conducted a study of the new Arctic oceanographic conditions and ambient noise by deploying in 2021 and in 2022 different moorings equipped with passive acoustic recorders and oceanographic sensors. Data did not show a clear relation between sea-ice concentration and noise levels in the marginal ice zone. However, noise levels indicate that the possible presence of transmissions ducts might increase high frequency noise in the subsurface area. Seasonality and variations in the overall soundscapes were also evident by the presence of sounds from biological sources. In 2023, with the support of the NATO Office of the Chief Scientist, CMRE started a long-term scientific endeavor to study how climate change might affect the Alliance’s security in the maritime domain. A significant research effort is focused on the Arctic, and in June–July 2023 CMRE deployed three deep moorings for monitoring the acoustic-oceanographic conditions in the long term. [Work supported by the NATO Allied Command Transformation.]

1:40

2pAO3. Marine and lacustrine ice fracture detection. John A. Case (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, jac7175@psu.edu) and Andrew Barnard (Graduate Program in Acoust., Penn State Univ., University Park, PA)

Ice fracturing has been extensively studied and modeled. With increased interest in ice mechanics and fracturing in recent years in climate science, fisheries, and for cultural impacts, detecting and classifying fracturing events has become an important problem to consider. Fractures primarily occur due to stress relief within an ice sheet during temperature shifts and ice movement. These events create mechanical waves within the sheet that couple into the water column which can then be detected as pressure and particle velocity fluctuations. Machine learning algorithms will be used to detect and classify ice cracking events though their acoustic signature. Different models will be compared to one another for effectiveness and accuracy. Data will be shown from several different locations, including Northern Alaska and the Great Lakes.

2:00

2pAO4. Estimation of surface layer and Pacific summer water properties from acoustic transmissions in the Beaufort duct using a tomographic array during 2016–2017. John A. Colosi (Oceanogr., Naval Postgrad. School, 833 Dyer Rd., Monterey, CA 93943, jacolosi@nps.edu), Heriberto J. Vazquez, Bruce Cornuelle, Peter F. Worcester, and Matthew A. Dzieciuch (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA)

The 2016–2017 Canada Basin Acoustic Propagation Experiment (CANAPE) was conducted to assess the effects of the changing Beaufort Gyre on low-frequency underwater acoustic propagation and ambient sound. A 150-km radius ocean acoustic tomography array was deployed with six transceivers and a distributed vertical line array (DVLA) measuring the impulse responses every four hours with broadband signals centered from 172.5 to 275 Hz. The nominal transceiver source depth was 175-m, placing them near the Beaufort duct axis, and the 60 hydrophone DVLA spanned 50 to 600 m. The Beaufort duct (approximately 90-m to 240-m depth) and the surface layer (approximately 0 to 90-m depth) form a coupled double-duct system. Observed arrivals in this system show reverse dispersion with the lowest Beaufort duct modes arriving first and higher double duct modes making up a transmission finale. In this talk, we investigate the oceanographic information content contained in the first and last arrivals which are the easiest to detect and track. The first arrival shows fluctuations from eddies, tides/inertial oscillations, and small seasonal heating/cooling. The last arrival shows a strong seasonal heating/cooling signal but is un-trackable during periods of significant ice cover due to enhanced transmission loss.

2:20

2pAO5. Ocean ambient noise field modelling and the optimized noise term. Nikita Kovaloff (Meteorology/Oceanogr. (METOC), Royal Canadian Navy, 2-70, Collins Grove, Dartmouth, NS B2W 4E6, Canada, s23808@gmail.com)

The objective of this thesis is to determine the frequency and wind-wave forcing dependent effective sea surface noise source level per unit area extracted from the hourly minimum sound power levels of six month-long acoustic recordings. The effect of the propagation environment is accounted for using Bellhop. The simulated environment is configured using climatological sound velocity profiles to capture seasonal effects and bottom sound speed estimates made from seabed sediment maps. Hourly meteorological data were extracted from ERA5 providing relevant wind and wave parameters from which noise levels may be predicted. A weighted composite model consisting of neutral wind and significant wave height leveraging the two-term exponential regression function proved to maximize model R2. Received level data originating from 16 hydrophone stations in the North Atlantic and Labrador Sea were combined with the Bellhop TL simulations in order to produce estimates of the effective noise source level per unit area (NSL/A) for changing surface environmental conditions and inter-compared. Hourly minimum sound power level derived model-data comparisons using horizontal wind speed magnitude 10 m above sea level expressed a decrease in NSL/A estimates versus Kewley (1990) by 10 to 15 dB from 1 to 3 kHz.

2:40

2pAO6. A study on the prediction model of water column sound absorption coefficients considering the characteristics of the East Sea of South Korea. Seung Uk IM (Faculty of Earth and Marine Convergence - Ocean System, Jeju National Univ., 102 Jejudaehak-ro, Jeju-si 63243, Republic of Korea, sklim1013@stu.jejunu.ac.kr), Cheong Ah Lee (Faculty of Earth and Marine Convergence - Ocean System, Jeju National Univ., Jeju, Republic of Korea), Changsoo Kim, and Dong-guk Paeng (Ocean System Eng., Jeju National Univ., Jeju-si, Republic of Korea)

East Sea of Korea has the deepest water depth among Korean seas, and the water column changes are distinct in space and time, resulting in a wide range of water column sound absorption coefficients. In this study, we calculate and analysis the spatiotemporal variation of water column sound absorption coefficient in the East Sea of Korea, which has unique oceanographic characteristics. Using extensive CTD data from the Korea Oceanographic Data Centre (KODC) from 1991 to 2021, we calculate the sound absorption coefficient using the widely applied Francois–Garrison model and reveal the unique absorption characteristics of the East Sea. The derived coefficients are then used to calibrate and improve past multibeam echosounder system (MBES) data, which suffered from incomplete water column information and arbitrary mean values. To facilitate the calculation and development of sound absorption coefficient, a MATLAB-based graphical user interface (GUI) was developed. The results of this study contribute to the calculation and inference of sound absorption coefficients along spatial and temporal dimensions of the water column in the East Sea and to the extraction of more accurate seafloor information from MBES data. This study developed the way for future research focusing on algorithmic advances and new methodologies for the calculation of sound absorption coefficients in the West and South Seas of Korea.

3:00–3:20 Break

3:20

2pAO7. Frequency dependent effects of environmental parameters on the broadband acoustic wave propagation in shallow water waveguides. Mohsen Badiey (Elec. and Comput. Eng., Univ. of Delaware, 139 The Green, Newark, DE 19716, badiey@udel.edu), Lin Wan, and Christian D. Escobar-Amado (Elec. and Comput. Eng., Univ. of Delaware, Newark, DE)

Acoustic wave intensity plays a crucial role in underwater acoustics, particularly in shallow water regions in which the signal suffers substantial energy fluctuations. The physical properties of the environment significantly impact the intensity of broadband acoustic waves in shallow water waveguides. The accuracy of the intensity is vital for the reliability of any inverse algorithm used to obtain the physical properties of the waveguide boundaries. In this paper, we explore the effects of sound speed fluctuations in the water column on broadband acoustic propagation across different frequencies. Our observations reveal that at higher frequencies the sound intensity is more sensitive to oceanographic variability. This poses challenges for geo-acoustic inversions when parameters in the water column are not well-known, particularly at higher frequencies. To validate our findings, we present simulations using normal modes and parabolic equation with examples from recent experimental observations conducted on the New England Mud Patch Area and the continental shelf region of the United States during 2017, 2021, and 2022. These experiments provide valuable insight into the intricate behavior of acoustic waves in shallow water environments at different frequency bands. [Work supported by ONR (Grant No. N00014-21-1-2760).]

3:40

2pAO8. Acoustic energy reflections from deep seabed layers in seismic reflection surveys. Alexander S. Douglass (Oceanogr., Univ. of Washington, 1501 NE Boat St., MSB 206, Seattle, WA 98195, asd21@uw.edu), Warren T. Wood, Benjamin J. Phrampus (Ocean Sci. Div., Naval Res. Labs., Hancock County, MS), and Shima Abadi (Univ. of Washington, Seattle, WA)

Data from marine seismic reflection surveys are generally used to study the composition of the seabed 100s to 1000s of meters below the seafloor. These surveys consist of an airgun array that broadcasts an impulsive signal towards the seafloor, the reflections of which are measured by towed hydrophone streamers and used to invert for geophysical properties. The abundance of data from these surveys provides many opportunities to conduct acoustic analyses and to utilize data-driven approaches. Here, the extensive datasets are used for analyzing the impacts of seabed characteristics on the acoustic fields in the water column. Various seabed components are considered and their influence on the acoustic field quantified. It is shown that layers well below the seafloor can contribute up to 50% of the acoustic energy reflected into the water column at frequencies below 100 Hz and ranges approximately three times the channel depth. Additionally, non-intuitive relationships between layer depths and acoustic energy are explored. [Work supported by ONR.]

4:00

2pAO9. A general model for sediment-column structure on the New England Mud Patch from Bayesian geoacoustic inversion of seabed reflection data. Yong-Min Jiang (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC V8W 2Y2, Canada, minj@uvic.ca), Charles W. Holland (Dept. of Elec. and Comput. Eng., Portland State Univ., Portland, OR), Stan Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada), and Jan Dettmer (Dept. of Geoscience, Univ. of Calgary, Calgary, AB, Canada)

Muddy sediments cover significant portions of continental shelves, but their physical properties remain poorly understood compared to sandy sediments. To explore the spatial and frequency dependencies of mud properties, wide-angle seabed reflection coefficients versus grazing angle and frequency were measured on the New England Mud Patch (NEMP) during the 2017 Seabed Characterization Experiment. This paper presents trans-dimensional Bayesian inversion of reflection coefficients within a frequency band of 1–3 kHz and an angular range of $\sim$15–25° to obtain geoacoustic profiles and associated uncertainties, as well as frequency dependencies of sound speed and attenuation. The estimated geoacoustic profiles are similar to those from previous inversions of reflection-coefficient data at lower frequencies (0.4–1.3 kHz) collected at two different sites on the NEMP. Based on the inversion results at all three sites, a general interpretive model for sediment-column structure and variability is synthesized for the NEMP. This model includes an upper mud layer in which sediment properties change slightly with depth due to near-surface processes, an intermediate mud layer with uniform properties, and a transition layer where properties change rapidly with depth due to increasing sand content in the mud above a sand layer. [Work supported by the Office of Naval Research.]

4:20

2pAO10. Sparse underwater acoustics signal reconstruction on shallow water and its performance evaluation. Dhany Arifianto (Eng. Phys., Institut Teknologi Sepuluh Nopember, Surabaya, Indonesia) and Aprianto D. Prasetyo (Eng. Phys., Institut Teknologi Sepuluh Nopember, Dept. of Eng. Phys., Institut Teknologi Sepuluh Nopember, Surabaya 60111, Indonesia, arkiven4@gmail.com)

We performed a simple underwater acoustics signal transmission on an experimental pond and shallow water in Surabaya harbor. First, we tested the sparse reconstruction algorithm on the underwater signal transmitted in the experimental pond to evaluate the sparsity limit of reconstructed signal. The transmitted signal consisted of a train of sinusoidal and pure tone with variation of frequency and duration. From the results, it is concluded that the proposed algorithm performed well in our experimental pond. However, it performed poorly with respect to distance on a shallow water environment.

4:40

2pAO11. Assessing the feasibility of conducting coastal acoustic tomography using BELLHOP modeling. Wan-Gu Kim (Korea Inst. of Ocean Sci. & Technol., 385, Haeyang-ro, Yeongdo-gu, Busan 49111, Republic of Korea, kimwangu@kiost.ac.kr), Byoung-Nam Kim, Bok Kyoung Choi (Korea Inst. of Ocean Sci. & Technol., Busan, Republic of Korea), Jungyong Park (Agency for Defence, Changwon, Republic of Korea), Ho Youn Ji, Min-Seok Choi (Korea Inst. of Ocean Sci. & Technol., Busan, Republic of Korea), and Seng Hyeon Park (Underwater Survey Technol. 21, Inc., Inchon, Republic of Korea)

In order to apply coastal acoustic tomography (CAT) efficiently to a specific coastal area, it is necessary to design a measurement system in consideration of various variables related to the marine environment of the area. In particular, to consider seasonal variations in the water temperature structure quantitatively is important regarding the formation of sound rays. In this study, the feasibility of performing CAT according to the seasonal variations in the water temperature structure of Yeosu and Gwangyang Port was studied through simulation using BELLHOP model. First, BELLHOP model simulations were performed using the precise chart data and seasonal water temperature structure data of Yeosu and Gwangyang Port as input data. As a result, signal received during transmitting and receiving between candidate locations of stations could be calculated for each season. And, by estimating the minimum sound source level required to obtain an identifiable peak from the SNR value of the signal, the feasibility of tomography measurement could be assessed. It is expected that some important information, in constructing a tomography system with specifications appropriate for any coastal area of interest, can be provided in advance utilizing the method used in this study.

5:00

2pAO12. Enhancing remote sensing of water temperature using acoustic tomography and 3-D-RPS algorithm. Yixin Gao (Ocean College, Zhejiang Univ., No.1 Zheda Rd., Dinghai Dist., Zhoushan, Zhejiang 316021, China, gaoyxcn@zju.edu.cn), Xinyi Xie, Danni Wei, and Haocai Huang (Ocean College, Zhejiang Univ., Zhoushan, China)

Remote sensing of water temperature and flow dynamics is of paramount importance for water environment management. Coastal acoustic tomography (CAT), an innovative method for remote sensing in underwater environments, utilizes the dual-way travel time of multiple sound paths to reconstruct the underwater temperature field. However, the transmission experiments are susceptible to data noise caused by station drift and inaccuracies in travel time extraction. In this study, a three-dimensional Rousseeuw Phase-Space (3-D-RPS) thresholding algorithm is proposed to remove outliers in the travel time data obtained from Huangcai Reservoir using three CAT systems on September 16, 2020. Additionally, this work includes grid partitioning based on sound path propagation, calculation of sound path lengths, and reference transmission time within each grid, construction of a sparse matrix, and inversion of the two-dimensional temperature field and its associated inversion error. The accuracy and applicability of the proposed method are confirmed through a comparison with data obtained from temperature–depth (TD) sensors. The results demonstrate the precision and suitability of this approach. By effectively mitigating the impact of station drift and extraction accuracy issues, this method provides a reliable temperature field estimation.

Session 2pBAa

Biomedical Acoustics and Physical Acoustics: Biomedical Acoustics in Ophthalmology II

Jonathan Mamou, Cochair
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Tadashi Yamaguchi, Cochair
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Invited Paper

1:00

2pBAa1. Biomechanical properties of the myopic choroid. Quan V. Hoang (Singapore Eye Res. Inst., Singapore National Eye Ctr., Duke-NUS Med. School and Yong Loo Lin School of Medicine, National Univ. of Singapore, The Academia, 20 College Rd., Level 6 Discovery Tower, Singapore 169856, Singapore, donny.hoang@duke-nus.edu.sg), Kazuyo Ito (Tokyo Univ. of Agriculture and Technol., Koganei, Tokyo, Japan), Cameron Hoerig (Radiology, Weill Cornell Medicine, New York, NY), Yee Shan Dan (Singapore Eye Res. Inst., Singapore National Eye Ctr., Duke-NUS Med. School, Singapore, Singapore), Sally A. McFadden (The Univ. of Newcastle, Callaghan, New South Wales, Australia), and Jonathan Mamou (Radiology, Weill Cornell Medicine, New York, NY)

Retina-derived growth signals relayed from the choroid to the sclera cause remodeling of the extracellular matrix, resulting in myopic ocular elongation. However, no studies have assessed changes in choroidal biomechanical properties during myopia progression. The present study utilized 7-μm-resolution scanning acoustic microscopy (SAM) to assess biomechanical properties of choroids in guinea pig eyes with form-deprivation (FD) induced myopia. Specifically, neonatal guinea pigs underwent unilateral FD for 1 week (resulting in moderate to high myopia). 12-μm-thick serial cryosections of eyes were scanned with SAM and two-dimensional maps of bulk modulus (K), mass density (rho) were calculated. We found that the choroid had considerable intrinsic strength arising from its biomechanical properties and these were differentially affected by myopia in central and peripheral regions. Choroidal biomechanical values were also highly correlated with those in adjacent scleral regions, and the choroidal-scleral association was stronger in myopic eyes. In conclusion, biomechanical changes observed in the choroid of myopic eyes were mirrored to those observed in the adjacent sclera. These new findings suggest that the choroid remodeling may accompany myopia and opens the door to the source of the signals that cause scleral remodeling in myopia.

Contributed Papers

1:20

2pBAa2. Spectral-based quantitative ultrasound for assessing vision degrading myodesopsia. Cameron Hoerig (Radiology, Weill Cornell Medicine, 416 E 55th St., MR-007, New York, NY 10022, cah4016@med.cornell.edu), Justin Nguyen (VMR Consulting, Inc., Huntington Beach, CA), Jonathan Mamou (Radiology, Weill Cornell Medicine, New York, NY), Cedric Venuat (Quantel Medical, Cournon D'Auvergne, France), J. Sebag (VMR Consulting, Inc., Huntington Beach, CA), and Jeffrey A. Ketterling (Radiology, Weill Cornell Medicine, New York, NY)

The presence of clinically significant vitreous opacities, referred to as vision degrading myodesopsia (VDM), reduces contrast sensitivity (CS), and visual quality of life. Quantitative ultrasound (QUS) methods operating on grayscale B-mode images were previously developed and validated for quantifying vitreous echodensities. Although a subsequent study comparing the QUS methods across clinical scanners demonstrated no change in QUS efficacy, the methods are sensitive to differences in system parameters and B-mode image formation unless the backscatter data is normalized. Technological advancements in clinical scanners allow for easy access to raw radiofrequency (RF) data for more advanced processing methods. Here, we develop QUS methods that exploit RF echo data to generate robust user- and system-independent parameters for quantifying vitreous echodensities based on the echo signal power spectrum (PS). However, the sparsity of echodensities in the vitreous body is incompatible with underlying assumptions of QUS methods based on measurements of the backscatter coefficient or envelope statistics. We, therefore, implement QUS methods that parameterize the PS and correlate these parameters with visual function as measured by CS. Results from this study are the first step toward developing and validating robust, user-, and system-independent QUS methods for evaluating VDM.

1:40

2pBAa3. Fully optic characterization of acoustic impedance implementable in conventional optical microscope. Kazuki Tamura (Hamamatsu Univ. School of Medicine, 1-20-1, Higashiku Handayama, Hamamatsu, Sizuoka 4313192, Japan, k.tamura@hama-med.ac.jp) and Shinpei Okawa (Hamamatsu Univ., School of Medicine, Hamamatsu, Japan)

The intrinsic acoustic impedance (Z) changes with inflammation and canceration. This paper proposes an optically Z measurement method using a common optical microscope for simultaneous measurement of optical-acoustical observation. Photoacoustic (PA) waves were generated by the PA effect of the pulsed laser (a 527 nm nanosecond laser light) at the black ink and resin pasted area on the outside of the bottom of the petri dish (surface A). The generated PA waves propagated to the inner boundary of the petri dish and reflected, with a reflection coefficient of *R*. The reflected waves propagated to surface A and vibrated it. The surface vibration was measured with a self-built Sagnac interferometer using a contentious light laser and a 10x objective lens. Measurements were made air-filled and water-filled petri

dishes. The observed times of echo signal corresponded to the sound speed of longitudinal and transverse propagation. The reflected wave amplitude of air was larger than in a water-filled petri dish. They related that R_{water} and R_{air} are 0.24 and 1.00, respectively. There corresponds to the Z ratio of the polystyrene petri dish and the water or air inside. Thus, this result showed that sample Z was measured optically.

2:00

2pBAa4. Intraocular pressure measurement method using acoustic signals. Gyungmin Toh (Dept. of Mech. Eng., Hanyang Univ., 222, Wangsimni-ro, Seoul, Seongdong-gu 04763, Republic of Korea, avlrudals@gmail.com), Seongwook Jeon, Junhong Park (Dept. of Mech. Eng., Hanyang Univ., Seoul, Republic of Korea), and Won June Lee (Dept. of Ophthalmology, Hanyang Univ. College of Medicine, Seoul, Seongdong-gu, Republic of Korea)

Glaucoma can cause blindness, and one of the main causes has been found to be compression of the optic nerve by high intraocular pressure (IOP). In glaucoma, IOP is the most important biomarker for disease diagnosis, treatment, and monitoring. Among methods designed to measure IOP, direct contact with the eyeball is the most commonly used method due to its high accuracy. However, this method requires the intervention of a medical staff, and has problems, such as infection, fear, and corneal damage, which occur when the patient's IOP is measured. Therefore, in this study, while overcoming these problems, we intend to present a new IOP measurement method that inherits the advantages of the existing method. The eyeball has a structure in which body fluid is contained in an outer shell of viscoelastic material, and the change in pressure applied to the outer skin according to the inflow of body fluid appears as IOP. When an acoustic signal is emitted to an eyeball with a different IOP, the wave propagation characteristics vary depending on the composition of the body fluid, and the IOP is estimated. It is expected that the patient can directly measure and manage IOP using the proposed method.

Session 2pBAb

Biomedical Acoustics and Physical Acoustics: Novel Ultrasound Image Acquisition Technologies and Techniques

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Davide Fontanarosa, Cochair
QUT, Brisbane, Australia

Chair's Introduction—1:15

Contributed Papers

1:20

2pBAb1. Kidney stone twinkling power in elevated oxygen, carbon dioxide, and nitrogen environments. Laura Brownstead (Graduate Program in Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, lgb5113@psu.edu), Necole Streeper (Penn State Health Milton S Hershey Medical Ctr., Hershey, PA), and Julianna Simon (Graduate Program in Acoust., The Penn State Univ., University Park, PA)

Clinically, ~66% of kidney stones exhibit the Doppler ultrasound twinkling artifact. Previous studies show breathing pure oxygen at elevated pressures increases twinkling compared to ambient air. Here, we investigate how different gases influence twinkling on *ex vivo* and *in vivo* human kidney stones. 38 kidney stones were imaged with a research ultrasound system and ATL L7-4 transducer in elevated oxygen, carbon dioxide, and nitrogen environments. *In vivo* stones were imaged with the same ultrasound system and ATL C5-2 transducer in 25 human subjects before and after breathing pure oxygen for 15 min. IQ data were processed by calculating the average Doppler power, or twinkling, for a region of interest containing the stone. In the lab, increasing oxygen by 55%, carbon dioxide by 100%, and nitrogen by 17% increased twinkling power an average of $70 \pm 28\%$, $137 \pm 53\%$, and $79 \pm 31\%$, respectively. In human subjects, breathing oxygen increased twinkling power an average of $27 \pm 58\%$, with seven subjects showing a strong increase (>30%); stone composition and patient age influenced twinkling in patients breathing oxygen. These results suggest that varying ambient gases may be leveraged to increase the consistency of twinkling and improve kidney stone detection with ultrasound. [Work supported by the PSU Center for Biodevices.]

1:40

2pBAb2. Ultrafast plane wave imaging of breast lesions. Chris L. de Korte (Radiology and Nuclear Medicine, Radboud Univ. Medical Ctr., MUSIC 766, PO Box 9101, Nijmegen 6500HB, Netherlands, chris.dekorte@radboudumc.nl), Gijs Hendriks (Radiology and Nuclear Medicine, Radboud Univ. Medical Ctr., Nijmegen, Netherlands), Marcus Radicke (Siemens Heathineers, Fotschheim, Germany), and Ritse Mann (Radiology and Nuclear Medicine, Radboud Univ. Medical Ctr., Nijmegen, Netherlands)

Ultrafast plane wave imaging has great potential for 3-D functional ultrasound imaging but also comes with decreased lateral resolution, contrast, and penetration depth with respect to conventional imaging (CI). By utilizing coherent plane-wave compound imaging (CPWCI) and different beam forming techniques (Delay and Sum (DAS), Stolt's-fk, and Lu's-fk), we have demonstrated to improve resolution, penetration, and contrast-to-noise ratio. In this work we evaluate this technique in patients. A Siemens Sequoia system equipped with an 18L6HD, 14L5, and 10L4 was used to acquire CI and CPWCI data in 77 patients suspected of breast cancer visiting our clinic. CI data was acquired at three angles (-11, 0, and 11 degrees) and beamformed in the Sequoia and incoherently compounded while CPWCI data was acquired for 33 angles (-16:1:16 degrees) and subsequently processed off-line. In total, 9 malignant lesions, 35 benign lesions, and 32 normal other structures were found. For all three transducers, improved penetration was observed with CPWCI compared to CI. Additionally, contrast to noise ratio was also improved by CPWCI, especially at larger imaging depths. Qualitatively, an improved resolution appeared to be present, probably also caused by the improved contrast. Currently, a reader study in which radiologists have to score CI and CPWCI images sight by side while blinded to the acquisition mode is being prepared to test the efficacy of CPWCI in clinical practice. In conclusion, CPWCI bears great promise for improved breast cancer detection.

2:00

2pBAb3. A novel denoising strategy for ultrasound elastography. Magdoom Kulam (The Henry M Jackson Foundation for the Advancement of Military Medicine, National Institutes of Health, 13 South Dr., Bethesda, MD 20892, kulamnajmudeem2@mail.nih.gov) and Peter Basser (National Institutes of Health, Bethesda, MD)

Ultrasound imaging, while offering real-time tissue monitoring, portability, and other benefits, often suffers from poor signal to noise ratio (SNR). Noise has a particularly detrimental effect in US elastography since it gets amplified during the elastic modulus reconstruction process due to computation of high-order spatial derivatives of the deformation field needed to calculate the shear wave velocity within the sample. In this work, we introduce a novel scheme to denoise deformation fields using physically motivated "compatibility conditions," which are imposed as constraints. These conditions, well established in the solid mechanics literature, ensure that the deformation field remains continuous without exhibiting cracks, folds, or slips. Spatially uncorrelated noise gets significantly reduced when we apply them. Importantly, this approach avoids unnecessary smoothing and the ensuing loss of accuracy commonly observed in other denoising methods. We demonstrate the efficacy of this method using planar deformation fields obtained from simulations and those measured on an elasticity quality assurance (QA) phantom using acoustic radiation force impulse imaging (ARFI). Our pipeline results in smoother elasticity maps with 2x improved SNR, confirming the effectiveness of the approach.

Invited Papers

2:20

2pBAb4. Shear wave elastography evaluation of passive muscle stiffness *in vivo*. Jacqueline Roots (QUT, 2 George St., Brisbane, Queensland 4000, Australia, j.roots@qut.edu.au), Gabriel S Trajano, Christopher Drovandi, Adam Bretherton, and Davide Fontanarosa (QUT, Brisbane, Queensland, Australia)

Shear wave elastography is emerging as a viable method for diagnosing changes in the stiffness of tissues and can be applied to many aspects of the musculoskeletal system to allow the detection of muscle properties that have previously been unseen. It can be used to evaluate the change in stiffness caused by injury, inflammation, pathology or surgery, as well as the response to interventions. Determining the appropriate protocols and interpretation of results is the current major hurdle, limiting radiology clinics from adopting the technology. To overcome this, our research involved multiple phases of data collection to determine the anatomical variation and factors that influence passive muscle stiffness, measured on multiple machine vendors and varying patient positions. Measurements were performed on healthy volunteers *in vivo*. This research has investigated the variability of shear wave velocities along the length of the biceps brachii, as well as confirming the decrease in velocities with elbow joint flexion. The intersession variability of machines was excellent. Most notably, we demonstrated that separate bellies of the same muscle can differ in stiffness suggesting the need for a strong morphological and biomechanical understanding of muscles before development of evaluation protocols.

2:40

2pBAb5. Real-time 3-D reconstruction from multiple simultaneous ultrasound scans for enhanced imaging. Maria Antico (Australian e-Health Res. Ctr., CSIRO, 296 Herston Rd., Brisbane, Queensland 4006, Australia, maria.antico@csiro.au), Ashley Gillman (Australian e-Health Res. Ctr., CSIRO, Brisbane, Queensland, Australia), Gregg Belous, Leo Lebrat (Australian e-Health Res. Ctr., CSIRO, Brisbane, Queensland, Australia), Lars Petersson (CSIRO, Brisbane, Queensland, Australia), Beat Schmutz (QUT, Brisbane, Queensland, Australia), Jurgen Mejan-Fripp, Jason Dowling (Australian e-Health Res. Ctr., CSIRO, Brisbane, Queensland, Australia), and Davide Fontanarosa (QUT, Brisbane, Queensland, Australia)

Ultrasound imaging offers numerous benefits compared to other modalities, including portability, affordability, and real-time volumetric imaging capabilities. However, its limited field of view hinders image interpretation, significantly limiting its potential clinical applications. This study focuses on creating tomographic ultrasound imaging by automatically registering partially overlapping 3-D ultrasound volumes (coarsely localized in space). This approach can be used to gather a comprehensive field of view of the anatomical area imaged and to enhance the acquired image information. Potential applications involve both tomographic/wide-view imaging using conventional ultrasound probes and future technologies, where multiple ultrasound probes will be able to scan simultaneously the anatomy of interest. This work describes the development of advanced artificial intelligence and image analysis methods to create real-time 3-D anatomical reconstructions of a designated area of interest. Simultaneously (or quasi simultaneously) acquired ultrasound volumes with partial overlap can be spatially aligned using rigid registration, because they present geometrically similar information. We used novel supervised and unsupervised transformers and compared the results with more traditional methods such as mutual information and cross-correlation. This application contributes to creating new imaging processing techniques for ultrasound imaging and making this complex modality more user-friendly and clinically usable.

3:00–3:20 Break

Contributed Papers

3:20

2pBAb6. Artificial intelligence technology applied to an unoccluded catheter for recognition of hemodynamic changes. Li-Cheng Huang (College of Eng. and Sci., Feng Chia Univ. Master's Program of Electro-Acoust., Feng Chia Univ., No. 100, Wenhua Rd., Xitun Dist., Taichung City 407102, Taiwan, leo86627@gmail.com), Yu-Ting Tsai (Master's Program in Electro-Acoust., Feng Chia Univ., Taichung City, Taiwan), and Ming-Chih Lin (Children's Medical Ctr., Taichung Veterans General Hospital, Taichung City, Taiwan)

This study seeks to provide a reliable means for physicians to diagnose hemorrhagic complications associated with patent ductus arteriosus (PDA) in extremely low birth weight (ELBW) infants. We employ the YOLOv4 algorithm to analyze cardiac parameters from the echocardiography device's ultrasonic wave, such as left ventricular ejection time (LVET), left ventricular internal dimension-diastole (LVIDd), left ventricular internal dimension-systole (LVIDs), posterior wall thickness at end-systole (HES), and RR interval. The proposed model has achieved an impressive average detection accuracy of 89.4% for these five parameters. Additionally, the further evaluation of cardiac function can be done by deriving the end-systolic wall stress (ESWS) and left ventricle rate-corrected mean velocity of circumferential fiber shortening (mVcFc) values from these parameters. As a result, it indicates that the accuracy of ESWS and mVcFc values is 88.7% and 90.3%, respectively. With this deep learning-based approach, physicians can confidently evaluate the cardiac function of ELBW infants and diagnose PDA-related hemorrhagic complications.

3:40

2pBAb7. Deep learning-based microbubble localization towards improved super-resolution ultrasound. Scott J. Schoen (Radiology, Harvard Med. School and Massachusetts General Hospital, 101 Merrimac St., Boston, MA 02114, sschoenjr@mgh.harvard.edu), Ali K. Tehrani, and Anthony E. Samir (Radiology, Harvard Med. School and Massachusetts General Hospital, Boston, MA)

Ultrasound (US) is an indispensable tool for visualizing the microvasculature noninvasively. Over the last decade, US localization microscopy (ULM), which exploits microbubble contrast agents as effective point scatterers, has dramatically improved the spatial resolution attainable with US subject to the so-called diffraction limit (millimeter scale), and enabled mapping of vessels on the order of 10 μm. However, microbubble localization, a principal component of ULM, typically relies on bubble sparsity and compounding of many (10^2 to 10^4) frames. Thus, to enable sensitivity to more transient phenomena at the smallest scales, optimally sensitive and specific means of localization are required. Drawing on newly available ground truth data (ULTRA-SR Challenge, IUS 2022), we apply convolutional neural network (DeepLabv3) to perform localization via segmentation of the microbubble regions and identification of their centroids. Following training on 80% of the available

2p TUE. PM

simulated US frames, the network demonstrated localization precision 0.88 and recall 0.50 (tolerance of 0.5λ) on the test frames, with bias on the order of 0.1λ, and inference time of 40 ms on Nvidia RTX3090. Such accurate and sensitive localizations have significant promise toward elucidating hemodynamics at still finer spatial and temporal resolutions.

4:00

2pBAb8. Blind-label acoustic subwavelength imaging. Chu Ma (Elec. and Comput. Eng., Univ. of Wisconsin-Madison, 1415 Eng. Dr., Rm. 3436, Madison, WI 53706, chu.ma@wisc.edu) and Jinuan Lin (Elec. and Comput. Eng., Univ. of Wisconsin-Madison, Madison, WI)

There is a long-existing tradeoff between the imaging resolution and the penetration depth in acoustic imaging caused by the diffraction limit. Most of the existing subwavelength imaging approaches that address this trade-off require exogenous "labels," such as metamaterials or contrast agents, to be deposited close to the objects. Those labels need to either remain static or be tracked precisely during imaging, therefore, are extremely restrictive in practical applications. This talk will present our recent work on a "blind label" approach for acoustic subwavelength imaging. The "blind labels" are randomly distributed acoustic scatterers with deep-subwavelength sizes whose exact locations and trajectories are unknown. A hardware/software co-design framework is developed that is composed of two parts: 1) spatial mixing: a physical process that converts the originally evanescent components in the scattered waves from the object to propagating components that can reach the far-field detector; 2) computational image reconstruction. Our experiments both in air at KHz frequency range and in water at MHz frequency range achieved multi-fold resolution enhancement. The proposed "blind-label" approach relaxes the restrictions of depositing controlled labels close to the object, therefore significantly improving the practicality of acoustic subwavelength imaging in acoustic sensing, imaging, and communication applications.

Invited Paper

4:20

2pBAb9. Quantification of liver fat using ultrasound technology. Heather Allen (Queensland Univ. of Technol., QUT Gardens Point Campus, Brisbane, Queensland 4000, Australia, h2.allen@qut.edu.au) and Christopher Edwards (Queensland Univ. of Technol., Brisbane, Queensland, Australia)

Non-alcoholic fatty liver disease (NAFLD) is a growing concern worldwide, creating significant clinical and economic burdens. NAFLD has the potential to progress to advanced liver disease, which is associated with significant morbidity and related mortality. Research to address these challenges is ongoing and requires the use of reliable liver fat quantification tools to evaluate effectiveness. Magnetic resonance imaging has been established as a reference standard to replace liver biopsy (the traditional gold standard); however, cost and access limit its use in large patient groups. Recent advances in ultrasound-based liver fat quantification technology are promising, offering a more accessible, cost-effective method for estimating the quantity of liver fat. In this presentation, the authors outline how ultrasound attenuation and backscatter coefficients can be used to estimate liver fat percentage, discussing how this new tool has the potential to greatly impact the assessment and management of fatty liver disease.

Contributed Papers

4:40

2pBAb10. Demonstration of medical imaging using flexible polymer ultrasound technology. Laurent Fillinger (Acoust. and Sonar, TNO, Oude Waalsdorperweg 63, The Hague 2597AK, Netherlands, laurent.fillinger@tno.nl), Lars Hörchens (Acoust. and Sonar, TNO, The Hague, Netherlands), Laurens C.J. M. Peters, Roy G.F. A. Verbeek, Bart Peeters (Holst Ctr., TNO, Eindhoven, Netherlands), Thijs Schrama, Egon J. W. Merks-Swolfs (Acoust. and Sonar, TNO, The Hague, Netherlands), Jan-Laurens van der Steen (Holst Ctr., TNO, Eindhoven, Netherlands), Arno W. F. Volker (Acoust. and Sonar, TNO, The Hague, Netherlands), Gerwin H. Gelinck (Holst Ctr., TNO, Eindhoven, Netherlands), and Paul L.M. J. van Neer (Acoust. and Sonar, TNO, The Hague, Netherlands)

We present a novel ultrasound transducer technology based on piezo polymers. Its fabrication process (PillarWave^{TM}), based on creating hexagonal micro-structured PVDF-TrFE pillars. This makes it fully flexible and allows scaling to large apertures ($>15 \times 15$ cm^2). Large area flexible transducers enable to select relevant field of view in automated post-processing. This can alleviate the need for a skilled sonographer and enable applications of echography outside clinics. To demonstrate its performance, a 128 element 8 MHz linear array is designed and manufactured. Its total thickness is 0.1 mm. Its acoustic characteristics are determined demonstrating 5.2 kPa/V transmit efficiency and 150 mV/Pa sensitivity. The receive bandwidth at -6 dB is larger than 100%. A peak pressure in excess of 1 MPa at 4 cm from the array is measured in water. The transducer array is connected to a Verasonics Vantage research medical ultrasound system. The imaging performance of the array is demonstrated experimentally. Real-time imaging of an *in vitro*- medical phantom using plane wave compounding and delay-and-sum beamforming is demonstrated at 15 frames per second, with point spread functions at -20 dB of 1.3 and 0.92 mm in the lateral and depth directions, respectively. Finally, imaging of a human carotid *in vivo* is demonstrated.

5:00

2pBAb11. Wearable ultrasound system for assessing muscle function during physical activity. Erica L. King, Ahmed Bashatah (Bioengineering, George Mason Univ., Fairfax, VA), Brian M. Guthrie, Margaret T. Jones (Kinesiology, George Mason Univ., Fairfax, VA), Qi Wei, Siddhartha Sikdar (Bioengineering, George Mason Univ., Fairfax, VA), and Parag V. Chitnis (Bioengineering, George Mason Univ., 4400 University Dr., 1J7, Fairfax, VA 22030, pchitnis@gmu.edu)

Conventional ultrasound transducers, which are meant for hand-held operation, are not ideally suited for MSK-US during dynamic physical activity. We have developed a custom wearable ultrasound system for Motion-mode (M-mode) MSK-US, which enables multi-site hands-free sensing of tissue movement during physical activity such as that encountered during rehabilitation exercises. Participants performed a jumping task on a force plate with a sensor attached to their right vastus lateralis (VL). Mean pixel value (MPV) and normalized pixel differences (NPDs) were acquired from the images during the jumping task. Signals were filtered and normalized to root-mean square baseline and percent change was calculated for peaks of jump take-off and landing. The correlation coefficient was calculated to determine a relationship between metrics. Concurrently acquired force-plate data showed that the percent change during take-off and landing was $0.28\% \pm 0.09$ and $0.66\% \pm 0.17$ relative to baseline for each peak, respectively. MPV values changed $0.42\% \pm 0.29$ and $0.81\% \pm 0.17$, and NPD changed $0.41\% \pm 0.32$ and $0.84\% \pm 0.12$ for corresponding peaks. Overall results indicated significant correlation between force data and MSK-US ($p < 0.05$). Our system can maintain acoustic coupling during complex movements providing real-time feedback and expanding the potential for wearable MSK-US sensors.

Session 2pCA

Computational Acoustics and Physical Acoustics: Data-Driven Methods in Acoustics and Vibration II

Marcus Maeder, Cochair
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Johannes D. Schmid, Cochair
Chair of Vibroacoustics of Vehicles and Machines, Technical Univ. of Munich, Boltzmannstraße 15, Garching 85748, Germany

Chair's Introduction—12:55

Invited Paper

1:00

2pCA1. Unveiling weak auditory evoked potentials using data-driven filtering. Stefan Jacob (Infrasound Res. Group, National Metrology Inst. of Germany, PTB, Bundesallee 100, Braunschweig 38116, Germany, stefan.jacob@ptb.de) and Christian Koch (Sound, National Metrology Inst. of Germany, PTB, Braunschweig, Germany)

Auditory evoked potentials (AEPs) are commonly used to objectively evaluate sound perception in humans. Close to the hearing threshold and for low frequencies, efficient filtering of AEP from other brain activities is of major concern due to weak potentials and the requirement of long averaging times. Filtered AEP data are well-interpretable and useful, especially in medical and psychological diagnostics. Here, we present two data-driven approaches for efficient AEP filtering. First, neural networks of different architectures trained for EEG denoising are used to extract weak late-response AEP for low-frequency and infrasonic stimuli. During the design of the networks, we leveraged knowledge of the specific characteristics of the expected AEP data. Second, a singular value decomposition (SVD) of EEG data is evaluated, attempting to create classifiers for the presence of weak late-response AEP modes. We anticipate that the evaluation of AEP with data-driven methods can support researchers and scientists, for example, with real-time evaluation and diagnosis of acoustic-induced discomfort.

Contributed Papers

1:20

2pCA2. Enhancing heart disease diagnosis through precise heart sound including infrasound. Youngsin Kim (Pohang Univ. of Sci. and Technol., 39 Jigok-ro, KIRO Rm. 416, Nam-gu, Pohang-si, Gyeongsangbuk-do 37666, Republic of Korea, kysin@postech.ac.kr), Mihyung Moon, Seokwhan Moon (Dept. of Thoracic and Cardiovascular Surgery, Seoul St. Mary's Hospital, College of Medicine, The Catholic Univ. of Korea, Seoul, Republic of Korea), and Wonkyu Moon (Pohang Univ. of Sci. and Technol., Pohang-si, Gyeongsangbuk-do, Republic of Korea)

Recent research has been pushing the frontier of diagnosing cardiovascular diseases, which have a high mortality rate globally. The need for swift responses to acute diseases has underscored the importance of continuous health monitoring and the development of monitoring diagnostic technology. Our work focused on an overlooked aspect—the microphones used to measure heart sounds. Existing ones, particularly capacitive MEMS types, measure only heart sounds primarily within the audible frequency range, potentially limiting the diagnostic accuracy. To overcome these limitations, we used microphones capable of measuring down to infrasound areas. We also designed an experimental environment to capture the precise heart sounds. The crux of our approach was using the full information of heart sounds through infrasound measurements. A convolutional neural network-based deep learning model was employed for algorithm development and validated using the PhysioNet 2016 CinC's open-source heart sound data. We conducted an experiment on the impact of low-frequency components on heart sound diagnosis using a new infrasound dataset measured at Seoul St. Mary's Hospital of the Catholic University of Korea. Accuracy and sensitivity, respectively, improved on average by 2% and 4%, showing that datasets including low-frequency components generally performed better.

1:40

2pCA3. Sensor fusion for simultaneous measurement of micro-vibrations. Luke Phillips (Ctr. for Audio, Acoust. and Vib., Univ. of Technol. Sydney, Ultimo, New South Wales 2007, Australia, luke.phillips-1@student.uts.edu.au), Sebastian Oberst, and Shahrokh Sepehrirahnama (Ctr. for Audio, Acoust. and Vib., Univ. of Technol. Sydney, Botany, New South Wales, Australia)

Low-amplitude micro-vibrations are common in nature and engineering, throughout structural applications, and biology. The ability to accurately measure and analyse these vibrations in the presence of noise (unwanted signal content) has far reaching consequences across many fields of acoustics. Methodologies for the enhancement and improvement of such signals are, therefore, sought. We explore the capability of sensor fusion in combination with Kalman filtering (KF), using pairs of accelerometers to improve measurement of low-amplitude micro-vibrations. Prior research of pairing sensor fusion with machine learning approaches like KF, support vector machines, or coherence analysis reported up to 90% reductions in "ghost" detections. This research attempts to extend this success to micro-vibrations where broadband noise, and external perturbations can have dramatic

impacts on measurability. A pair of accelerometers has been placed both parallel and perpendicular to the axis of an excitation in pine timber planks of varying dimensions and cuts. Simultaneous measurement of $\sim$5 N excitations with an automated hammer at varying distances are recorded and patterns observed in the time domain through preliminary analysis in MATLAB. Features identifiable from this data become clearer as compared to conventional approaches and have potential applications in non-invasive early predictive analysis of structures for timber pest control.

Invited Paper

2:00

2pCA4. Recurrence rate spectrograms for impact localization in wood. Thore Hertrampf (Ctr. for Audio, Acoust. and Vib., Univ. of Technol. Sydney, Sydney, New South Wales, Australia), Sebastian Oberst (Ctr. for Audio, Acoust. and Vib., Univ. of Technol. Sydney, 123 Broadway, Ultimo, New South Wales 2007, Australia, sebastian.oberst@uts.edu.au), and Shahrokh Sepehrirahnama (Ctr. for Audio, Acoust. and Vib., Univ. of Technol. Sydney, Botany, New South Wales, Australia)

Characteristics like cellular grain structure, inhomogeneous density, aging, and altering environmental conditions (moisture, temperature) give wood highly anisotropic viscoelastic properties and non-linear vibrational wave propagation properties. Nonlinearity limits the use of linear methods, such as modal analysis and parameter identification via transfer functions. Acoustic localization of natural damage to wood, like crack growth, is of general interest in structural health monitoring of timber structures. Time-difference of arrival or energy attenuation is commonly used for localization, which are prone to boundary reflections or require the frequency response function. Recent advancements in machine learning-based classification of non-linear signals can achieve a much higher accuracy when recurrence rate-based spectrograms are used compared relative to conventional short-time Fourier transforms, especially in the presence of noise. Hence, in this work, multi-sensor measurements of impulse induced vibration in wood beams are classified by their distance to the excitation, based on their time series, avoiding a priori knowledge of a transfer function for the localization. The machine learning model is trained across various widths and thicknesses of samples, giving a localization estimate independent of beam dimensions. This research will contribute to early detection of damage in the field of vibration-based structural health monitoring of wood.

Contributed Paper

2:20

2pCA5. Generative model-based transmission health monitoring of construction machinery in real factory. Hao Di (Waseda Univ., Nishi-Waseda Campus,3 Chome-4-1 Okubo, Shinjuku City, Tokyo 169-8555, Japan, dihao@toki.waseda.jp), Xinpei Li, Yasuhiro Oikawa, and Shoichi Kiya (Komatsu, Ishikawa-ken, Japan)

In real factory production processes, transmission systems occasionally produce abnormal noises that deviate from their normal sound patterns. Detecting these anomalies is crucial for identifying the underlying causes and ensuring the quality of products. The traditional health monitoring of the transmission system in construction machinery relies on the expertise of skilled workers. In order to enhance detection capabilities during instances of abnormal noise occurrence, conserve human resources, and provide a technological foundation for future automation in production, we propose a transmission system sound detection method based on a generative model. In this study, normal transmission system sounds were collected from Komatsu Corporation using professional equipment, and log-Mel features were extracted from the raw audio data. The generative model was employed as a classifier and trained on normal transmission system sounds, with reconstruction error serving as the threshold to distinguish between normal and abnormal sounds. The experimental comparison with traditional deep learning methods verified the effectiveness of our approach. The results demonstrate the feasibility of applying the generative model as a classifier for health monitoring of the transmission system in Komatsu's construction machinery.

2:40–3:00 Break

Invited Paper

3:00

2pCA6. Diagnosis of robotic arm anomalous sounds using unsupervised learning algorithms. Zi-Wei Zheng (Feng Chia Univ., No. 100, Wenhua Rd., Xitun Dist., Taichung 407102, Taiwan, P1100405@o365.fcu.edu.tw), Tzu-huan Peng, and Yu-Ting Tsai (Feng Chia Univ., Taichung city, Taiwan)

In this paper, we develop a technique that integrates acoustics detection with artificial intelligence, empowering robotic arms in production environments with the capacity to monitor, diagnose, and predict anomalies. We combine the deep learning techniques of AI with audio signals collected by an array of microphones. We predict the health status of equipment systems and prevent equipment failure by using this sound data. Our novel approach focuses on converting and analyzing environmental acoustics data. The distinctive contribution of this research lies in its strategy learning and inference. By harnessing large-scale audio data information, we establish AI learning models that automatically parse input data to recognize audio features. We also employ device scheduling operations for comparative detection, endowing the equipment with abnormal diagnosis and prediction capabilities. We utilize commercially available microphone devices for sound collection, voice activity detection, and spectrogram reduction for signal preprocessing and capture before we create a dataset. Finally, we use the Gaussian mixture model (GMM) for unsupervised learning, estimating the probability density of data, and calculating GMM scores of data points for in-depth analysis. In this way, we can judge whether the sound signals collected by the microphone are abnormal.

3:20

2pCA7. Machine learning predictive model for motor moise by expanding dataset using electromagnetic force characteristics. Wontae Jeong (NVH Fundamental Tech. Cell, Mobis, Mabuk-ro 240 beon-gil, Hyundai Mobis, Yongis-Si, Gyeonggi-do 16891, Republic of Korea, wtjeong@mobis.co.kr) and Wontae Jeong (NVH Fundamental Tech. Cell, Mobis, Yongis-Si, Gyeonggi-do, Republic of Korea)

To understand the variations of electric vehicle motor noise, the electromagnetic force change due to the tilting and the eccentricity between the rotor and the stator on the noise should be studied. However, it takes enormous computation time to consider all the various conditions, so machine learning (ML) model that can reduce the analysis time of electromagnetic force and noise was developed. The developed ML model showed very high accuracy prediction performance with an R^2 value higher than 0.99 even for data that was not used at all for training. The speed increase was more than 100 times of FE analysis. The authors doubt that the high accuracy of the developed model implies the potential for electromagnetic force data structure to be much simpler than it appears. Therefore, various analyzes were conducted on the force and uneven magnetic pull. The change in the temporal-spatial main frequency component was not large, and only the change in the side order was meaningful. It was confirmed that to create a predictive model using a smaller dataset for not only the UMP but also all the individual teeth force is possible. This new way of creating a training dataset for an ML model helps to increase the training efficiency. Based on the generated prediction model, it can be used to identify changes in noise characteristics in major rotational orders and derive suitable improvements.

3:40

2pCA8. Fundamental study for vehicle abnormal sound detection and the direction estimation methods using machine learning with noise reduction. Masanori Takagi (Mech. Eng., Osaka Inst. of Technol., 5-16-1, Omiya, Asahi-ku, Osaka-shi, Osaka 535-8585, Japan, m1m22420@st.oit.ac.jp)

Detecting abnormal sounds during vehicle running condition is important to find out the vehicle's problems at the early stage. In the mass production process, the detection test is generally carried out by the inspector. However, continuous inspection by a human for long time is hard and miss false inspection may occur. In addition, normal running noise, such as road noise, deteriorates the inspection accuracy. Therefore, an automatic accurate abnormal sound detection method under the normal noise is necessary to improve the detection accuracy with small man-hour. Furthermore, if the method has an ability to localize the abnormal sound direction, the countermeasure also becomes easy. In this study, we considered the noise detection method using machine learning with a noise reduction method and attempted the noise direction using two microphones. As a fundamental study, we applied these methods in an open room and the result showed that the method could detect the abnormal sound under normal noise condition and indicate the direction well in the simple verification test.

4:00

2pCA9. An improved computational bioacoustic monitoring approach for detecting sparse features. Ben J. McEwen (Comput. Sci. and Software Eng., Univ. of Canterbury, 20 Kirkwood Ave., Upper Riccarton, Christchurch 8041, New Zealand, ben.mcewen@pg.canterbury.ac.nz), Kaspar Soltero, Isaac Cone, Stefanie Gutschmidt (Mech. Eng., Univ. of Canterbury, Christchurch, New Zealand), Andrew Bainbridge-Smith, James Atlas, and Richard Green (Comput. Sci. and Software Eng., Univ. of Canterbury, Christchurch, New Zealand)

The collection and annotation of bioacoustic data presents a number of challenges to researchers, often constraining analysis to highly vocal species. Computational tools allow monitoring to be extended to less vocal and more challenging species but data limitations remain an issue. We present a human-in-the-loop approach that combines the efficiency of computational tools with the accuracy of human analysis. We use a wavelet-based segmentation method that automatically extracts transient features within field recordings which can reduce data by up to 90% and requires as few as one reference feature. Segmented features are then used to fine-tune a transformer-based model, audio spectrogram transformer (AST), the output of which is verified by a human and the adjusted data fed back into the model to improve performance over time. We also present an outlier detection approach based on Mel-frequency Cepstral Coefficients. Coefficients are projected to 2-D and outliers are detected using silhouette score. This approach was able to achieve 98.8% validation accuracy on a binary classification task using a limited dataset of 200 5-min recordings with sparse features (occurrence rates of less than 1%). This approach makes real-time bioacoustic monitoring of less-vocal species a possibility.

4:20

2pCA10. Application of one-dimensional convolutional neural network in guitar factory quality inspection. I-Hsin Chen (College of Eng. and Sci., Feng Chia Univ., 5F.-3, No. 45, Fengda Rd., Xitun Dist., Taichung City, Taiwan (R.O.C.), Taichung 407032, Taiwan, t28650636@gmail.com), Yu-Ting Tsai (Master's Program in Electro-Acoust., Feng Chia Univ., Taichung city, Taiwan), and Zi-Wei Zheng (College of Eng. and Sci., Feng Chia Univ., Taichung, Taiwan)

After the completion of guitar manufacturing, it is necessary to conduct quality testing and final adjustments to ensure that the guitar meets the factory standards. This stage often requires the involvement of professional musicians. In this project, a supervised learning approach is employed, where guitar vibration signals are inputted and correspond to the output sound quality. Multiple test subjects perform guitar strumming, and the guitars are adjusted to simulate various abnormal conditions for experimental comparison. By classifying the differences in energy and frequency, different guitar factory standards are identified, and the stability of guitar quality is explored. For data analysis, a one-dimensional convolutional neural network (1D CNN) is used as the base model, analyzing the Mel-frequency cepstral coefficients to identify elements of stable guitar quality and tonal characteristics. The research results show that the model can predict variations due to factors, such as environmental temperature, humidity, and worn-out strings, ensuring objective and consistent analysis of each guitar's factory quality. This innovative technology eliminates the need for traditional labor-intensive inspection procedures, enabling the traditional luthier industry to utilize automated methods for rapid factory testing.

Session 2pNSa

Noise and Physical Acoustics: Aeroacoustic Sources and Fields II

Danielle Moreau, Cochair
UNSW Sydney, Sydney 2015, Australia

Con Doolan, Cochair
School of Mechanical and Manufacturing Engineering, UNSW Sydney, Sydney 2052, Australia

Chair's Introduction—12:55

Contributed Papers

1:00

2pNSa1. Experimental investigations on aerodynamic and psychoacoustic characteristics of three-blade looprop propeller. Jianwei Sun (Chiba Univ., 1-33 Yayoi-cho Inage-ku, Chiba-shi, Faculty of Eng. 14, Chiba 2638522, Japan, sunjianwei0325@gmail.com), Koichi Yonezawa (Central Res. Inst. of Elec. Power Industry, Abiko, Japan), Eiji Shima (Japan Aerosp. Exploration Agency, Tokyo, Japan), and Hao Liu (Chiba Univ., Chiba, Japan)

Aeroacoustic noise in multiple rotor drones has been increasingly recognized as a crucial issue. This study focuses on addressing this challenge by introducing a novel low-noise looprop propeller design with three blades. Unlike traditional propellers, the looprop propeller utilizes three closed-loops to generate thrust. Through comprehensive experimental investigations conducted in an anechoic chamber using a hover stand test, we conducted an integrated study of the propeller's aerodynamic, aeroacoustic, and psychoacoustic characteristics. The results demonstrate that the three-blade looprop propeller achieves a notable reduction in the overall sound pressure level (OASPL), particularly in tonal noise at the blade passing frequency where the tone reduction amounts to approximately 15 dB. Additionally, we employed Zwicker psychoacoustic annoyance models to evaluate the propellers' psychoacoustic performance and discussed the environmental noise mask effect. The findings indicate that the three-blade looprop propeller exhibits an improved sound quality while maintaining aerodynamic performance comparable to that of conventional propellers. Overall, our study suggests that the looprop propeller design offers significant advancements in acoustic performance, particularly in terms of psychoacoustic characteristics. These findings hold promising implications for the field of drone technology and the potential for notable improvements in noise reduction.

1:20

2pNSa2. Numerical investigation on the rotor-turbulence interaction noise. Denghui Qin (College of Eng., Peking Univ., Beijing, China, qindenghui@pku.edu.cn) and Xun Huang (Peking Univ., Beijing, China)

Noise from a turbomachinery component, such as compressors and turbines, is an important issue to be resolved in aerospace and ocean engineering applications. The rotor of a turbomachinery assembly usually could operate downstream to other flow structures (fuselages, stators, control surfaces, etc.) that produce either large-scale or small-scale turbulence structures, which are highly complex and are being distorted while they are ingested into the rotor, which generate the rotor-turbulence interaction noise. The presented work conducted numerical simulation research on a ten-bladed rotor ingesting different turbulence structures. The large eddy simulation (LES) and Ffowcs-Williams and Hawkings (FW-H) methods were used to predict the flow-induced noise from the rotor. The results show that the time-space evolution of the upstream turbulent structure when it passes towards the rotor. In addition, the differences in the turbulence ingestion noise at different inflow speeds and rotational speeds were studied. Furthermore, the sound generation mechanism of turbulent ingestion noise was explored using a newly proposed near-field sound source analysis method. In summary, this study can help to improve the understanding of the physical mechanisms of unsteady forces and noise generated by a rotor when it interacts with ingesting turbulent flows.

1:40

2pNSa3. A new model of Crossflow fluid-structure instabilities for helicopter blades. Richard Howell (Platforms Div., DSTG, 506 Lorimer St., Port Melbourne, Victoria 3207, Australia, richard.howell2@defence.gov.au), Tony Lucey (Curtin Univ., Bentleigh, Western Australia, Australia), Thomas Ng, Konstantinos Tsigklifis, and Paul Dylejko (Platforms Div., DSTG, Melbourne, Victoria, Australia)

Crossflows that induce fluid-structure instabilities arise in many engineering applications, such as flow over a road bridge or an aircraft wing. At a certain flow speed, an aeroelastic flutter instability is induced arising from the coupled translational and rotational movement of the structure. This instability is modified in a subgroup of these applications that include the additional effect of rotation, e.g., propellers. The structure now experiences a linearly varying effective mean flow (faster near the tip than at the root) and an induced stiffness (via tension) due to the centrifugal force that is also spatially varying. The flutter instability can create vibration fatigue issues and can also create a significant acoustic signature. This paper reports a new linearised numerical model of the problem that fully couples a finite-difference Euler–Bernoulli beam model to a Theodorsen unsteady lift model that includes these effects of rotation. Comparisons with theoretical predictions in non-rotating flow demonstrate the effect on the trends for the onset of linear instability. The model is then applied to the problem of helicopter blade flutter, and the effects of various strategies for delaying the onset of linear instability are modelled.

2:00

2pNSa4. Aeroacoustic and aerodynamic measurements of a ducted propeller. Jingang Li (School of Mech. and Manufacturing Eng., UNSW, Rm. 408, Ainsworth Bldg., Eng. Rd., Kensington, Sydney, New South Wales 2052, Australia, jingang.li@student.unsw.edu.au), Justin Malkki (School of Mech. and Manufacturing Eng., UNSW, Sydney, New South Wales, Australia), Yendrew Yauwenas (School of Mech. and Manufacturing Eng., UNSW, Sydney, New South Wales, Australia), Con Doolan (UNSW, Sydney, New South Wales, Australia), and Danielle Moreau (School of Mech. and Manufacturing Eng., UNSW, Sydney, New South Wales, Australia)

The flexibility and wide range of possible applications in both civilian and military areas make unmanned aerial vehicles (UAVs) an object of continuous research. An important focus of such work is the propulsion system and ways of improving its efficiency. A ducted propeller is a common solution to increase the thrust and power efficiency of the propulsion system, which is especially beneficial for UAVs driven by electric motors due to their limited battery capacity; however, noise created by UAVs can not only annoy the public but may also reduce its acoustic stealth. Hence, the aim of this paper is to describe a test rig that can take aeroacoustic and aerodynamic measurements of both ducted and open propellers and present some initial experimental results. Both aeroacoustic and aerodynamic measurements were taken in an anechoic chamber with an arc of 6 $^1/_2$ in. microphones to record noise level from different directions. The APC 12×8 propeller was used together with the SunnySky x2820 brushless motor.

2:20

2pNSa5. An investigation of drone propeller noise generation in a turbulent wake. Zhengpang Bian (Univ. of New South Wales, School of Mech. and Manufacturing Eng., UNSW Sydney, Kensington, New South Wales 2052, Australia, z5197992@ad.unsw.edu.au), Yendrew Yauwenas, Danielle Moreau, and Con Doolan (UNSW, Sydney, New South Wales, Australia)

Unmanned aerial vehicles (UAVs) are becoming popular as they can be used in a range of applications including surveillance, delivery, agriculture, aerial photography, and search and rescue. This paper presents an experimental study to characterise the effect of turbulent inflow disturbance on the noise generated by a drone propeller using a low-noise, open-jet anechoic wind tunnel facility at UNSW. Advanced precision composites (APC) two-bladed, 12-inch rotors were tested at rotational speeds between 4000 and 8000 RPM at a freestream velocity of 15 m/s. The inflow disturbance upstream of the propeller is the wake of a NACA0012 profile wing to which the propeller is mounted. Hot-wire anemometry has been used to characterise the wake of the strut. The turbulence intensity, mean velocity, and turbulence length scale is calculated to characterise the turbulence ingested by the propeller. Acoustic measurements have been taken with multiple microphones to study the sound directivity and noise radiation of the propeller. A load cell is used to measure the torque and thrust produced by the propeller. This experimental program aims to characterise the wake profile ingested by the propeller and examine the associated noise generation.

2:40

2pNSa6. Leveraging machine learning and similarity judgements to identify perceptually relevant acoustic features of small, unmanned aircraft system signal similarity. Mason A. Reeves (Ball Aerosp. & Technologies Corp., 2610 Seventh St., Building 441, WPAFB, OH 45433, mason.reeves.ctr@us.af.mil), Frank S. Mobley, Steven C. Campbell, Alan T. Wall (Air Force Res. Lab., WPAFB, OH), Reese Rasband, and Gregory Bowers (Ball Aerosp. & Technologies Corp., Fairborn, OH)

Auditory signals can be described quantitatively by a set of measurable acoustic features (i.e., zero crossing rate, attack slope, etc.) or qualitatively with adjectives, such as whooping, thunderous, melodic, or in comparative terms such as different, louder, etc. Listeners can rate the similarity of signals and assign qualitative descriptions relatively easily; however, most lack the ability to articulate the quantitative basis of these judgments. Because the qualitative differences in signals typically correlate to a measurable difference in the acoustic features, signal similarity ratings can be used to recover the acoustic features that define signal similarity. In the present study, subjects were given combinations of small, unmanned aircraft systems (SUAS) signals consisting of either two different SUASs or SUAS and non-SUAS and asked to rate similarity on a scale from non-similar to highly similar. Using these similarity ratings along with acoustic difference features, machine learning algorithms were trained to predict human responses. These algorithms predict the position of a withheld UAS signal within the similarity feature-space. Crucial prediction acoustic difference features are extracted from the algorithms via feature importance and sensitivity analysis techniques. The extracted acoustic difference features may be inferred as prominent information impacting the human perception of signal similarity.

3:00

2pNSa7. Prediction and analysis of aeroacoustic characteristics of a real-size high-speed train running in open field using compressible Large Eddy Simulation and vortex sound source. Kwongi Lee (School of Mech. Eng., Pusan National Univ., 2, Busandaehak-ro 63beon-gil, Busan, Republic of Korea, dlrnjsrl93@pusan.ac.kr), Cheolung Cheong (School of Mech. Eng., Pusan National Univ., Busan, Republic of Korea), and Jaehwan Kim (Hyundai Rotem, Uiwang-si, Republic of Korea)

With the rapid advancement of technology, high-speed trains have been requested to increases their speeds. Aerodynamic noise generated from the external flow fields around these trains has become a critical factor to be addressed during a design phase. Accurate prediction of flow-induced noise in high-speed trains requires high-resolution sound source generation in the near-field and precise noise propagation analysis in the acoustic field without numerical dissipation. This necessitates the appropriate design of grids, taking into account aerodynamic noise generation and propagation in conjunction with the length scales of the relevant components of a real-size train. To tackle this challenge, the present research employs a three-dimensional compressible Large Eddy Simulation (LES) technique with high-resolution grids to simultaneously calculate the external flow and acoustic fields of a real-size high-speed train consisting of five cars running in an open field. A comprehensive analysis is conducted to evaluate the contributions of major components, namely, a cap, pantograph, bogie, intercoach, and HVAC cover, which are well-known as significant contributors to high-speed train flow-induced noise. The study utilizes the vortex sound source approach to analyze the generation mechanism of flow-induced noise for each component, providing valuable insights that could potentially aid in reducing aerodynamic noise.

Session 2pNSb

Noise: Sonic Boom II

Alexandra Loubeau, Cochair
NASA Langley Research Ctr., 2 N Dryden St., Hampton, VA 23681

Victor W. Sparrow, Cochair
Grad. Prog. in Acoustics, Penn State Univ., 201 Applied Science Bldg., University Park, PA 16868

Contributed Paper

2:00

2pNSb1. Analytical closed-form solution of the Navier–Stokes equations for the aerodynamic near-field and sonic boom from axisymmetric bodies. Steven A. Miller (Mech. and Aerosp. Eng., Univ. of Florida, PO BOX 116250, 939 Sweetwater Dr., Gainesville, FL 32611, saem@ufl.edu)

An analytical closed-form solution is presented for the aerodynamic near-field and ground signature from an axisymmetric body with a low thickness ratio. The Navier–Stokes equations are formulated as a boundary value problem that incorporates the incoming ambient flow-field and the aerodynamic properties on the body surface. The shape of the aerodynamic body is defined as a product of generalized functions. A direct solution for the density of the aerodynamic near-field, represented as a function of both space and time, is proposed through the integration of the Navier–Stokes equations in a generalized functional form. Pressure, temperature, velocity, and Mach number are then derived in the near-field. The methodology, being fully nonlinear, surpasses the traditional F-function, impulse, and hypersonic similarity theories originally developed for near-field prediction. The presentation outlines the major steps in deriving the analytical solution and provides predictions from an aerodynamic body in the near-field, along with the associated ground signature. The methodology is focused on aerodynamic bodies operating at high-speeds, ranging from the supersonic to the hypersonic regime. [Work supported by the Defense Advanced Research Project Agency, under Grant No. W911NF-21-1-0342.]

Invited Papers

2:20

2pNSb2. Developments regarding Type II secondary sonic booms and topography. Victor W. Sparrow (Grad. Prog. in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, vws1@psu.edu) and Kimberly Riegel (Phys., Farmingdale State College, Farmingdale, NY)

Type II secondary sonic booms are those that reflect from the primary boom carpet, enter the stratosphere, and turn back toward the ground if the upper atmospheric winds are favorable. A study initially described by the authors at the Forum Acusticum meeting in September 2023 predicted that the Type II secondary sonic boom rays could be profoundly affected by small angle changes in the topography of the ground. The current work provides an update on those ray tracing predictions, namely, that the amplitudes of Type II secondary sonic booms are substantially reduced by the topography. Such results may explain why secondary sonic booms historically have been reported for supersonic operations over water but not for over land. A grid refinement study will also be summarized. [Work supported by the FAA through ASCENT Project 57 under the supervision of Sandy Liu. 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.]

2:40

2pNSb3. Exercising prediction validation methods for analysis of quiet supersonic overflight. Alexandra Loubeau (NASA Langley Res. Ctr.1 NASA Dr. MS 463, Hampton, VA23681, a.loubeau@nasa.gov), Kathryn Ballard, William Doebler, and Nathan B. Cruze (NASA Langley Res. Ctr., Hampton, VA)

En route noise certification procedures for quiet supersonic aircraft are in development under the International Civil Aviation Organization for a proposed standard that could enable civil supersonic overland flight. While these quiet supersonic aircraft do not exist yet, methods are being developed and evaluated using existing N-wave empirical datasets and low-boom simulation datasets. One potential aspect of future certification methods is the ability to use a validated prediction model to predict the certification noise level under reference day weather conditions. Methods to validate predictions, such as the two-sample t-test, Fisher's combined probability test, two-sample Kolmogorov–Smirnov test, and equivalence testing, have been exercised and modified for this application. While some methods compare mean values, others compare values across the distribution, and a combination of methods can be used. The methods were applied to a subset of an N-wave dataset to demonstrate the applicability and identify potential areas for improvement. Atmospheric turbulence variability was included as a factor in these analyses through KZKFourier propagation simulations with parameters matching

actual flights and weather conditions. Results are presented for six noise metrics (PL, ASEL, BSEL, DSEL, ESEL, and ISBAP), and plans to expand the analyses are discussed.

3:00

2pNSb4. Air show primary sonic booms across a seismic network. Sampath Rathnayaka (Geosciences, Penn State Univ., University Park, PA), Andrew A. Nyblade (Geosciences, Penn State Univ., University Park, PA), Victor W. Sparrow (Grad. Prog. in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, vws1@psu.edu), and Kimberly Riegel (Phys., Farmingdale State College, Farmingdale, NY)

Cates and Sturtevant (2002) and others have shown that sonic booms can be recorded using existing seismic networks. To revive this capability for current research, data from permanent seismic stations were utilized to extract primary sonic boom signals from known supersonic flights. The sonic booms were produced on October 14, 2022 near Edward Air Force Base, CA, associated with an Air Show corresponding to the 75th anniversary of the first supersonic flight. The sonic booms were created by F-22, F-18, and F-15 aircraft. The experimental results show that primary sonic booms were measured at different times by seismic networks in Southern California over an area of not less than 10 000 km^2 ($\sim$3900 mi^2). Secondary booms did not occur during this event due to inadequate upper atmospheric winds. [Work supported by the FAA through ASCENT Project 57 under the supervision of Sandy Liu. 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.]

3:20–3:40 Break

3:40

2pNSb5. A procedure for obtaining forecast turbulence parameters in the atmospheric boundary layer for acoustic propagation. William Doebler (NASA Langley, 1 NASA Dr., MS 463, Hampton, VA 23681, william.j.doebler@nasa.gov) and Alexandra Loubeau (NASA Langley, Hampton, VA)

Propagation through turbulence causes a mean reduction in loudness of sonic booms and in some cases can cause significant fluctuations in loudness about that mean. Understanding these effects is important for accurately planning loudness levels during upcoming community noise tests with the X-59 aircraft as well as mitigating the risk for excessive loudness due to turbulence. Current methods for modeling acoustic propagation through turbulence require several atmospheric parameters including the atmospheric boundary layer (ABL) height, friction velocity, mixed-layer velocity scale, surface-layer temperature scale, as well as ambient pressure, temperature, and humidity within the ABL. Accurate forecasts of these parameters are needed for X-59 flight planning and may also be useful for planning future supersonic aircraft certification flight tests. This presentation showcases one method for obtaining the forecast turbulence and ambient atmospheric parameters from freely available Climate Forecast System Version 2 data. Forecast parameters are compared to measurements to assess accuracy and utility.

4:00

2pNSb6. Measured turbulence statistics from two NASA sonic boom test campaigns. Mark C. Anderson (Dept. of Phys. and Astronomy, Brigham Young Univ., ESC N201, Provo, UT 84602, mark.anderson@byu.net), Kent L. Gee, Kaylee Nyborg (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), and Alexandra Loubeau (Structural Acoust. Branch, NASA Langley Res. Ctr., Hampton, VA)

Turbulence in the lower atmosphere can have substantial effects on measured sonic boom waveforms on the ground. These turbulent distortions vary spatially and temporally and are best understood statistically. Simultaneous atmospheric and acoustic data were acquired as part of the NASA Sonic Booms in Atmospheric Turbulence (SonicBAT) and Carpet Determination in Entirety Measurements (CarpetDIEM) tests. Data from these tests, both conducted at NASA Armstrong Flight Research Center, enable a detailed study of atmospheric effects on sonic boom waveforms. The present study expands on previous work [Anderson *et al*., J. Acoust. Soc. Am. **152**, A127 (2022)] by analyzing more booms with additional statistical analyses including metric distributions, standard deviations, and confidence intervals. Results from CarpetDIEM indicate that the confidence interval widths of mean metric levels computed across a seven-microphone array spanning 122 m vary by 1-5 dB, depending on the loudness metric. The effects of peak clipping are also considered, and an analysis is conducted to determine how to best recover from peak clipping for different metrics. [Work supported by NASA Langley Research Center through the National Institute of Aerospace and Analytical Mechanics Associates.]

4:20

2pNSb7. Application of turbulence filters on PCBoom predictions for two NASA flight test campaigns. Kaylee Nyborg (Dept. of Phys. and Astronomy, Brigham Young Univ., N284 ESC, Provo, UT 84602, kaylee.nyborg@byu.net), J. T. Durrant, Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), William Doebler, and Alexandra Loubeau (NASA Langley Res. Ctr., Hampton, VA)

In preparation for the NASA X-59 community noise tests, the effects of turbulence on predictions of sonic boom levels are studied. This work builds on previous studies by including the effects of turbulence via finite impulse response filters implemented in the PCBoom prediction program. The resulting turbulized sonic boom predictions are compared to measured boom metric levels from previous NASA flight tests: Quiet Supersonic Flights 2018 and Carpet Determination in Entirety Measurements Phase I. To determine the factors driving differences between predictions and measured levels, least absolute shrinkage and selection operator (Lasso) regression is used to create a reduced-order model for each flight test campaign. [Work supported by NASA Langley Research Center through Analytical Mechanics Associates.]

4:40

2pNSb8. Progress update on inclusion of atmospheric profiling for sonic boom propagation through turbulence. Joshua L. Kapcsos (Grad. Prog. in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, jlk642@psu.edu) and Victor W. Sparrow (Grad. Prog. in Acoust., Penn State Univ., University Park, PA)

Research continues in the field of sonic boom propagation to aid international efforts in technical standards of civil supersonic aircraft, and simulation is necessary to validate the data produced by up-and-coming demonstration aircraft. Turbulence is important when considering how sonic booms propagate; their effects can be heard at the ground as demonstrated through the SonicBAT project. Current efforts are being made to improve the turbulent sonic boom propagation code KZKFourier to increase accuracy and robustness. This presentation will serve as progress update on including atmospheric profiling in KZKFourier. Previous versions did not have this feature. The inclusion of atmospheric profiling in KZKFourier will allow for quantities such as humidity, density, and temperature to vary within the atmospheric boundary layer. The presentation will discuss both the analytical inclusion of profiling on the Khokhlov–Zabolotskaya–Kuznetsov (KZK) equation methodology and its implementation in the C++ code. [Work supported by the FAA through ASCENT Project 57 under the supervision of Sandy Liu. 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.]

Contributed Paper

5:00

2pNSb9. Challenges in measuring and quantifying sonic booms from Falcon-9 booster landings. J. T. Durrant (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), Kaylee Nyborg (Dept. of Phys. and Astronomy, Brigham Young Univ., N284 ESC, Provo, UT 84602, kaylee.nyborg@byu.net), Mark C. Anderson, Kent L. Gee, Grant W. Hart (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), and Lucas K. Hall (Dept. of Biology, California State Univ. Bakersfield, Bakersfield, CA)

Sonic booms from Falcon-9 booster landings contribute to the overall noise of the vehicle, but obtaining high-fidelity acoustical measurements and robust metric calculations of each sonic boom pose several challenges. This paper discusses three such challenges: sonic boom metric variability from differences in vehicle trajectory and meteorology, poor low-frequency response of low-sensitivity microphones, and contamination of the sonic boom by the landing burn noise. To quantify the variability between measurements, boom metrics are compared across four separate launches from Vandenberg Space Force Base. Boom metrics calculated from microphone stations at the same location differ by as much as 10 dB from launch to launch. Next, a digital pole-shift filter helps correct the low-frequency response of low-sensitivity microphones near the pad. These filters are adjusted so the waveforms more closely match those from microphones with a superior low-frequency response. Last, to obtain a clean sonic boom waveform, tracking the noise source location and peak frequency may help to distinguish where the sonic boom ends and landing burn noise starts. However, due to similar frequency content and close spacing in time, separating the boom from the landing burn noise near the pad remains a challenge for some booms. [Work supported by USACE.]

5:20–5:40
Panel Discussion

Session 2pPA

Physical Acoustics, Architectural Acoustics and Engineering Acoustics: Acoustical Measurements and Sensors for Challenging Environments II

Cristian Pantea, Cochair
Los Alamos National Laboratory, PO Box 1663, MS D429, Los Alamos, NM 87545

Akira Nagakubo, Cochair
Engineering, Osaka Univ., M1-523, 2-1, Yamada-oka, Suita 565-0871, Japan

Chair's Introduction—12:55

Invited Papers

1:00

2pPA1. Integrating ultrasound transducers in polymer additive manufacturing technologies to fabricate engineered composite materials with tailored properties. Bart Raeymaekers (Mech. Eng., Virginia Tech, 635 Prices Fork Rd., 445 Goodwin Hall, Blacksburg, VA 24061, bart.raeymaekers@vt.edu)

Engineered composite materials that comprise filler material embedded in a polymer matrix may exhibit exotic physical properties that derive from the specific type, geometry, organization, and orientation of the particles in the matrix. However, existing techniques to manufacture such engineered materials are limited to laboratory scale, specific materials, and/or 2-D implementations, because it is challenging to organize and orient large quantities of particles into specific patterns. We utilize the acoustic radiation force associated with a standing ultrasound wave field to organize and orient particles of any material type dispersed in a fluid medium, into a user-specified pattern. We integrate the ultrasound wave field with several polymer additive manufacturing technologies to manufacture engineered materials in a layer-by-layer fashion, where in each layer we organize a user-specified pattern of filler using the acoustic radiation force. We demonstrate and discuss the integration of ultrasound transducers with several AM technologies and highlight their operating envelope and limitations. This research finds application in 3-D printing engineered materials with tailored properties by tuning the geometry, surface area, local packing density and, thus, the properties of the material.

1:20

2pPA2. Optical generation and detection of GHz longitudinal and transverse acoustic waves in solids assisted by two-dimensional metallic grating structure. Osamu Matsuda (Faculty of Eng., Hokkaido Univ., N13W8, Kita-ku, Sapporo, Hokkaido 060-8628, Japan, omatsuda@eng.hokudai.ac.jp)

Absorption of laser pulses with picosecond temporal width in a two-dimensional nano-scale metallic grating deposited on a transparent substrate launches both longitudinal and transverse acoustic waves in to the sample. The periodic metallic structure makes the generated acoustic waves propagate several different directions. The propagating acoustic waves are monitored optically by the delayed light pulses (probe light) as the transient reflectivity change which is known as the Brillouin oscillation. The probe light is also diffracted by the metallic grating, allowing the detection of the acoustic waves propagating in the multiple directions efficiently. The analysis of the Brillouin oscillation frequencies brings the longitudinal and transverse sound velocities and the optical refractive index simultaneously from a single measurement. We verify the above mentioned scheme using a sample with two-dimensional square lattice (period 380 nm) of aluminum square islands of lateral dimension 200×200 nm^2 and thickness of 50 nm on a fused silica substrate of thickness 1 mm. From the obtained Brillouin oscillation frequencies in several tens GHz, the acoustic and optical properties of the substrate material are successfully retrieved. The technique would be applicable to investigate the acoustic and optical properties of more complicated systems, such as anisotropic and/or inhomogeneous medium.

1:40

2pPA3. Machine learning for acoustic sensing in challenging environments. John Greenhall (Los Alamos National Lab., P.O. Box 1663, Los Alamos, NM 87545, jgreenhall@lanl.gov)

Acoustic measurements provide unique capabilities that make them attractive for a wide range of monitoring, characterization, and classification applications where other sensing techniques based on electromagnetic waves or radiography have physical limitations. One of the main challenges in real-world applications of acoustic sensing is the information-rich nature of the measurements. These measurements contain vast amounts of information about the system being investigated, but it can be difficult to extract the useful aspects of the data from the complex raw data. Thus, machine learning (ML) techniques have been gaining popularity rapidly in acoustics, by identifying complex correlations in the measurements that are not apparent from traditional techniques. In this presentation, I

focus on performing acoustic measurements in challenging environments, where data is highly complex and/or noisy. I discuss how ML can be applied to the main categories of acoustic data, i.e., time-series, frequency-spectra, and numeric features, and provide examples where each category is useful in a different challenging environment, such as in materials with thermal gradients, multi-component systems with complex vibrational coupling, and in fast-moving streams of flowing biomass.

2:00

2pPA4. Ultrasound pulse-echo measurements in a large-volume press at the High-Pressure Collaborative Access Team (HPCAT). Rostislav Hrubiak (X-ray Sci. Div., Argonne National Lab., 9700 S. Cass Ave., Bldg. 434E, Lemont, IL 60439, hrubiak@anl.gov), Blake T. Sturtevant (Dynamic Experiments Div., Los Alamos National Lab., Los Alamos, NM), Curtis Kenney-Benson, Arun Bommannavar, Eric Rod, Maddury Somayazulu (X-ray Sci. Div., Argonne National Lab., Lemont, IL), and Nenad Velisavljevic (Phys. Div., Lawrence Livermore National Lab., Lemont, IL)

The large-volume press at beamline 16-BM-B, at the High-Pressure Collaborative Access Team (HPCAT) facility at the Advanced Photon Source (APS), Argonne National Laboratory, offers a comprehensive platform for *in-situ* characterization of materials' atomic structure and bulk properties under high pressure and/or high-temperature (HP-T) conditions. The beamline is equipped with a Paris-Edinburgh type press with *in-situ* X-ray, electrical, optical, and ultrasound probes. The ultrasound pulse-echo technique, in conjunction with other techniques, such as energy-dispersive X-ray diffraction, X-ray radiography, and thermal/electric measurements, provides relevant data on material properties, including elastic constants, equation of state, and phase transitions, at HP-T conditions. The beamline is currently undergoing a major redesign to coincide with the APS upgrade. Among other developments, the ultrasound measurements at the upgraded beamline will be optimized with the addition of rotational X-ray tomography, as well as new capabilities for online data analysis. These advancements, together with the improved beamline optics, will improve accuracy and significantly reduce the time needed for data collection, allowing for faster experiments and higher throughput. The presentation will provide an overview of beamline 16-BM-B with a specific focus on acoustic measurements using the ultrasound pulse-echo technique and relevant applications. The beamline's expanded capabilities and scientific examples will be presented.

Contributed Papers

2:20

2pPA5. Low-frequency acoustic collimated beam. Cristian Pantea (Los Alamos National Lab., PO Box 1663, MS D429, Los Alamos, NM 87545, pantea@lanl.gov), Eric S. Davis (MPA-11, LANL, Los Alamos, NM), and John Greenhall (Los Alamos National Lab., Los Alamos, NM)

In recent years, we have developed a variety of unique acoustic sources that can generate a highly collimated acoustic beam (very low beam spread) of low frequency (10-250 kHz) without any side lobes. An acoustic beam with the above characteristics is extremely important and leads to deeper penetration, due to low frequency, and increased lateral resolution, due to collimation. These novel acoustic sources are based on frequency mixing in nonlinear medium, linear array superposition, generation of Bessel-like acoustic beams through radial modes of piezo disks, etc. These sources were used extensively for wellbore integrity monitoring and fracture detection on simulated boreholes in the laboratory. A few different approaches of generating a Low-Frequency Collimated Beam, along with potential applications will be discussed.

2:40

2pPA6. Resonant ultrasound spectroscopy of hybrid metal additive manufacturing. Jazmin Ley (Mech. and Mater. Eng., Univ. of Nebraska Lincoln, City Campus, W342 NH, Lincoln, NE 68588-0526, jley3@huskers.unl.edu), Cristian Pantea, John Greenhall (Los Alamos National Lab., Los Alamos, NM), and Joseph A. Turner (Mech. and Mater. Eng., Univ. of Nebraska-Lincoln, Lincoln, NE)

Additive manufacturing has been targeted as the next high-impact fabrication technique for parts and components. Hybrid metal additive manufacturing (AM) refers to the 3-D printed fabrication process involving secondary manufacturing processes or energy sources and multifunctional printing. Specific layers are altered within the build using additional processes (i.e., milling or peening) that are synergistic with the additive process. This combination alters the sample microstructure and can refine grains, increase dislocation density, or induce residual stresses. The effect of these hybrid layers is typically not confined within the layer alone but has a compounding effect on preceding layers. The goal is to control the changes in print parameters throughout the build to enhance component performance, but unique challenges remain for nondestructive validation of such samples. Traditional ultrasonic methods on hybrid-AM components have successfully mapped material variations with sufficient spatial resolution. However, the use of resonance ultrasound spectroscopy (RUS) for hybrid-AM is less developed. In this presentation, the use of RUS is described relative to the characterization of hybrid AM 316L stainless steel samples. The spatial organization of the hybrid samples affects the resonances relative to their mode shape. Computational models are used to quantify the impact of the hybrid processes.

3:00–3:20 Break

3:20

2pPA7. Spatio-temporal measurement of density distribution in oscillatory boundary layer using phase-shifting interferometer. Eita Shoji (Tohoku Univ., 6-6-04, Aramaki Aza Aoba, Aoba-ku, Sendai, Miyagi 980-8579, Japan, eita.shoji@tohoku.ac.jp), Anis Maddi (Le Mans Univ., Le Mans, France), and Tetsushi Biwa (Tohoku Univ., Sendai, Japan)

The objective of this study is to directly measure the acoustic density fluctuations of a gas confined between two parallel plates. For this purpose, we originally developed a phase-shifting interferometer, where a polarizing Mach–Zehnder interferometer was incorporated with a polarization camera. This enabled us to achieve a time resolution of 70 FPS in addition to the high-resolution measurement by the phase-shifting technique. The test beam passes through the two parallel plates of width 10 mm, where an acoustic field is generated using a loudspeaker that produces high acoustic pressures (up to 3000 Pa) at frequencies ranging from 0.5 to 5 Hz. In this work, we successfully measure the density fluctuations, which are otherwise difficult to obtain using conventional optical interferometers. Additionally, the experimental results are compared to the visco-thermal acoustic theory, where a good agreement was reached, affirming the effectiveness of this phase-shifting interferometer. Furthermore, using the proposed setup, we are able to measure and deduce the spatiotemporal variations of the pressure, density, and temperature without interfering with the acoustic field, which distinguishes it from measurement probes like thermocouples. This interferometer can be used to study more complex phenomena such as edge effects in thermoacoustic engines.

3:40

2pPA8. Acoustic attenuation coefficient measurement in extreme environments: Experimental characterization of uncertainty in multiple-reflection method. Suzi Liang (Univ. College London, Malet Pl. Eng. Bldg., Gower St., London, London WC1E 6BT, United Kingdom, SUZI.LIANG.21@UCL.AC.UK), Bradley E. Treeby, and Eleanor Martin (Univ. College London, London, London, United Kingdom)

Measuring acoustic properties in extreme environments, involving hazardous chemical samples or temperature above 50°C or below 0°C, is challenging yet crucial for understanding waveform interactions with materials, with applications in fields like non-destructive testing (NDT) and medical applications. Multiple-reflection methods (MRM), using a buffer-rod between transducer and sample, offer a solution. Yet, the impact of variables like contact force, buffer-rod size, and surface roughness on the accuracy and precision of MRM measurements remains unvalidated. To address this, we introduce a system and experiments to quantify the accuracy of measurements of acoustic attenuation coefficient. Results demonstrate the substantial impact of contact force on system accuracy and precision, reflected by a decrease in the measured attenuation coefficient at 0.5 MHz from 3.8 to 1.5 dB/cm with increasing contact force. Effects of buffer-rod width were less pronounced and supported the theory that a cylindrical buffer-rod with twice the radius of the transducer is sufficient. Interestingly, surface roughness, despite being over 150 times smaller than the wavelength of the acoustic signal, influences measurement accuracy. This study illuminates the crucial role of system factors on measurement accuracy and uncertainty, which will aid optimization of measurements for diverse practical applications.

4:00

2pPA9. Ultrasonic sensing system with thermophone and its applications. Takaaki Asada (Murata Manufacturing Co., Ltd., 2288, Ooshinohara, Yasu, Shiga 520-2393, Japan, asada@murata.com), Shinichi Sasaki (Murata Manufacturing Co., Ltd., Yasu, Japan), Yuma Watabe (Murata Manufacturing Co., Ltd., Yasu, Japan), Yasufumi Yamada (Hiroshima Univ., Hiroshima, Japan), Kanta Hasegawa (Doshisha Univ., Kyoto, Japan), Yuuka Sato (Doshisha Univ., Kyoto, Japan), and Shizuko Hiryu (Doshisha Univ., Kyoto, Japan)

In recent decades, thermophone has been studied as a novel electro-acoustic transducer since it has notable features, such as thin, light weight, acoustic purity, and wide frequency range. Though power consumption and resulting temperature rise are considered fatal problem, they are drastically reduced by limiting operation within ultrasonic domain and adopting the pulse current drive. Ultrasonic sensing system consists of dedicated thermophone as transmitter and commercially available MEMS microphone as receiver is presented to demonstrate typical airborne ultrasonic application, such as range finder or obstacle detection. Applying pulse compression technique, few orders of improvement in distance resolution and effective noise rejection have been achieved. Details of the sensing system and applications cases as listed below will be disclosed. 1. Multiple miniature autonomous vehicles equipped the obstacle sensing system in order to verify the bats jamming avoidance behavior. 2. Bio-inspired active sonar system onboarded to the drone and demonstrated the obstacle avoidance navigation in a wild environment. 3. Small step identification with the obstacle sensing system for electric wheelchair and automatic guided vehicle (AGV). 4. Ultrasonic micro-meter level displacement measurement system for non-contact cardiac pulse monitoring.

4:20

2pPA10. Ultrasonic transducer for high temperature thickness monitoring. Pierre Belanger (Ecole de technologie superieure, 1100 Rue Notre Dame O, Montreal, QC H3C 1K3, Canada, pierre.belanger@etsmtl.ca), Sevan Bouchy (Matrius Technologies, Montreal, QC, Canada), and Ricardo Zednik (Ecole de technologie superieure, Montreal, QC, Canada)

Refineries typically undergo 30–60 days of programmed shutdowns every 4 to 6 years to assess the integrity of infrastructure not accessible during service. So far, the literature demonstrated the feasibility of using ultrasonic probes for continuous long-term monitoring up to 350°C. However, when the temperature keeps rising, an air- or water-cooling system is required or a long delay line is used to move the probe away from the heat source. Providing a real time monitoring solution for the most critical components operating at high temperature would increase safety and reduce the maintenance burden. In this talk, an ultrasonic probe operating completely immersed inside a 600°C (1112°F) environment for extended periods of time is presented. The design of the transducer will be discussed. Its small footprint enables it to be mounted at several critical and difficult to access locations. In order to validate performances transducers were mounted on plates and pipes of different materials and thicknesses and the assembly was put inside a furnace. The results of long-term stability at 600°C, the consistency of the measurements over a temperature range from 20°C to 600°C, and the robustness during aggressive thermal cycling will be presented.

4:40

2pPA11. A single crystal Lamb wave sensing array. Eliza Baddiley (Defence Sci. and Technol. Group, 506 Lorimer St., Port Melbourne, Victoria 3207, Australia, elbaddiley@gmail.com), Scott Moss (Defence Sci. and Technol. Group, Port Melbourne, Victoria, Australia), Ben Vien (Monash Univ., Melbourne, Victoria, Australia), Jaslyn Gray, Nik Rajic, Cedric Rosalie, David Munk (Defence Sci. and Technol. Group, Port Melbourne, Victoria, Australia), Crispin Szydzik, Arnan Mitchell (R.M.I.T Univ., Melbourne, Victoria, Australia), and Wing Chiu (Monash Univ., Melbourne, Victoria, Australia)

This paper reports on a multi-element relaxor ferroelectric single crystal (RFSC) based acoustic emission (AE) sensor called LAMDA (*linear array for modal decomposition and analysis*). This sensor is being developed for acoustic-signature detection applications, including for AE-based damage detection on space-based and undersea platforms. RFSC [011] $Pb(In_{1/2}Nb_{1/2})O_3$-$Pb(Mg_{1/3}Nb_{2/3})O_3$-$PbTiO_3$ (or PIN-PMN-PT) was machined into 200 μm thick × 1 mm long × 4 mm wide elements, with 16 of these elements arranged on a flexible-polyimide-carrier to form a LAMDA. This sensor was bonded to a 1.6 mm thick aluminium plate with dimensions 600 mm x 600 mm. Acoustic emissions consisting of Lamb waves (e.g., A0, S0) were generated in the plate using both piezoceramic (Pz26) disk-actuators and pencil lead breaks (PLBs). LAMDA measurements and one-dimensional laser-vibrometry were undertaken on the plate and compared with two-dimensional multi-physics model predictions. The plane-strain model geometry comprised a rectangular-section 1.6 mm thick and 600 mm long with a LAMDA located centrally. An AE, being a 5.5-cycle Hann-windowed function with centre-frequencies 300–1000 kHz, was introduced 150 mm from the LAMDA via point loading perpendicular to the surface. The Lamb wave dispersion-curves generated from the model showed good agreement with the experimental curves determined from LAMDA and laser-vibrometry.

5:00

2pPA12. Using non-destructive evaluation to monitor the influence of processing parameters on the curing process of thermoset polymers. Gonzalo Seisdedos Rodriguez (Mech. and Mater. Eng., Florida Int. Univ., 10555 W Flagler St., Miami, FL 33174, gonzalo6d2@gmail.com), Edgar Viamontes, Eduardo Salazar (Mech. and Mater. Eng., Florida Int. Univ., Miami, FL), Cristian Pantea, Eric S. Davis, Tommy Rockward (Los Alamos National Lab., Los Alamos, NM), and Benjamin Boesl (Mech. and Mater. Eng., Florida Int. Univ., Miami, FL)

The processing parameters used to manufacture thermoset polymers highly impact their curing process and final properties. However, these parameters, including stoichiometry and temperature, are difficult to monitor and control. This work presents a novel method to non-destructively evaluate in real-time the curing kinetics, mechanical properties, and chemical structure of an epoxy resin due to variations in the processing parameters using ultrasonics and Fourier transform infrared spectroscopy (FTIR). Samples with a different curing agent to epoxy ratio and varying curing temperatures were manufactured and tested. Changes in chemical structure and longitudinal sound speeds were monitored during the resin's curing process, showing variations in the cure kinetics and final mechanical properties as a function of the processing parameters. The influence of these parameters on the glass transition temperature and thermal decomposition of the epoxy was also determined. Using ultrasonics and FTIR has the potential to non-destructively aid in the tailoring of polymeric materials by performing *in-situ* monitoring during their curing process. This method can also be implemented as quality control in both an in-field and manufacturing setting.

Session 2pPP

Psychological and Physiological Acoustics: Auditory Cognition in Interactive Virtual Environments II

Janina Fels, Cochair
Institute for Hearing Technology and Acoustics (IHTA), RWTH Aachen Univ., Kopernikusstr. 5, Aachen 52074, Germany

Joerg M. Buchholz, Cochair
Linguistics - Audiology, Macquarie Univ., Australian Hearing Hub, Level 1, 16 University Ave., 2109, Australia

Chair's Introduction—12:55

Contributed Papers

1:00

2pPP1. Visual speech benefit provided by realistic sentences in noise. Ronny Ibrahim (ECHO Lab., Macquarie Univ., 16 University Ave. Australian Hearing Hub, Macquarie Park, New South Wales 2109, Australia, ronny.ibrahim@mq.edu.au), Kelly Miles (ECHO Lab., Macquarie Univ., New South Wales, Australia), Ralph Peter Derleth (SONOVA AG, Staefa, Switzerland), and Joerg M. Buchholz (ECHO Lab., Macquarie Univ., Macquarie Park, New South Wales, Australia)

Understanding speech in background noise presents a significant challenge to listeners with hearing loss. Even though hearing aids are very successful in restoring listening in quieter conditions, current devices provide only limited benefit in noise. While seeing a talker can improve intelligibility, visual benefit has typically been quantified using rather unrealistic audio-visual speech materials. Here, we investigated visual benefit using realistic effortful speech from the ECO-SiN corpus. Audio-only, audio-visual, and visual-only sentences were presented in three simulated real-world environments at their realistic sound levels in a 3-D loudspeaker array equipped with integrated high-resolution video projectors. Ten young normal-hearing (NH) listeners as well as 18 older listeners with sensorineural hearing loss participated in the experiment with and without their own hearing aids. While the NH listeners rapidly hit ceiling, listeners with hearing loss struggled to understand the audio-only sentences in the louder background noise—even when aided. There was, however, demonstrable visual benefit of up to 16% in the audio-visual condition for both groups, which is significantly less than commonly reported. Future research will explore if the individual outcomes of the audio-visual ECO-SIN test better reflects real-world hearing experience and how far this is maintained when using virtual reality glasses.

1:20

2pPP2. Comparing the benefits of hearing aids and cochlear implants in real-world listening environments. Lisa Maggs (Linguist., Macquarie Univ., 16 University Ave., New South Wales 2019, Australia, lisa.maggs@mq.edu.au), Megan Gradden (Speech and Hearing Clinic, Macquarie Univ., New South Wales, Australia), Alan Kan (School of Eng., Macquarie Univ., Macquarie Park, New South Wales, Australia), Mridula Sharma (Linguist., Macquarie Univ., New South Wales, Australia), Zachary Smith (Cochlear, Sydney, New South Wales, Australia), Brett A. Swanson (R&D, Cochlear Ltd., Sydney, New South Wales, Australia), and Joerg M. Buchholz (Linguist., Macquarie Univ., Macquarie Park, New South Wales, Australia)

Comparing outcomes between patients with bilateral hearing aids (HAs), bilateral cochlear implants (CIs), and bimodal fittings (CI + HA) are difficult because of the variations in hearing performance between the different devices and patient groups. This may impact patient counselling, device selection, and hearing outcomes for listeners with more severe hearing loss. This study seeks to identify the factors impacting the listening abilities of adults fitted with bilateral HAs, bilateral CIs, and bimodal fittings in noisy environments by comparing outcomes from commonly used clinical tests and a new task that emphasises realistic listening in background noise. By identifying these limiting factors from a variety of tests, optimising hearing aid fittings, and comparing between device configuration groups, the study will help understand where hearing aid devices no longer meet satisfactory individual outcomes, and a cochlear implant may improve long-term performance. This approach pushes toward considering real-world outcomes through realistic test measures when informing clinical counsel and device implantation or configuration recommendations. As such, adults with hearing loss can receive more tailored advice that validates their daily listening concerns and clinicians are granted a better understanding of how to improve the quality of life for their clients.

1:40

2pPP3. Implementation of a virtual reality-based executive function improvement system for children with ADHD using natural soundscape stimuli. Donghyun Ahn (Dept. of Psychiatry, Hanyang Univ. Medical Ctr., Seoul, Republic of Korea), Ga Hyun Lee (Dept. of Early Childhood Education, Hanyang Univ., Wangsimni-ro, Seongdong-gu, Seoul 04763, Republic of Korea, sagegaga@gmail.com), Sungwon Roh (Dept. of Psychiatry, Hanyang Univ. Medical Ctr., Seoul, Republic of Korea), Hyowon Yoon, and Jin Yong Jeon (Architectural Eng., Hanyang Univ., Seoul, Republic of Korea)

The aim of this study is to propose a new therapeutic approach for attention-deficit/hyperactivity disorder (ADHD) by integrating the healing elements of natural soundscapes with a virtual reality system (cave automatic virtual environment, CAVE) designed to stimulate multiple senses and enhance the functionality of individuals with ADHD. The participants, children aged 6 to 12, were selected based on the severity of their ADHD symptoms and were divided into two groups: those diagnosed with ADHD and those without. The participants were invited to partake in a CAVE/VR-based ADHD functional enhancement program, with two 40-min sessions per week over a total of 12 sessions. Biometric, cognitive, executive, and behavioral responses were measured to assess the effectiveness of the system for improving ADHD functionality. The collected data will be analyzed to identify differences in functional changes due to ADHD. Based on these findings, we expect to contribute to the development of more effective and targeted ADHD treatments in the future by better understanding the specific impacts of VR-based therapeutic interventions on the symptoms of children with ADHD.

2:00

2pPP4. Impact of direct-to-reverberation ratio and interaural cross-correlation on externalization in binaural audio rendering. Taehun Choe (Elec. Eng., Korea Adv. Inst. of Sci. and Technol. (KAIST), 291, Daehak-ro, Yuseong-gu, Daejeon 34141, Republic of Korea, sirius314@kaist.ac.kr) and Jung-Woo Choi (Elec. Eng., Korea Adv. Inst. of Sci. and Technol. (KAIST), Daejeon, Republic of Korea)

To achieve realistic binaural reproduction of spatial audio in a headphone scenario, addressing inside-the-head-locatedness is crucial, which can be accomplished through appropriate externalization techniques. Two acoustic cues, the direct-to-reverberation ratio (DRR) and interaural cross-correlation (IACC), are known to influence perceived externalization. However, the precise relationship between these cues and their combined effect has not been extensively studied. In this investigation, we aim to explore the relative contribution of DRR and IACC to externalization by examining various combinations of these two acoustic cues. To this end, participants were exposed to various binaural room impulse responses (BRIRs) synthesized to produce three distinct levels of DRRs and IACCs for two different source directions. The listening test was conducted in an anechoic chamber with participants' eyes blindfolded to minimize the influence of room divergence effect. We also investigated the influence of head tracking by conducting experiments with and without head trackers. The correlation maps between externalization and the two acoustic cues were generated for each source direction. The study reveals that the contribution of the two acoustic cues differs with and without head tracking, and the joint contribution also varies depending on the source direction and amount of reflections included in the BRIR.

2:20–2:40 Break

2:40

2pPP5. Impression difference to vehicle interior sound between active and passive driving situation. Nawoki Okabayashi (Mech. Eng., Osaka Inst. of Technol., 5-16-1, Omiya, Asahi-ku, Osaka-shi, Osaka 535-8585, Japan, m1m22407@st.oit.ac.jp)

In near future, autonomous vehicles are expected to be more popular due to the technology development. This transition from active to passive driving condition may change the driver's impression to vehicle interior sound. And the interior sound improvement for autonomous vehicles should be done according to the difference. In this study, we investigated whether driver's impression to interior sound changes or not according to the driving situation through subjective evaluation test using a driving simulator. In the test, an original driving simulator was prepared to reproduce identical sounds and conditions at both active and passive driving conditions. As the presented sounds, sound pressure level (SPL) decreased or increased sound at low, mid, or high frequency bands were reproduced with the original sound. Participants drove the simulator under various driving conditions including acceleration and deceleration and evaluated discomfort about the sound. As the result, low or high frequency SPL decreased sounds were evaluated to be more discomfort at passive condition than that at active condition. Especially this tendency became significant at the acceleration and deceleration condition. From these results, high or low frequency noise reduction was found to be important in the future autonomous vehicle development.

3:00

2pPP6. Psychophysiological restoration effects through the experience of Gugak soundscape content. Haram Lee (Architectural Eng., Hanyang Univ., 222, Wangsimni-ro, Seongdong-gu, Seoul 04763, Republic of Korea, haramchi7@naver.com), Hyowon Yoon, Jin Yong Jeon (Architectural Eng., Hanyang Univ., Seoul, Republic of Korea), Donghyun Ahn (Dept. of Psychiatry, Hanyang Univ., Seoul, Republic of Korea), and June Sic Kim (Clinical Reserch Inst., Konkuk Univ. Medical Ctr., Seoul, Republic of Korea)

The psychophysiological restoration effects of content combining Korean natural soundscape with Gugak, traditional Korean music, were validated through a VR CAVE environment. Participants experienced the content of their choice while seated in a comfortable capsule-shaped chair over a total period of 6 weeks. Psychological responses, heart rate variability (HRV), and electroencephalogram (EEG) were measured before and after each experience, including 4 weeks after the last (10th week). Acute and chronic effects were confirmed through various correlations derived from the data related to physical parameters that depict the characteristics of the soundscape environment, subjective evaluation results of the soundscape, and physiological response measurements, as well as significant changes in psychophysiological parameters related to restoration.

3:20

2pPP7. The contributions of level and near-field head-related transfer functions cues on auditory distance perception in a free field. Jun Zhu (School of Phys. and Optoelectronics, South China Univ. of Technol., Bldg. 18, South China Univ. of Technol., Tianhe, GuangZhou, GuangZhou 510641, China, 913906883@qq.com), Bosun Xie, and Tong Zhao (School of Phys. and Optoelectronics, South China Univ. of Technol., Guangzhou, China)

In a free field, distance-dependent level (or loudness) and near-field HRTF (head-related transfer functions) are considered to be two primary cues for auditory distance perception. However, some recent experiments using virtual auditory display (VAD) have exhibited inconsistent or even controversial results, especially regarding with the contribution of near-field HRTF cues to distance perception. Possible reasons for the inconsistency may be that static VAD with coarse approximation of near-field HRTFs were used in these experiments, which cause poor perception of externalization and error in HRTF-related cues. To address this problem, a set of psychoacoustic experiment using dynamic VAD with accurate near-field HRTFs is conducted in present work. The perceived distances of virtual source at target distance less than 1.0 m and with or without level normalization are evaluated. The results indicate that HRTF cue alone enable distance perception in the lateral region. In contrast, HRTF cue alone is insufficient for distance perception in the front region. A collaboration of HRTF and level cue enhances distance perception in all directions. Therefore, the present experiment yields results similar to previous experiment using real source and validate the contribution of level and near-field HRTF cues to distance perception.

3:40

2pPP8. Low-frequency limit for binaural timing cues. Jiaxin Luo (Univ. of California, Irvine, Medical Sci. E, Health Sci. Rd., Irvine, CA 92617, jiaxinjasonluo@gmail.com) and Fan-Gang Zeng (Univ. of California, Irvine, Irvine, CA)

Sound localization uses both interaural time differences (ITDs) and interaural level differences (ILDs). The ITD cue has a high-frequency boundary at 1500 Hz but is assumed to be limited only by the lowest audible frequency. We tested whether there is a low-frequency boundary for the ITD cue. Nineteen young, normal-hearing subjects participated in four experiments, in which 500-ms, 70-phon pure tones with various binaural cues were presented as a function of frequency from 62.5 to 2000 Hzin octave steps. The 70-phon level assured that audibility did not confound performance. Experiment 1 served as a control, in which a 10-dB ILD cue produced ~90% correct lateralization at all frequencies. Experiment 2 measured lateralization accuracy with a 90-degree ITD cue, Experiment 3 measured the just-noticeable-difference from a 0-degree ITD standard, while Experiment 4 measured the binaural masking level difference in white noise. As expected, the listeners could not use the timingcue to perform these binaural tasks at 2000 Hz; Surprisingly, they showed significantly poorer performance at 62.5 Hz than mid frequencies. The band-pass characteristic result suggested a low-frequency boundary for binaural timing cues. This low boundary is not due to audibility but likely physical or physiological processes in the ear.

Session 2pSA

Structural Acoustics and Vibration, Engineering Acoustics and Physical Acoustics: Acoustic Metamaterials II

Christina Naify, Cochair
Applied Research Labs, UT Austin, 10000 Burnet Ave., Austin, TX 78758

Nathan Geib, Cochair
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Applied Research Laboratories and Walker, Dept. of Mech. Eng., The Univ. Texas at Austin, 10000 Burnet Rd., Austin, TX 78758

Chair's Introduction—1:15

Invited Paper

1:20

2pSA1. Vibroacoustic response of a cylindrical shell with a multilayered locally resonant coating. Cikai Lin (School of Mech. and Manufacturing Eng., Rm. 408, Ainsworth Bldg., UNSW Sydney, Sydney, New South Wales 2052, Australia, kailinaus@gmail.com), Gyani Shankar Sharma (Platforms Div., Defence Sci. and Technol., Sydney, New South Wales, Australia), Alex Skvortsov, Ian MacGillivray (Platforms Div., Defence Sci. and Technol., Melbourne, Victoria, Australia), and Nicole Kessissoglou (School of Mech. and Manufacturing Eng., UNSW Sydney, Kensington, New South Wales, Australia)

A multilayered viscoelastic material used as an external coating on a marine vessel for underwater noise control is presented. The multilayered coating is designed using regular distributions of vacuous cavities and hard steel scatterers that yield monopole and dipole scattering responses, respectively. The coating is modelled as an equivalent fluid in which the layers of inclusions are modelled as homogenized layers with some effective properties. The coating is externally applied to a submerged cylindrical shell. Fully coupled fluid–structure interaction between the coated shell and the surrounding water is taken into account. The material and geometric parameters of the inclusions are observed to have a significance influence on the radiated sound from the shell. Different combinations of the material distribution of the inclusions on the vibroacoustic response of the submerged shell are presented, showing that the local resonances of the inclusions can be tuned to reduce the radiated sound in a broad frequency range.

Contributed Papers

1:40

2pSA2. Tuning the acoustic characteristics of bottle plant garden. Preeti Gulia (Mech. Eng., National Inst. of Technol. Agartala, Tripura, India, Main Bldg., Dept. of Mech. Eng., NIT Agartala, Jirania, Tripura, Jirania, Tripura 799046, India, preetigulia19@gmail.com)

This work presents a finite element study to compute the acoustic performance of a garden that consists of periodically arranged bottle plants. The purpose of this work is to achieve low- and high-frequency noise reduction using plastic waste bottles. A harmonic excitation is given at one end, and transmission loss is calculated as a function of frequency between the two ends of the computational domain. The result shows that bottle plants produce a high-frequency wide band (1300–2100 Hz) around Bragg's frequency as well as a low-frequency sharp narrow band at the resonance frequency (∼305 Hz) of the bottle. To achieve a wider band gap at low frequencies, water level in the bottles is kept at different levels. The result showed that placing bottles with various water levels next to one another causes a bigger band gap at low frequencies (255–360 Hz). As a result, the low-frequency band gap can be tuned by changing the water level in the bottles. Plants' ability to absorb sound further dampens it at specific frequencies. Utilizing the plastic bottle to form such a garden has a major environmental impact and can function as a visually pleasing area while promoting calm and tranquilly.

2:00

2pSA3. 3-D printed membrane-type acoustic metamaterials with tailored performance. William Johnston (Mech. Eng. Mech., Michigan Technol. Univ., 1400 Townsend Dr., Houghton, MI 49931, wjohnsto@mtu.edu) and Bhisham Sharma (Mech. Eng. Mech., Michigan Technol. Univ., Houghton, MI)

While traditional membrane-type acoustic metamaterials act as effective lightweight sound insulation materials, their geometries can neither be easily customized nor incorporated into other acoustic designs in a single workflow. Here, we leverage our established techniques at 3-D printing thin membranes to create custom acoustic metamaterials with tailored performance that can be easily implemented into other acoustic designs without any post-processing. In this study, we present the workflow to additively

manufacture flexible membranes with added textural features using low-cost extrusion-based printing methods. Using a custom G-code, we print membranes with controlled mass, stiffness, and porosity while superimposing various surface pattern designs. Using a four-microphone normal-incidence impedance tube setup, we demonstrate the effect of each parameter on the membrane's sound transmission loss behavior and compare the results against traditional acoustic mass law.

2:20

2pSA4. Design Of aesthetic acoustic metamaterials window panel based on Sierpiński triangle for sound-silencing with free airflow. Sanjeet K. Singh (Dept. of Design, Indian Inst. of Technol. Kanpur, D214 Hall8, Kanpur, Uttarpradesh 208016, India, sanjeet@iitk.ac.in) and Shantanu Bhattacharya (Mech. Eng., Indian Inst. of Technol. Kanpur, Kanpur, Uttarpradesh, India)

Design of high-efficiency, low frequency (<1000Hz) soundproof window or wall absorber which is transparent to airflow is presented. Due to the massive rise in human population and modernization, environmental noise has significantly risen globally. Prolonged noise exposure can cause severe physiological and psychological symptoms like nausea, headaches, fatigue, and insomnia. There has been continuous growth in building construction and infrastructure like offices, bus stops, and airports due to urban population. Generally, a ventilated window is used for getting fresh air into the room, but at the same time, unwanted noise comes along. Researchers used traditional approaches like noise barrier mats in front of the window or designed the entire window using sound-absorbing materials. However, this solution is not aesthetically pleasing, and at the same time, it is heavy and not adequate for low-frequency noise shielding. To address this challenge, we design a transparent hexagonal panel based on Sierpiński fractal triangle which is aesthetically pleasing, demonstrates normal incident sound absorption coefficient more than 0.96 around 700 Hz and transmission loss around 23 dB, while maintaining air circulation through triangular cutout. Next, we present a concept of fabrication of large acoustic panel for large-scale applications which lead to suppressing the urban noise pollution

2:40

2pSA5. Engineering Helmholtz resonators for directional low-frequency sound attenuation. Roger Domingo-Roca (Univ. of Strathclyde, 99, George St., Glasgow G1 1RD, United Kingdom, roger.domingo-roca@strath.ac.uk), Andrew Feeney (Univ. of Glasgow, Glasgow, United Kingdom), James Windmill (Univ. of Strathclyde, Glasgow, United Kingdom), and Joseph Jackson (Sydney, New South Wales, Australia)

Controlling the absorption and diffusion of sound in the audible range constitutes an exciting field of research. Acoustic absorbers and diffusers perform extraordinarily well at high frequencies with sizes comparable to the wavelength of the working frequency. On the other hand, efficient low-frequency attenuators demand large volumes leading to unpractical sizes and high manufacturing costs. However, can the size of the resonator be reduced while also decreasing the working frequency? The answer is, counterintuitively, yes. This work investigates this phenomenon by studying a series of 3-D-printed monoatomic metamaterials based on membrane-coupled Helmholtz resonators. The results reveal that these systems, apart from significantly decreasing the bandgap frequency by around 36%, produce a cardioid directional response that contrasts with the omnidirectional response from traditional Helmholtz resonators. The obtained results suggest that by following this approach, low-frequency attenuation can be achieved via miniaturised devices while additionally providing them with a sense of directionality. This combination of features makes these sensors the perfect candidates for the next generation of acoustic hearing devices and acoustic attenuators, since the freedom that 3-D-printing provides allows for fine control of wave manipulation.

3:00–3:20 Break

3:20

2pSA6. Sound insulation design using membrane and locally resonant structure. JunYoung Jang (Mech. Eng., Pusan National Univ., 2, Busandaehak-ro 63beon-gil, Geumjeong-gu, Bldg. #303, Busan 46241, Republic of Korea, skyblue6465@gmail.com) and Kyungjun Song (Mech. Eng., Pusan National Univ., Busan, Republic of Korea)

In this study, we propose a design for sound insulation that utilizes a membrane and a locally resonant structure in tandem. A membrane serves as an excellent sound insulation structure in itself, but it can also be used to improve sound insulation performance by changing the dynamic characteristics of the sound insulation panel. In the case of the latter, there is an advantage of improving sound insulation performance in several bands without a large increase in weight. However, because the resonance band occurs at a relatively lower frequency than the anti-resonance band, it is difficult to block noise at frequencies below the anti-resonance band. To address this, we added a locally resonant structure that operates within the associated band. Because the anti-resonance caused by the locally resonant structure occurs at a lower frequency than the resonance, it can compensate for the deterioration band caused by the membrane-cavity. These effects have been verified through a sound transmission loss simulation of the unit structure at the normal and oblique incidence condition. Furthermore, the soundproofing bands of the product (plate and enclosure) were validated through SPL experiments.

3:40

2pSA7. Acoustic metasurface for perfect absorption using helmholtz resonators with non-uniformly partitioned cavities by membranes. Eunji Choi (Mech. Eng., Korea Adv. Inst. of Sci. and Technol., Yuseong-gu, D, 291, Daehak-ro, Daejeon 34141, Republic of Korea, cej9049@kaist.ac.kr) and Wonju Jeon (Mech. Eng., Korea Adv. Inst. of Sci. and Technol., Daejeon, Republic of Korea)

We propose an acoustic metasurface consisting of sub-wavelength Helmholtz resonators whose cavities are non-uniformly partitioned by membranes, aiming at perfect absorption of low-frequency sound with the aid of hybrid resonance. Compared to the existing unit cells based on Helmholtz resonators, the proposed ones with embedded membranes have lower resonance frequencies, which implies that it is possible to achieve perfect absorption of sound by using a thinner metasurface. This study provides a theoretical model for fast and accurate design considering the visco-thermal losses in narrow orifices and the higher-order modes of the membrane. By optimizing geometrical parameters and locations of the embedded membranes, we design the metasurface for perfect absorption based on the theoretical model. Experimental validation is performed in impedance tube via fabrication.

4:00

2pSA8. Reconfigurable acoustic metamaterial chamber for low-frequency noise mitigation. Heow Pueh Lee (Dept. of Mech. Eng., National Univ. of Singapore, 9 Eng. Dr. 1, Singapore 117575, Singapore, mpeleehp@nus.edu.sg), Zhenyu Chen, Yung Boon Chong, and Kian Meng Lim (Mech. Eng., National Univ. of Singapore, Singapore, Singapore)

In this study, a new reconfigurable sonic crystal metamaterial chamber is proposed for noise mitigation. Each element of the sonic crystal is made on non-circular cross-section with built-in Helmholtz resonators. Numerical and experimental investigation of a single unit cell and the whole acoustic metamaterial chamber highlights the reliability and feasibility of the proposed sonic-crystal based meta-structure. Highly efficient sound insertion loss can be captured with a simple and easily accessed resonator. By carefully designing rotation support in each unit cell and manufacturing a 3-D printed base with periodic holes, the assembled resonators can be easily rotated, and the finite acoustic metamaterial chamber can be tuned to

another configuration without the need of reprinting and redesigning the whole structure. By changing the configuration of the full chamber, the device can keep the insertion loss at a high level, while exhibiting a better ventilation ability. The reconfigurable, scalable, feasible, and controllable system can be easily applied to several practical scenarios like the silent zone in the city center, the ventilated noise isolate device for the machines, and the fully optically transparent ventilation sound-insulating structures.

4:20

2pSA9. Beyond-nearest-neighbour metamaterials. Gregory J. Chaplain (Phys., Univ. of Exeter, Stocker Rd., Exeter EX4 4QL, United Kingdom, g.j.chaplain@exeter.ac.uk), Dan Moore, Ian Hooper, Alastair Hibbins, John Sambles, and Timothy Starkey (Phys., Univ. of Exeter, Exeter, United Kingdom)

Engineering the dispersion of acoustic or elastic waves using coupling terms that spatially reach beyond the immediate local environment, or unit cell, is an "emerging topic" in Metamaterial design. In this talk, we present experimental studies in acoustic and elastic systems that realize beyond-nearest-neighbour coupling to introduce dispersion relations with extrema within the first Brilloun Zone. In acoustics, we use mixed waveguide-surface wave coupling, while in elasticity we develop an elastic scaffold (made from meccano) with reconfigurable coupling elements and demonstrate the effects of structural symmetries on these exotic dispersion relations. Applications to enhanced energy harvesting structures are presented by leveraging zero-group-velocity modes.

4:40

2pSA10. Acoustic cloning. Dirk-Jan van Manen (Geophys., ETH Zurich, Sonneggstrasse 5, Zürich 8092, Switzerland, dirkjan.vanmanen@erdw.ethz.ch), Jonas Müller (Geophys., ETH Zurich, Zurich, Switzerland), Theodor Becker, Xun Li, Johannes Aichele (Geophys., ETH Zurich, Zurich, Switzerland), Marc Serra-Garcia (Hypersmart Matter, AMOLF, Zurich, Switzerland), and Johan Robertsson (Geophys., ETH Zurich, Zurich, Switzerland)

Cloning refers to producing copies of objects that cannot be distinguished from the original based on their behavior or response. Here, we present a general methodology to clone objects that scatter acoustic waves and demonstrate it experimentally. We acquire digital twins and bring them back to life—a simple two-step process. First, we place the scattering object in a circular receiver aperture and insonify it from the outside using simple speakers. From the recorded data, which may be reverberative, we retrieve the object's scattering Green's functions for radiation conditions using a technique called multi-dimensional deconvolution. This process recovers the temporal and spatial bandwidth and removes the scattering interactions with the boundaries of the experimentation domain, if present. In the second step, the acoustic scatterer is removed and reconstructed holographically using the acquired scattering Green's functions. The hologram scatters arbitrary incident wavefields in real-time exactly like the original object. Low-latency feedback enables reproducing all orders of wave interactions between physical scatterers and the numerical hologram. The two-step process is demonstrated by cloning several rigid scatterers in a two-dimensional acoustic waveguide. Applications range from fully realistic digital scattering models to efficient meta-material experimentation. [Work supported by SNSF grant 197182.]

5:00

2pSA11. Noise reduction in an open structure by tailoring complex impedances on acoustic metasurfaces. Eunjin Yang (Korea Advanced Institute of Science and Technology, 291, Daehak-ro, Yuseong-gu, Daejeon 3414, Republic of Korea, yoasdf@kaist.ac.kr), Jiwan Kim, and Wonju Jeon (Korea Advanced Institute of Science and Technology, 291, Daehak-ro, Yuseong-gu, Daejeon 3414, Republic of Korea)

In this study, we design acoustic metasurfaces to the inner surfaces of the open structure to mitigate noise emanating through the opening by manipulating complex impedances along unit cells called impedance tiles. The impedance tile is comprised of subwavelength Helmholtz resonators, which have a theoretical model that accurately predicts the effective impedance, considering visco-thermal losses in narrow orifices. By adjusting the geometrical parameters of these resonators, the surface impedance of the impedance tiles can be finely tuned to minimize noise escaping from the open structure at any desired target frequency. To evaluate the noise reduction performance, the designed metasurfaces were fabricated via 3D printing, and the sound pressure levels (SPLs) were measured before and after applying the metasurfaces inside the open structure. The experimental results show that the $\lambda/31$-thick metasurfaces achieve SPL reduction exceeding 12.6 dB at the target frequency of 360 Hz.

Session 2pSC

Speech Communication: Speech Perception (Poster Session)

Justin Aronoff, Chair
Speech and Hearing Science, Univ. of Illinois at Urbana-Champaign, 901 S 6th St., Champaign, IL 61820

All posters will be on display from 1:00 p.m. to 5:00 p.m. To allow authors in this session to visit other posters, authors of odd-numbered abstracts will be at their posters from 1:00 p.m. to 3:00 p.m. and authors of even-numbered abstracts will be at their posters from 3:00 p.m. to 5:00 p.m.

Contributed Papers

2pSC1. When speaking clearly does not enhance comprehension: Comparing intelligibility of hard-of-hearing- and non-native-directed speech for native and non-native listeners. Nicholas Aoki (Lingustics, Univ. of California, Davis, 2100 5th St. (Apt 131), Davis, CA 95618, nbaoki@ucdavis.edu) and Georgia Zellou (Lingustics, Univ. of California, Davis, Davis, CA)

While clear speech is more intelligible than casual speech, some prior work indicates that the clear speech benefit is reduced for non-native listeners. It is unclear, however, how intelligibility for native and non-native listeners might differ across clear styles directed towards different imagined interlocutor types. If clear speech enhancements benefit the intended listeners, then (1) for non-native listeners, non-native-directed speech should be more intelligible than hard-of-hearing-directed speech; (2) native and non-native listeners should benefit equally from non-native-directed speech, while the advantage of hard-of-hearing-directed speech should be greater for native listeners. Native English speakers were recorded producing casual, hard-of-hearing, and non-native-directed speech to imagined interlocutors. Results from a speech-perception-in-noise task indicate that (1) native listeners had higher transcription accuracy than non-native listeners; (2) hard-of-hearing-directed speech was more intelligible than both non-native-directed and casual speech; (3) non-native-directed and casual speech did not differ in intelligibility; and (4) native and non-native listeners did not differ in the relative benefit of hard-of-hearing-directed speech. Contra our hypothesis, non-native-directed speech did not benefit non-native listeners despite being hyper-articulated compared to casual speech. This study highlights that speakers' expectations of what will be helpful for listeners do not always provide the intended benefit.

2pSC2. Identifying linguistic contributions to variable speech perception in Spanish/English bilingual children and adults using the ascending target-to-masker ratio method. Tiana Cowan (Boys Town National Res. Hospital, 555 N 30th St., Omaha, NE 68131, Tiana.cowan@boystown.org), Emily Buss (Otolaryngol./Head and Neck Surgery, Univ. of North Carolina, Chapel Hill, NC), Lori Leibold, Suresh Portillo (Boys Town National Res. Hospital, Omaha, NE), and Anne Olmstead (Commun. Sci. and Disord., Penn State Univ., University Park, PA)

When people encounter variable speech, like an unfamiliar accent, they map new productions to existing word forms. This skill appears to develop through adolescence for monolingual children, with school-age children perceiving unfamiliar accented speech less accurately than adults. For monolingual children, vocabulary size and phonemic awareness predict performance. Less is known about which factors account for individual differences in variable speech perception in multilingual school-age children. The goals of this study were to (a) assess group differences in variable speech perception in Spanish/English bilingual and English monolingual children and adults, (b) validate the ascending target-to-masker (TMR) method for assessing speech reception thresholds (SRTs) for foreign accented speech, and (c) identify linguistic skills that account for individual differences in performance. Participants completed speech perception and language testing. SRTs in speech-shaped noise were measured for Midland-, Spanish-, and Korean-accented English; these accents ranged from familiar to unfamiliar across participants. In each accent condition, one list of Bamford-Kowal-Bench test sentences was presented, with each sentence presented at multiple TMRs. Preliminary data suggest differences in how bilingual and monolingual groups perceive variable speech in noise. While linguistic skills accounted for individual differences in performance, different patterns emerged in children and adults.

2pSC3. Categorization and discrimination of Mandarin lexical tones by naïve English listeners. Yanping Li (the MARCS Inst., Western Sydney Univ., Locked Bag 1797, Penrith, New South Wales 2751, Australia, yanping.li@westernsydney.edu.au), Michael Tyler (Independent Researcher, Sydney, New South Wales, Australia), Denis Burnham, and Catherine Best (the MARCS Inst., Western Sydney Univ., Sydney, New South Wales, Australia)

Unlike tone languages such as Mandarin, English lacks tones at the sublexical level. Accordingly, English listeners have difficulty perceptually assimilating tones as categorized or uncategorized native segments (Perceptual Assimilation Model, PAM: Best, 1995). While English listeners can categorize the four lexical tones of Mandarin, i.e., level contour, rising, dipping, and falling, when given *question, statement, exclamation*, and *uncertainty* intonations as category choices (So & Best, 2011, 2014), this does not address tone assimilation at the segmental level. We reasoned that they might assimilate tones as non-assimilable nonspeech patterns if given visual icons as tone category choices (flat, rising, dipping, and falling lines, respectively), with no reference being made to English intonation categories. Accordingly, 76 monolingual English listeners ($M_{\text{age}} = 24.85$ years, 50 females) were set two tasks: to use visual icons to categorize Mandarin tones in naturally produced tone-words (/*ga*, *ti*, *tu*, *pu*/ $\times$ 4 tones) and to discriminate all six pairwise tone contrasts. All tone pairs showed ceiling-level discrimination, and listeners split their categorizations of falling and level stimuli between the falling and flat icons, suggesting that when given visual icons, tone-naïve English listeners perceive Mandarin tones as nonspeech acoustic patterns, which is consistent with PAM's non-assimilation predictions.

2pSC4. Amodal phonological abstraction in infants at 10 months. Eylem Altuntas (MARCS Inst. for Brain, Behaviour, and Development, Western Sydney Univ., 160 Hawkesbury Rd., Sydney, New South Wales 2145, Australia, eylem.altuntas@westernsydney.edu.au), Catherine Best (MARCS Inst. for Brain, Behaviour, and Development, Western Sydney Univ., Sydney, New South Wales, Australia), Marina Kalashnikova (Basque Ctr. on Cognition, Brain, and Lang., San Sebastian, Spain), Antonia Goetz (MARCS Inst. for Brain, Behaviour, and Development, Western Sydney Univ., Sydney, New South Wales, Australia), and Denis Burnham (MARCS Inst. for Brain, Behaviour, and Development, Western Sydney Univ., Sydney, New South Wales, Australia)

Infants can abstract amodal information about place of articulation at 4-5 months, before perceptual attunement to native consonants occurs (Altuntas *et al*., in preparation). We tested infants in the same task at 10 months, after native consonant attunement has commenced, which may affect phonological abstraction relative to 4–5 months. 29 infants acquiring Australian English were trained to associate different cartoon animals with audio-only words from two artificial mini-languages (Language A: Kangaroo; Language B: Kookaburra), which differed in use of only labial or only coronal sets of consonants. They were tested with video-only words (silent talking face) paired with Congruent (matching) or Incongruent (mismatching) animals. We hypothesized that infants would look longer at the Congruent than Incongruent pairings during the test trials, indicating generalization from the auditory (training) to the visual modality (test), i.e., amodal abstraction. Surprisingly, looking times did not differ between Congruent and Incongruent trials. Attunement to native consonants at 10 months may interfere with phonological abstraction of the place of articulation feature across different consonants that share the same place. Examining other phonological abstractions or comparing infants acquiring languages with different consonant inventories could shed light on how perceptual attunement affects amodal phonological abstraction.

2pSC5. Probing the phonological-coding deficit in dyslexia with vowel length perception. David J. Morris (Nordic Studies and Linguist, Univ. of Copenhagen, Emil Holms Kanal 2, Copenhagen DK-2300, Denmark, dmorris@hum.ku.dk) and Holger Juul (Nordic Studies and Linguist, Univ. of Copenhagen, Copenhagen, Denmark)

There is an established linkage between dyslexia and anomalous processing of speech sounds. We probed this in a Danish language context with vowel length tasks based on "kugle" *ball* /kuːlə/ and sequential deletions of the vowel portion to yield "kulde" *coldness* /kulə/. Vowel length continua are methodologically advantageous as they do not involve the perception of a sudden phonetic change that may instead tap other auditory processing abilities. Identification and discrimination tasks were administered to tertiary ($n = 28$), reading impaired ($n = 26$), and lower secondary students ($n = 20$), and the latter were approximately aged matched to the reading-impaired group. Identification functions derived from regression modelling of the responses showed that the dyslexics had significantly flatter curves than the other groups. Moreover, the secondary and dyslexic groups differed at the long vowel extremity of the continuum. Discrimination results showed that mean peak sensitivity of the tertiary students was higher than that of the secondary and dyslexic students. These results indicate that the phonological-coding deficit observed in dyslexics may be indexed by vowel length identification. Furthermore, identification results suggest that the nature of the phonological-coding deficit concomitant with dyslexia may stem from a lack of precision in processing the minimally modified longer vowel stimuli.

2pSC6. The lexical representation of coda /t/ by 14-month-olds. Ekaterina Khlystova (Linguist, UCLA, 3125 Campbell Hall, Los Angeles, CA 90095-1543, ekhlystova@g.ucla.edu) and Megha Sundara (Linguist, UCLA, Los Angeles, CA)

We evaluated 14-month-olds' lexical representation of coda /t/. Pronunciation variation of /t/, one of the most frequent segments in English, is well documented. Crucially, in American English infant-directed speech, /t/s are most often produced as glottal stops. This is particularly the case in word-final position, where words like cat and night are produced as [kæʔ] and [naiʔ] around 42% of the time, but produced with word-final canonical stop consonants (e.g., [kæt] or [nait]) only 25% of the time. In Experiment 1, we found that infants preferred to listen to familiar monosyllabic words ending in glottal stops (like [kæʔ]) when compared to nonce words ([kɪp]). This suggests that infants encode the most frequently occurring variant in their lexical entries. We are now testing whether the preference for familiar words produced with a glottal stop in Experiment 1 is the result of coda underspecification or encoding of glottal stop. If codas are underspecified in infant's lexical representations, we expect no preference between lists of familiar words with glottal stops ([kæʔ]), compared to minimal pairs with coda mispronunciations ([kæk]). We will discuss implications of these findings for infants' developing lexical representations and knowledge of allophony, focusing on the contribution of input frequency.

2pSC7. Correlates of word likeness in the perception of pseudowords. Benjamin V. Tucker (Commun. Sci. and Disord., Northern Arizona Univ., 4-32 Assiniboia Hall, Univ. AB, Edmonton, AB T6G 2E7, Canada, bvtucker@ualberta.ca) and Matthew C. Kelley (English, George Mason Univ., Seattle, WA)

New words are often introduced into languages from borrowings or neologisms. Pseudowords (possible words of a language that have not yet been assigned meaning) are used in many experiments to test aspects of speech perception. However, what we know about the perception of pseudowords is relatively limited. In this study, we investigate word likeness ratings of pseudowords on a 1–7 scale. A total of 659 listeners responded to sets of 200 pseudowords from an item list of 9599 English-sounding pseudowords varying in number of syllables and morphological complexity. We analyze the listeners' responses with the following predictors: phonotactic predictability, duration, phonological neighborhood density, phonological uniqueness point, pseudo morphological complexity, and acoustic distinctiveness. We analyze the ordinal data to identify how these predictors influence listeners' ratings. We also take average word likeness ratings of the pseudowords and use them to predict a different set of listeners' reaction times in a lexical decision task. In this presentation, we discuss what properties of the pseudowords influence what makes them seem more word-like or not and how word-likeness affects auditory word recognition.

2pSC8. Your brain on accents: Profiles of event related potentials in cross-dialectal listening in the US English context. Abby Walker (Virginia Tech, 181 Turner St. NW, Blacksburg, VA 24061, ajwalker@vt.edu), Holly Zaharchuk, Adriana Miller, and Janet Van Hell (Penn State, State College, PA)

Behavioral studies have established that cross-dialectal communication is typically harder than within-dialect communication: listeners make more mistakes and are slower to respond to less familiar accents. In this study, we use event-related potential (ERP) analysis of electroencephalography (EEG) to capture the neurocognitive correlates of these patterns. Speakers of Mainstream US English from Western Pennsylvania ($N = 23$) participated in an auditory go-no-go task where they heard both Southern- and Mainstream-accented US English real and nonsense words (and were asked to respond to real, animal words). We see effects of accent for real words on the P200 and N400, reflecting more effortful processing in both the acoustic-phonetic and lexical-semantic stages for Southern versus Mainstream accents. We see no differences in nonsense word processing, and together, these findings suggest that the difficulty in normalizing Southern-accented tokens at the acoustic-phonetic level disrupted lexico-semantic access later on. We are currently running the same study with Southern-accented listeners in Southwest Virginia, to see whether we see inverse results or whether listeners with substantial exposure to both dialects (which we expect to be true of our Southern listeners) show different response profiles.

2pSC9. Early prosodic marking of irony and its perception in autistic and neurotypical adults. Csilla Tatar (Linguist, Univ. of Michigan, 440 Lorch Hall, 611 Tappan St., Ann Arbor, MI 48109, cstatar@umich.edu), Jonathan Brennan, Jelena Krivokapić, and Ezra Keshet (Linguist, Univ. of Michigan, Ann Arbor, MI)

The present study examines the utterance-initial prosodic marking of irony and its perception in neurodiverse populations. We ask (i) whether

speakers and listeners use prosody to mark irony in the early, "pre-target" portion of an utterance (before a "target" word most closely associated with ironic intent), (ii) whether individuals vary in how they mark irony, and (iii) whether listener accuracy varies by speaker or autism condition. Eight American English speakers were recorded producing utterances presented in contexts conducive to either irony or sincerity. The "pre-target" parts were presented in a forced-choice experiment to 55 autistic and 45 neurotypical listeners and were examined for syllable duration and F0-related properties (maximum, minimum, range, mean, and wiggliness [Wehrle, Cangemi, Krüger, & Grice. (2018) Proceedings of AISV]). Results show that speakers mark irony in the "pre-target" region, with duration being the most salient feature, and wiggliness and F0-range being variable. Most listeners recognize irony from "pre-target" fragments, but there is variation in how well each speaker is perceived. Whether the listener is autistic or neurotypical does not predict accuracy. Results, thus, show utterance-initial prosodic marking to contribute to irony recognition, with the proviso that speaker and listener variation be taken into account.

2pSC10. Acoustic analysis of video conferencing platforms: Evaluating communication effectiveness for individuals with hearing loss. Arun Sebastian (National Acoust. Labs., Hearing Australia, 16 University Ave. Macquarie Park, Sydney, New South Wales 2109, Australia, Arun.Sebastian@nal.gov.au), Vicky Zhang, and Pádraig Kitterick (National Acoust. Labs., Sydney, New South Wales, Australia)

AIM: This research investigates the impact of video conferencing platforms on the transmitted sound to evaluate their effectiveness in facilitating communication for individuals with hearing loss. The preliminary analysis includes comparing the transmitted sound deviations from source recordings in Teams, Zoom, and Google Meet and exploring differences in audio quality across different settings within each platform (not focused on DSP limitations or audio transmission fidelity). METHODS: In this study, we employed babble noise, BKB speech, and a combination of babble noise and BKB speech (SNR: 9dB) as stimuli. We compared different acoustic features of the received signal, including automatic audio gain adjustments, frequency range, and frequency response, with the source sound. RESULTS: Zoom applies a smooth high-frequency roll-off for speech in a noisy signal, beginning at approximately 14.5 kHz and removing all frequency content above 18 kHz. The roll-off for Teams begins at a relatively low frequency around 7 kHz. Although Google Meet starts the roll-off around 20 kHz, the gain was inconsistent and had a sharp roll-off. CONCLUSION: Each platform exhibited distinct characteristics and introduced noticeable modifications to the stimuli. Results indicated that Zoom with optional audio settings enabled demonstrates the highest fidelity to dynamic range and frequency spectrum.

2pSC11. Investigating the effects of phonological neighborhood density, word frequency, and phoneme confusability on the English word recognition by native speakers of Japanese. Takeshi Nozawa (Ctr. for Lang. Education and Res., Ritsumeikan Univ., 1-1-1 Nojihigashi, Kusatsu, Shiga 5258577, Japan, t-nozawa@ec.ritsumei.ac.jp) and Ratree Wayland (Linguist, Univ. of Florida, Gainesville, FL)

This study aims to bridge the gap between previous research on the impact of phonological neighborhood density on word identification, which did not consider phoneme confusability, and studies on phoneme identification and discrimination, which did not account for the effects of phonological neighborhood density. Native Japanese listeners participated in this study, where they were tasked with identifying monosyllabic and disyllabic English words. The words varied in both phonological neighborhood density and word frequency and were produced by native speakers of American English and Japanese. Participants responded by typing the words they heard. The findings of this study indicated that there were no significant effects of phonological neighborhood density on word identification. However, frequent words were identified with greater accuracy compared to infrequent words. Moreover, words produced by native speakers of American English were more accurately identified than those produced by Japanese speakers. Japanese listeners' inaccurate perception of phonemes has resulted in word identification errors in both dense and sparse phonological neighborhoods. For example, they often mistakenly identify /l/ as /r/ and vice versa. Low and central vowels /æ, ɑ, ʌ/ are frequently misidentified as one another. These confusions are commonly observed among Japanese learners of English.

2pSC12. A preliminary investigation of the acoustic factors impacting decision making in speaker attribution. Debbie Loakes (School of Lang. and Linguisics, The Univ. of Melbourne, Babel Bldg., Parkville, Victoria 3010, Australia, dloakes@unimelb.edu.au), Helen Fraser (School of Lang. and Linguisics, The Univ. of Melbourne, Parkville, Victoria, Australia), and Kirsty McDougall (Theor. and Appl. Linguist, Faculty of Modern & Medieval Lang. & Linguist, Univ. of Cambridge, Cambridge, United Kingdom)

Attributing utterances to speakers in a good quality recording of clear speech might seem to be a simple and straightforward task, yet transcribers have been shown to "regularly and obliviously get it wrong" (Love 2020: 156). This issue has received very limited attention, and our research focuses on the issue from a forensic linguistic perspective where attributing an utterance to a speaker in a forensic case could effectively be accusing them of having committed a crime. Literature on speaker attribution/recognition never shows 100% accuracy even in very good acoustic conditions, with marked drops in performance when features are masked (as they often are in forensic situations). In this preliminary study, five phonetically trained listeners transcribed an audio file of six people engaged in a panel discussion from television (we focus on audio only). The speech of two female speakers is correctly attributed in almost all cases, so we pay attention to the acoustic features of four male voices. Listeners were almost never correct in their judgements about the exact number of speakers in the recording and made multiple attribution errors. We show that this is because of the fundamental nature of the speech acoustics; within-speaker variation is larger than between speaker variation for F0, intonational properties, and speech rhythm. Love, R. (2020). Overcoming challenges in corpus construction: The spoken British National Corpus 2020. *Routledge Advances in Corpus Linguistics*. New York: Routledge.

2pSC13. Auditory separation and perceptual advantage in team sports players: A behavioral study. Reethee Antony (Binghamton Univ., 4400 Vestal Parkway East, Vestal, NY 13850, rantony@gradcenter.cuny.edu), Stephanie Fazio, Isabel Falguera, Erica Scheinberg, Emma Schaedler (Misericordia Univ., Dallas, PA), and Sreedhanya Padappam Kandi (Rajiv Inst. of Speech and Hearing, Kozhikode, India)

There is a dearth of research related to perceptual advantages in noise, specifically in speech perceptual skills. In an initial study from this lab, the investigators observed that perceptual advantage was present in team sports players with higher percent correct scores relative to non-team sports players, specifically in noise. Research is required to understand if the perceptual advantage is present only for players engaged in specific team sports or not, hence the need for this study. The purpose of this study was to compare speech discrimination and identification skills in soccer players, other team sports players, and non-team sports persons. Thirty participants in the age range of 20-45 years were recruited: 10 soccer players, 10 other team sports players, (e.g., football), and 10 with no experience in team sports. Speech sounds /a/ and /s/ were presented in quiet and in the presence of background noise using speech babble at 0 signal-to-noise ratio. The procedure included a speech discrimination task using AX paradigm and speech identification using four closed-set response. Percent correct responses and reaction times were measured and analyzed using mixed model ANOVA. The findings from the study help towards applying speech in noise perception in sports rehabilitation.

2pSC14. Is secondary stress phonetically real for second-language learners? Evidence from Japanese-accented English. Kiyoko Yoneyama (Daito Bunka Univ., 1-9-1 Takashimadaira, Itabashi, Tokyo 1758571, Japan, yoneyama@ic.daito.ac.jp), Keiichi Tajima (Hosei Univ., Chiyoda-ku, Japan), and Mafuyu Kitahara (Sophia Univ., Chiyoda-ku, Cabo Verde)

Speakers of a non-stress language such as Japanese often have difficulty producing lexical stress distinctions in languages such as English.

Distinguishing primary-stressed versus unstressed syllables may be difficult, but distinctions involving secondary stress may be even more challenging. Some studies report that Japanese learners of English (JE) do not distinguish secondary-stressed and unstressed syllables in terms of vowel duration and quality whereas native American English speakers (AE) do. However, the data are limited and inconclusive. To resolve this issue, the present study asked AE and two groups of JE differing in English proficiency to produce 83 English words that exhibit stress shift, e.g., *phótogràph-phòtográphic-photógraphy* and conducted acoustic analysis of vowels in corresponding positions, which have the same consonantal context but differ in stress level. Preliminary results indicate that JE frequently produce devoiced vowels in unstressed syllables, particularly in phonetic environments similar to where devoicing occurs in Japanese. Devoicing, however, does not seem to occur in secondary-stressed syllables, suggesting that JE might distinguish secondary-stressed and unstressed syllables with respect to devoicing. Results from duration and formant analysis will also be reported and discussed in relation to theories of native-language influences on second-language speech learning. [Work supported by JSPS.]

2pSC15. Manipulation of affective speech and hemoglobin changes: A case study. Chorong Oh (School of Rehabilitation and Commun. Sci., Ohio Univ., Grover Ctr. W218, Athens, OH 45701, ohc@ohio.edu)

Despite the wide use of functional near-infrared spectroscopy (fNIRS) in various cognitive research, it is under-investigated whether fNIRS can offer neural evidence of emotion vocalization, especially in older adults. In this study, an 74-year-old male participant without a cognitive or physical difficulty completed two speech tasks while an fNIRS device recorded Changes in Oxy-hemoglobin (HbO) and total hemoglobin (HbT) levels in the pre-frontal cortex. The three speech tasks were (1) affective speech (i.e., talking about happy and sad life events) and (2) neutral speech (i.e., explaining how to make a peanut butter and jelly sandwich). The speech tasks were audio recorded which was acoustically analyzed using Praat for frequency and amplitude measures. In general, the frequency measures were highest when talking about sad life event while amplitude measures were highest when producing neutral speech. The HbO and HbT levels were positive and highest during the happy speech manipulation. During the other two speech tasks, the HbO levels were all negative with neutral being the lowest. The HbT levels showed a more complex change. More data are needed to confirm the pre-frontal lobe (de)activation associated with manipulation of affective prosody. This case study will facilitate future research in this line.

2pSC16. Is vowel devoicing in Japanese lexically specified?—An eye-tracking study. Mafuyu Kitahara (Faculty of Foreign Studies, Sophia Univ., Chiyoda-ku, Cabo Verde, mafuyu@sophia.ac.jp), Ayako S. Shirose (Faculty of Education, Tokyo Gakugei Univ., Koganei-shi, Tokyo, Japan), and Natsuya Yoshida (Faculty of Education, Tokyo Gakugei Univ., Koganei-shi, Japan)

In Tokyo Japanese, high vowels can be devoiced typically between voiceless consonants. Traditionally, devoiced vowels are treated as allophones of corresponding vowels in which the devoicing process is a result of phonetic implementation commonly affected by speech rate. An opposing view is that devoicing is lexically specified since there are cases of interactions between devoicing and compounding that can be naturally accounted for by lexical phonology. To investigate this issue, the present study conducted an eye-tracking experiment. Edited mismatch stimuli where vowels were voiced before a voiceless consonant (M1) or devoiced before a voiced consonant (M2) were auditorily presented with pictures of the target word, a competing word, and a distractor presented visually. This design is based on a prediction that M1 and M2 invoke confusion in the recognition process if devoicing is specified lexically. Sixteen participants were recruited for an experiment of about 200 trials. Results showed that only M1 stimuli invoked a different pattern of fixation rates than M2 and other natural stimuli. This can be interpreted as partial support for the traditional view of devoicing. The asymmetry between M1 and M2 calls for an account based on markedness theory which is independent of the lexical specification.

2pSC17. Visual cues in sound change: A cross-modal perceptual account for the typological rarity of labial palatalization. Jonathan Havenhill (Dept. of Linguist, The Univ. of Hong Kong, Pokfulam, Hong Kong) and Baichen Du (Dept. of Linguist, The Univ. of Hong Kong, Run Run Shaw Tower, Pokfulam Rd., Central and Western, Hong Kong, baichen@connect.hku.hk)

Auditory perceptibility is widely recognized as a major contributing factor to listener-based sound change. Although speech perception is also known to be influenced by vision, the role of visual cues in facilitating or inhibiting sound change is not well understood. Palatalization of /t^{j}/ and /k^{j}/ to /tʃ/ is a typologically common change that is frequently attributed to misperception, while full palatalization of labial consonants is exceedingly rare. We test whether visual cues contribute to this asymmetry by way of the visual distinctiveness of labial closure. An audiovisual perception experiment was conducted in which 40 native Korean speakers were presented with synthetically altered tokens along 5-step continua from /p^{h}/, /t^{h}/, or /k^{h}/ to /t?ç/ in audio-only, audiovisual-congruent, and audiovisual-incongruent conditions. Acoustic cues to place of articulation, F2 onset, and spectral frequency during aspiration, were manipulated. Results show that presence of the visual lip closure cue significantly inhibited perception of labials as palatals: the /p^{h}/-/tç^{h}/ boundary was shifted toward /tç^{h}/ in the AV-congruent group, and labials were more likely to be perceived as palatals in the absence of lip closure. This finding suggests that visual speech cues may contribute to the typological asymmetry and that sound change can be modulated by visual cues.

2pSC18. The effect of perceived human-likeness on voice-user interface–directed speech. Marcell Maitinsky, Maddy Walter, Amanda Cardoso, Md Jahurul Islam (Linguist, Univ. of BC, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, jahurul.islam741@gmail.com), and Bryan Gick (Linguist, Univ. of BC, Vancouver, BC, Canada)

Many interact daily with voice–user interfaces (VUIs), but acoustic research on VUI-directed speech (VDS) is relatively new. Prior work indicates intensity and F0 correlate with VDS [Cohn *et al.*, 2022, *JPhon 90*]. Multiple acoustic variables of VDS were analyzed to explore if VDS is a register distinct from human-directed speech (HDS) and whether perceived human-likeness of VUI voices affects VDS characteristics. 27 participants' Zoom recordings of 13 pre-scripted prompts and pre-recorded responses from two Amazon AWS-Polly-generated VUI voices (rated for human-likeness independently and by participants) were acoustically analyzed for word-initial voiceless plosives voice onset time (VOT), pitch variation, and vowel quality and quantity. Results of linear mixed-effects models indicate evidence of VDS-specific acoustic characteristics, some of which are affected by participants' perceived human-likeness of the voices. Differences in pre-exposure and VUI interactions occur for /p/ VOT in consonant clusters, vowel duration (except /ɪ/), and /ɑ/ F2. Statistical differences are found for /p/ VOT in consonant clusters and vowel quality (e.g., /ɑ/ F1 and F2, and /ɪ/ F2) based on perceived human-likeness by participants. This study contributes to the growing VDS work examining how humans speak with devices and what affects VDS, which may influence considerations of VUI voice development.

2pSC19. Children's listening effort for L2-accented speech. Ruth Altmiller (Psychol. and Brain Sci., Washington Univ. in St. Louis, 1 Brookings Dr., Campus Box 1125, St. Louis, MO 63130, ruth.altmiller@wustl.edu) and Kristin J. Van Engen (Psychol. and Brain Sci., Washington Univ. in St. Louis, St. Louis, MO)

Speech perception is a complex task that can become challenging in certain conditions, such as listening to speech in unfamiliar accents. Using task-evoked pupillary response (TEPR) and dual-task procedures, prior research has shown that adults recruit additional cognitive resources when processing fully intelligible L2-accented speech compared to speech in their own L1 accent (McLaughlin and Van Engen, 2020; Brown *et al.*, 2020). In the current study, we used pupillometry to investigate this effect in children ($n = 35$, ages 5;0 – 8;11) and ask whether prosodic variation can explain any differences in listening effort. Using Growth Curve Analysis, we found that unlike adults, children do not appear to recruit extra cognitive resources to

process intelligible L2-accented speech. Like adults, however, pupillary response declined more rapidly for L2-accented speech across the experiment than for L1-accented speech, indicating either rapid adaptation or fatigue. Additionally, we identified three prosodic measures (relative word duration, pitch stability, and pitch range) that were related to speech adaptation/fatigue for both accent conditions, but in opposite directions. Although children do not appear to upregulate listening effort for L2-accented speech like adults, they appear to use prosodic cues differently for L1- versus L2-accented speech as they adapt to speech.

2pSC20. Using L1 tone and intonation categories to investigate patterns in nonnative tone identification. Jessica L. Chin (The MARCS Inst. for Brain, Behaviour and Development, Western Sydney Univ., Locked Bag 1797, Penrith, New South Wales 2751, Australia, jessica.chin@westernsydney.edu.au) and Mark Antoniou (The MARCS Inst. for Brain, Behaviour and Development, Western Sydney Univ., Penrith, New South Wales, Australia)

Many languages utilise lexical tone, a feature which assigns meaning to words through variations in pitch contours. In some (but not all) cases, prior tone language experience can facilitate the learning of a nonnative tone language. Regardless, multiple sessions of tone training can improve nonnative tone perception for both tone and nontone L1 listeners. However, the influence of L1 tone and intonation categories on nonnative tone learning is underexplored. Across five sessions of tone training, listeners of nontonal English ($n = 25$), tonal Mandarin ($n = 23$), and tonal Vietnamese ($n = 25$) learned tones from an artificial language. While all groups improved in tone identification and tone word learning by session five ($p < .001$), Mandarin listeners outperformed the other groups overall (identification, $p < .001$; word learning, $p < .05$). Tone experience was a main effect only in word learning ($p < .05$). Tone- and language-specific identification patterns were also observed. The low level tone was easier for all listeners to perceive, while the tonal groups also showed greater accuracy for the mid level tone. Contrastingly, Mandarin listeners were hindered by their single L1 falling tone category when attempting to identify two distinct nonnative falling tones.

2pSC21. Dialect perception in song versus speech. Maddy Walter, Grace Bengtson, Marcell Maitinsky, Md Jahurul Islam (Linguist, The Univ. of BC, Vancouver, BC, Canada), and Bryan Gick (Dept. of Linguist, University of BC, Vancouver, BC, Canada, gick@mail.ubc.ca)

Investigations into style-shift in singing led to the proposal of genre-specific musical sociolects [Coupland, 2011, J. of Soc. 15]. Gibson [2019, Univ. of Canterbury Dissertation] demonstrated that style-shift in popular music is automatic. Additionally, Mageau *et al*. [2019, Phonetica 76] found non-native English speakers have a less perceptible foreign accent while singing than speaking. These findings provide possible behavioral analogues to the differential processing of speech and song. Gibson [2019, UofCanterbury Dissertation] accounts for this using Todd *et al*'s [2019, Cognition 185] exemplar theory, emphasizing the role of sung and spoken contexts in perception. We investigate the role of musical context in accent identification. 24 participants completed a dialect-identification task of 32 musical clips from two genres with strong sociocultural associations: country and reggae. 16 clips contained the original instrumental-removed vocals, and 16 different clips from the same songs additionally underwent monotonization and rhythm-normalization. Responses to the manipulated stimuli were more varied than the original vocals. Listeners' judgements may be more closely tied to country and reggae's sociocultural associations than speech information itself. These findings counter Mageau *et al*., illuminating a more complex relationship between accent perception, music, and genre. Future work will investigate this for dialect-nonspecific genres and adjust approaches to stimuli manipulation.

2pSC22. Cross-linguistic variability in audiovisual enhancement of coronal consonants. Jonathan Havenhill (Dept. of Linguist, Univ. of Hong Kong, Pokfulam, Hong Kong), Ming Liu (Dept. of Linguist, Univ. of Hong Kong, Pokfulam Rd. 109, Hong Kong 999077, Hong Kong, u3007723@connect.hku.hk), and Shuang Zheng (Dept. of Linguist, Univ. of Hong Kong, Pokfulam, Hong Kong)

The role of auditory perceptibility in phonological contrast is uncontroversial. In clear speech, speakers increase acoustic distance between contrastive phones, thereby improving perception. Although visual cues are also known to improve perception, it is unclear whether speakers actively enhance their speech for visual perceptibility. 20 native English and 20 native Mandarin speakers participated in a two-part production experiment. Words containing /l d n s r/ (and English /θ/) were elicited, with vowels balanced for height, backness, and rounding. Speakers first recited each word while alone in a soundproof booth. Words were then spoken for a native listener (who attempted to guess the word) over Zoom with heavily degraded audio. Audio and high-speed video were recorded. English speakers showed a significant increase in visually exaggerated articulations for /l, θ/ (but not /n, d, s, r/), including interdental and linguolabial productions with external lingual protrusion. Mandarin speakers increased their use of interdental articulation for /n, l, t, s/ (but not /r/), although lingual protrusion was limited. These results demonstrate that speakers may enhance their speech for visual over auditory perceptibility but suggest that enhancement is mediated by language-specific factors including phonological contrast and the articulatory characteristics of sounds in normal speech.

Session 2pSP

Signal Processing in Acoustics, Noise and Physical Acoustics: Signal Processing for Active Sound and Vibration Control II

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Invited Papers

1:00

2pSP1. A directional spatial active noise control system with a sound field separation algorithm. Huawei Zhang (Audio & Acoust. Signal Processing Group, Australian National Univ., 115 North Rd., Canberra, Australian Capital Territory, Australia, huawei.zhang@anu.edu.au), Jihui (Aimee) Zhang (Inst. of Sound and Vib. Res. (ISVR), Univ. of Southampton, Southampton, United Kingdom), Fei Ma (Audio & Acoust. Signal Processing Group, Australian National Univ., Canberra, Australian Capital Territory, Australia), Huiyuan Sun (School of Elec. and Information Eng., The Univ. of Sydney, Sydney, New South Wales, Australia), and Prasanga N. Samarasinghe (Audio & Acoust. Signal Processing Group, Australian National Univ., Canberra, Australian Capital Territory, Australia)

Spatial active noise control (ANC) systems aim to control noise fields within the entire region of interest. However, current spatial ANC systems commonly lack the spatial selectivity to target a specific noise field from the whole noise fields, which cannot satisfy user requirements in some cases. This paper designs a directional spatial ANC system to control noise fields selectively based on the direction of arrival (DOA). The key idea is to introduce a DOA-informed sound field separation (SFS) algorithm to design the directional spatial ANC system. In detail, the directional spatial ANC system is designed to control noise fields selectively, with the ability to select the target sound field introduced by the SFS algorithm. Simulation results demonstrate that the designed directional spatial ANC system can control noise fields selectively within the region of interest in a noisy environment.

1:20

2pSP2. Directional active noise control for drones. Hanwen Bi (School of Eng., The Australian National Univ., Canberra, Australian Capital Territory, Australia), Thushara Abhayapala (School of Eng., The Australian National Univ., 115 North Rd., Canberra, Australian Capital Territory 2601, Australia, thushara.abhayapala@anu.edu.au), Fei Ma, and Prasanga N. Samarasinghe (School of Eng., The Australian National Univ., Canberra, Australian Capital Territory, Australia)

Drones have become ubiquitous in various everyday applications, but their noise pollution remains an annoying issue. In this paper, we study the possibility of applying active noise control on drones with attached speakers. We propose the active noise control algorithms designed to iteratively calculate driving signals of speakers for canceling the outgoing noise from drones, specifically focusing on the blade passage frequencies within a directional spherical sector region. Two active noise control strategies are proposed: (i) minimizing the residual signal coefficients and (ii) minimizing the acoustic potential energy within the sector region. We derive update equations for two variables: (a) the loudspeaker weights and (b) secondary source spherical sector harmonics coefficients. Considering the power and loading limitations of drones, simulations are conducted under constraints of secondary source numbers to investigate the noise reduction performance with various secondary source setups. Simulation results demonstrate that the proposed method can effectively reduce the noise level over the directional sector region with a fast convergence speed.

1:40

2pSP3. Speech enhancement for helicopter headsets: Simulation and implementation on an FPGA platform. Johannes Timmermann (Helmut Schmidt Univ./Univ. of the Federal Armed Forces Hamburg, Holstenhofweg 85, Hamburg 22043, Germany, johannes.timmermann@hsu-hh.de), Florian Ernst, and Delf Sachau (Helmut Schmidt Univ./Univ. of the Federal Armed Forces Hamburg, Hamburg, Germany)

Clear and effective communication is crucial for the safe operation of aircraft. In helicopters, high levels of noise are generated by the engine, gears, and aerodynamics, which negatively impact speech intelligibility. To address this issue, modern aircraft headsets utilize active noise control (ANC) to reduce noise levels for both the crew and passengers. However, the speech signals captured by these headsets often contain high levels of background noise, thereby hindering internal and external flight communication. This paper introduces a dual microphone dual stage speech enhancement algorithm that combines basic spectral subtraction with a Wiener Filter, enhanced by the a priori and a posteriori signal-to-noise ratio. Audio data from within a helicopter cabin were recorded during a test flight. In a series of simulations, the Wiener Filter implementation is compared to other algorithms based only on spectral subtraction methods. The results are evaluated using established performance measures for speech quality. The Wiener Filter implementation results in the highest speech quality and is, therefore, implemented on an FPGA-platform for validation in a laboratory experiment. The simulations and measurements demonstrate significant improvements in speech quality and, consequently, enhance speech intelligibility using the proposed method.

2:00

2pSP4. Influence of background noise on spatial audio localization for helicopter pilot headsets. Florian Ernst (Helmut Schmidt Univ./Univ. of the Federal Armed Forces Hamburg, Holstenhofweg 85, Hamburg 22043, Germany, ernstf@hsu-hh.de) and Delf Sachau (Helmut Schmidt Univ./ Univ. of the Federal Armed Forces Hamburg, Hamburg, Germany)

A helicopter pilot is subjected to high workloads. A high noise level in the cockpit further increases stress. This stress can be reduced through headsets for hearing protection including active noise control (ANC). An actual research topic is the improvement of the human-machine-interface by 3-D-audio systems. These systems can improve pilot's situational awareness and, therefore, reduce stress. This paper presents a 3-D-audio system for headphones. This binaural system uses head-tracking to process spatial information about the location of the sound source. The spatial sensation is achieved by convolving the source signal with head-related transfer functions. This sensation is influenced by background noise, which can be reduced by ANC. A first listening test with *physical* sound sources provides information on the localizability of sound sources at different elevation angles. This is compared to a second listening test with *virtual* sound sources generated by the 3-D-audio system. In a third listening test, the *virtual* sound source is located at variable azimuth and elevation angles. The system is evaluated with respect to sound source localization and the influence of background noise and ANC. This study highlights the potential of the 3-D-audio system to improve the helicopter pilot's situational awareness in spatial acoustic environments.

2:20

2pSP5. Tonality masking system: Design and implementation of an industrial noise masking application. Tim Beresford (Acoust., Norman Disney & Young, Norman Disney & Young, Level 1, 29 Customs St. West, Auckland 1010, New Zealand, t.beresford@ndy.com) and Max Cyril (Acoust., Norman Disney & Young, Perth, Western Australia, Australia)

A noise source that exhibits tonal qualities is often deemed to be more annoying than one which is broadband in nature. In environmental noise assessments, it is common to apply a penalty for tonality (typically up to +5dB). Tonality is often objectively classified as a function of the spectrum shape (typically in one-third octave bands), where a prominent band relative to its adjacent bands indicates the presence of tonality. A potential means of removing this tonality is to introduce masking noise, altering the spectrum shape to cover up the peaks. By adding targeted masking, in principle, it is possible to not only remove the tonality classification and penalty but also minimise the increase in overall noise level to less than a few decibels. This paper discusses the practical implementation of a tonality masking system for a low-frequency industrial noise emission problem occurring at neighbouring residential properties. The masking system was designed to detect tones generated by a large piece of factory equipment and calculate the optimum masking signal using custom-made software. The masking signal was reproduced through a subwoofer located inside the factory. With the system successfully commissioned, the resultant noise levels at the residential neighbours were no longer classified as being tonal.

2:40–3:00 Break

3:00

2pSP6. Vibration damping by maximizing the reactive power of an inertial shaker. Tim Karl (Mechatronics, HSU, Holstenhofweg 85, Hamburg 22043, Germany, karlt@hsu-hh.de) and Delf Sachau (Mechatronics, HSU, Hamburg, Germany)

Passive and active vibration control is required to reduce noise, prevent damage, or maintain the stability of a structure. Active measures are of particular importance due to their narrowband performance. They use inverted or phase shifted signals to shift the kinetic energy of the vibrating structure to frequency bands where it does not have a negative effect on the sound emission and auditory sensation. In this paper, an active-mass-damper is realized by an inertial shaker as actuator by controlling its current. The power that the active-mass-damper transmits to the vibrating structure is measured by the transmitted force and the acceleration of the point of force application. The control system maximizes the reactive power and, therefore, acts as an active-mass-damper to minimize the vibration of a structure. An arbitrary structure represented by a single DOF mass-spring-damper system which is excited by a modal shaker. An electromechanical model of the inertial shaker is formulated, and a control system is designed. The simulation is validated with laboratory experiments. The experimental results show that the vibration of the structure can be minimized by increasing the reactive power transmitted by the active-mass-damper to the structure.

Invited Paper

3:20

2pSP7. Simulation of active vibration control of a moving stage. Zimu Guo (School of Mech. and Mechatronic Eng., Faculty of Eng. and IT, Univ. of Technol. Sydney, 32-34 Lord St., UTS Tech Lab, Botany, New South Wales 2019, Australia, zimu.guo@student.uts.edu.au), Sipei Zhao, Benjamin J. Halkon, and Lee Clemon (School of Mech. and Mechatronic Eng., Faculty of Eng. and IT, Univ. of Technol. Sydney, Ultimo, New South Wales, Australia)

Active vibration control (AVC) has gained considerable interest due to its inherent adaptability and cost-effectiveness, with its ability to suppress vibrations in the controlled system across various applications. Existing studies focus on the AVC of non-moving systems and usually assume that the vibration signal to be controlled is known and can serve as the ideal reference signal in feedforward control systems. However, in many practical applications, the ideal reference signal is not accessible, and a sensor must be used to detect the reference signal. This can degrade the AVC performance, especially for a moving system. This study, therefore, aims to explore the potential application of piezoelectric stack actuators (PSA) to control the vibration of a moving stage driven by a stepper motor. The secondary path was estimated offline, and vibration data were collected at different moving speeds. The effects of the location of the reference sensor were initially investigated. Then, extensive simulations were conducted to evaluate the performance of different adaptive control algorithms regarding vibration reduction, convergence speed, computational complexities, etc. This research underlines the potential of integrating PSA within a moving system to effectively control its vibration.

Contributed Papers

3:40

2pSP8. Spatial sampling versus acquisition time of room impulse responses for low-frequency sound zones. José Cadavid (Electron. Systems Dept., Aalborg Univ., Fredrik Bajers Vej 7B, Aalborg, Nordjylland 9220, Denmark, jmct@es.aau.dk), Martin B. Møller, Søren Bech (Bang & Olufsen a/s, Struer, Denmark), Toon van Waterschoot (Dept. of Elec. Eng. (ESAT), KU Leuven, Leuven, Belgium), and Jan Østergaard (Electron. Systems Dept., Aalborg Univ., Aalborg, Denmark)

Sound zone methods aim to create multiple listening areas within a room, allowing independent playback of different audio content. Achieving this requires a loudspeaker array, multiple microphones per sound zone, and a set of control filters, which design involves the room impulse responses (RIRs) between loudspeakers and microphones. Under certain acoustic conditions, acquiring these RIRs is possible using very short acquisition times. At lower frequencies, longer wavelengths enable spatial sampling of the sound zones with fewer points, reducing the number of microphone positions required and amount of data to process. This study exploits this characteristic to render and evaluate low-frequency sound zones in two real rooms, with few spatial sampling points. We assess the performance by objectively evaluating multiple spatial sampling configurations and RIR acquisition times. Furthermore, we evaluate the system's ability to control sound beyond the sampled regions by assessing the sound field over wider areas. For sound zones roughly the size of a human head, satisfactory performance is achieved in low reverberation conditions with at least two microphones per ear, even for RIR acquisition times as short as 150 μs. Conversely, the performance degradation in highly reflective environments cannot be compensated using additional microphones nor longer acquisition times.

4:00

2pSP9. Analysis of binaural sound pressure control and virtual source characteristics in stereophonic reproduction. wenjie Ding (Acoust. Lab., School of Phys. and Optoelectronics, South China Univ. of Technol., 301, Bldg. 18, South China Univ. of Technol., Guangzhou, Guangdong 510006, China, 18254275297@163.com) and Bosun Xie (Acoust. Lab., School of Phys. and Optoelectronics, South China Univ. of Technol., Guangzhou, China)

In stereophonic reproduction, the manipulation of relative magnitude and phase between two loudspeakers enables to create virtual sources within an azimuth region in frontal–horizontal plane. The law of sine, which is based on evaluating the interaural phase difference and a conventional method for stereophonic analysis at low frequencies, fails to explain some characteristic in stereophonic reproduction, including the influence of span angle of loudspeakers on virtual source localization and the stability of virtual source created by out-of-phase loudspeaker signals. Analogy to the method in active sound control, an alternative method for analyzing the characteristics of virtual source in stereophonic reproduction is presented in this work. On the basis of shadowless head model at low frequency, the power effort of loudspeaker signals for creating virtual source in stereophonic reproduction is evaluated. The results indicate that for frontal virtual source within the span of loudspeakers, the power effort of in-phase loudspeaker signals increases with the increase in the span angle of loudspeakers. For lateral virtual source outside the span of loudspeakers, the power effort of out-of-phase loudspeaker signals increases with the target azimuth of virtual source. Increase in power effort makes the control of binaural pressures difficulty and, thus, virtual source instable.

4:20

2pSP10. Enhancing generalization performance of anomaly detection-based active sonar classifier. Geunhwan Kim (Dept. of Ocean Systems Eng., Sejong Univ., 209, Neungdong-ro, Gwangjin-gu, Seoul 05006, Republic of Korea, kimgw200@sejong.ac.kr), YoungSang Hwang, Sungjin Shin (Dept. of Ocean Systems Eng., Sejong Univ., Seoul, Republic of Korea), and Youngmin Choo (Dept. of Defense System Eng., Sejong Univ., Seoul, Republic of Korea)

It is critical to automatically discriminate between target and clutter in developing future unmanned underwater surveillance systems. However, due to similar physical experiences between target echoes and clutter, active sonar classification remains a challenging problem. Recently, a novel anomaly detection-based active sonar classifier, bi-sphere anomaly detection (BiSAD), was proposed, demonstrating improved generalization performance over conventional supervised learning-based approaches. In the present study, we provide a brief overview of active sonar datasets and the BiSAD. Then, we explore modifications to BiSAD, aiming at further improving its generalization capabilities. Through examination using *in situ* active data, we validate the efficacy of these modifications, showing superior generalization performance compared to conventional supervised learning-based classifiers and the original BiSAD.

Session 2pUW

Underwater Acoustics, Acoustical Oceanography and Physical Acoustics: Effects of Shear Waves on Propagation and Scattering of Underwater Sound II

Oleg A. Godin, Cochair
Department of Physics, Naval Postgraduate School, 833 Dyer Road, Bldg. 232, Monterey, CA 93943

Alex Skvortsov, Cochair
Platforms Div., Defence Sci. and Technology Group, Melbourne, Australia

Contributed Papers

1:20

2pUW1. Estimation of seabed properties from ocean ambient noise in shallow water: A review. Md Ayub (Defence Sci. and Technol. Group, DSTG EDN Site, Edinburgh, South Australia 5111, Australia, md.ayub@defence.gov.au), Zhiyong Zhang (Defence Sci. and Technol. Group, Edinburgh, South Australia, Australia), and Daniel Boettger (Defence Sci. and Technol. Group, Edinburgh, South Australia, Australia)

Ocean ambient noise is the sound persisting in the ocean, which includes contributions from natural and anthropogenic sources, such as rain, wind, shipping, and marine mammals. Multiple interactions of the sound with the surface and bottom in shallow water affect the spatial characteristics of ambient noise. Thus, the spatial noise properties, coherence, and directionality of ambient noise depend on the characteristics of the shallow water environment, such as depth, sound speed profile, and seabed sediment properties. Acoustic properties of seafloor sediments can be extracted from ambient noise based on these noise properties using various types of noise data and geo-acoustic inversion methods. This paper presents a review of literature concerning investigations into available methods, the implementations of these methods, and their verification and validation. The efficacy and the applicability of these inversion methods for a wide range of ocean noise data are also examined in this review.

1:40

2pUW2. Correction factors for loss estimation of sound propagation over high shear speed seabeds. Matthew W. Koessler (Jasco Appl. Sci., 2305–4464 Markham St., Victoria, BC V8Z7X8, Canada, matthew.koessler@jasco.com), Craig McPherson (Jasco Appl. Sci., Capalaba, Queensland, Australia), and Alec J. Duncan (Ctr. for Marine Sci. and Technol., Curtin Univ., Perth, Western Australia, Australia)

Modelling range dependent sound propagation over layered elastic seafloors with high shear speeds has proved to be a difficult problem for many widely used under water acoustic sound propagation models. The geologic evolution of large portions of the Australian continental shelf result in it being a commonly occurring problem for sound propagation modelling. There has been success in developing numerical methods that have shown that it is possible to obtain accurate results for these types of range dependent environments; however, there may be issues with stability and efficiency. This article explores the appropriateness of correction factors to estimate the loss due to elastic seabeds with high shear speeds. The focus is on layered calcareous seafloors that are typical of the Otway shelf, located on the western flank of Bass Strait. Several methods for obtaining a dB correction that can be applied to a fluid only propagation model are presented and compared. The methods are validated against measurement data from an oil and gas campaigns in the Otway basin.

2:00

2pUW3. Manifestations of the weak shear rigidity of marine sediments in amplitudes and travel times of acoustic normal modes and lateral waves. Oleg A. Godin (Dept. of Phys., Naval Postgrad. School, 833 Dyer Rd., Bldg. 232, Monterey, CA 93943, oagodin@nps.edu)

Compressional-to-shear wave conversion at interfaces within unconsolidated marine sediments and interference of shear waves within thin sediment layers have been found to make significant contributions to sound attenuation [O. A. Godin, J. Acoust. Soc. Am. **149**, 3586–3598 (2021)] and produce unexpectedly strong perturbations in the normal mode group speed even when the shear speed is small compared to the compressional wave speed. This paper extends the previous analysis to the lateral waves and their applications to characterize the compressional wave speed and attenuation in the seabed. Shear wave-induced contributions to horizontal refraction of normal modes over horizontally inhomogeneous seabed are modeled to characterize the bearing and travel time errors that arise in the fluid-bottom approximation. The acoustic travel time and amplitude effects of the shear waves are found to accumulate at different rates with the propagation range depending on the stability of the shear wave interference within the seabed. The ways to distinguish between the manifestations in acoustic observables of the weak shear rigidity in the seabed and the other environmental uncertainties are discussed for range-independent, range-dependent, and horizontally inhomogeneous shallow-water waveguides. [Work supported by ONR.]

2:20

2pUW4. Shear and Scholte wave measurements on surficial intertidal mudflat sediments using medical sonoelastography techniques. Gabriel R. Venegas (Ctr. for Acoust. Res. and Education, Univ. of New Hampshire, 33 Academic Way, Rm. W137, Durham, NH 03824, g.venegas@unh.edu), Yu-hsuan Chao, Kang Kim (Dept. of Bioengineering, Univ. of Pittsburgh, Pittsburgh, PA), and John M. Cormack (Dept. of Medicine, Univ. of Pittsburgh, Pittsburgh, PA)

In low-energy environments, the water–sediment interface often contains a dynamic fluffy transition layer (FTL) of suspended organic and inorganic matter, which is transformed by complex physical, biological, and chemical processes. The FTL mechanical properties are important for understanding carbon cycle dynamics and for robust sediment acoustic characterization, but erodibility and size make it difficult to study non-invasively and at sufficient resolution. In medical ultrasound sonoelastography, shear waves are excited within the body either using the acoustic radiation force (ARF) or via external vibration at the skin surface. Sub-micrometer displacements are measured remotely by subsequent ultrasound imaging at up to 10 kHz frame rate with approximately 0.5 μm spatial resolution. In this

study, sediment cores were extracted from the intertidal mudflats from the Great Bay Estuary, USA. Compressional wave speed and attenuation measurements were performed using a core logger. Shear and Scholte waves were excited in the FTL using ARF- and external-vibration-based techniques and the motion detected with clinical ultrasound imaging arrays. Upward refraction of the shear wave was observed, and shear and Scholte wave speed and attenuation were measured. Finally, scanning electron microscopy was used to image the microstructure. Comparison between wave properties and microstructure will be discussed.

2:40

2pUW5. Sound pressure and particle motion calculations considering pile skin friction. Matthew W. Koessler (Jasco Appl. Sci., 2305–4464 Markham St., Victoria, BC V8Z7X8, Canada, matthew.koessler@jasco.com), David E. Hannay, and Alex MacGillivray (Jasco Appl. Sci., Victoria, BC, Canada)

Modelling sound propagation over seabeds where the seabed cannot be approximated as an equivalent fluid due to limestone layers with high shear wave speeds is a common underwater acoustic propagation modelling problem for Australian shallow marine environments. With the increase in prospective areas for offshore windfarm development across Australia, construction operations across large regions of the continental shelf area are set to increase. Windfarm construction will require the installation of a large number of foundation piles near potentially sensitive marine habitat areas. Predicting the underwater noise from these activities requires suitable models. This paper investigates the modelling approaches for including a full elastic parameterisation of the seabed when conducting numerical predictions of pile driving sound emissions. The approach considers buried sources and the excitation of shear waves via a skin friction model for the stress–strain relation at the pile-seabed boundary. Shear waves are considered in the source model and in the propagation model. Sound pressure level metrics and particle motion metrics are considered as part of the numerical predictions.

Invited Papers

3:00

2pUW6. Measurements of Scholte and Stoneley waves in the seabed. James H. Miller (Ocean Eng., Univ. of Rhode Island, Narragansett, RI, miller@uri.edu), Gopu R. Potty, Cecilia Schneider (Ocean Eng., Univ. of Rhode Island, Narragansett, RI), and Ying-Tsong Lin (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA)

As a part of the Seabed Characterization Experiment carried out in May 2022 in the New England Mud Patch, Scholte and Stoneley waves in the seabed were generated by the interface wave sediment profiler (iWaSP), a piezoelectric bender beam transducer. The iWaSP was deployed four times during in May. These waves were received at ranges of 40–100 m by bottom-mounted ocean bottom recorders (OBXs) each equipped with a three-axis geophone and hydrophone. Miller *et al.* (2023) found the speeds of these interface waves to range from 39 to 133 m/s in the first of the four deployments. This paper described the results of the following three deployments. Following the classification suggested by Wilson (P. Wilson, personal communication, July 15, 2023), the layers consist at least four distinguishable sediment layers. The upper-most is a relatively thin, fluid-like layer (1) of mud. Below that lies more rigid mud (2). Below the mud is a sand layer (3) with a thickness of 10 m. Finally, there lies a half-space layer (4) with a higher density and faster compressional and shear speeds. We conclude that the Scholte waves traveled on the 1–2 interface and the Stoneley waves propagated on the 2–3 interface. [Work supported by ONR Code 322OA.]

3:20–3:40 Break

3:40

2pUW7. Estimation of sediment compressional and shear properties in the New England Mud Patch using acoustic pressure and particle velocity data. Gopu R. Potty (Dept. of Ocean Eng., Univ. of Rhode Island, 215 South Ferry Rd., Narragansett, RI 02882, gpotty@uri.edu) and James H. Miller (Dept. of Ocean Eng., Univ. of Rhode Island, Narragansett, RI)

New England Mud Patch was the site of the Seabed Characterization Experiment in multiple years (2017, 2021, and 2022). This site is characterized by a layer of very fine grained surficial sediment layer over sand. Analysis of broadband data measured on the ocean bottom recorders (OBXs) and hydrophones will be discussed. OBX is a geophone–hydrophone system which measures three components (two orthogonal horizontal components and a vertical component) in addition to acoustic pressure. Five OBXs were deployed on the seabed and four hydrophones were configured as a tetrahedral array on the bottom mounted Geosled. Data from these sensors will be analyzed and different arrivals corresponding to compressional, shear and interface wave (Scholte and love waves) presence will be examined. Inversion schemes will be implemented to estimate the compressional and shear wave speeds and attenuation. A priori information from seismic surveys and sediment cores will be incorporated into the inversion schemes. Outputs of these inversions will be compared with inversion results from other investigators based on different inversion techniques and using data from the same location. [Work Supported by Office of Naval Research.]

4:00

2pUW8. Vector acoustic observations of ship noise and influence of sediment compressional speed gradients and shear rigidity in modeling. Peter H. Dahl (Appl. Phys. Lab. and Mech. Eng., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, dahl@apl.washington.edu) and David R. Dall'Osto (Acoust., Appl. Phys. Lab. at Univ. of Washington, Seattle, WA)

For acoustic measurements made within the water column, it is well understood that observability of the effects of seabed shear rigidity varies widely depending, e.g., on source type, frequency range and measurement geometry. Measurements of underwater noise from a passing ship made with a vector sensor are discussed in this context. Data are expressed in a bounded, non-dimensional form known as circularity that provides a signal independent of the ship noise-source spectrum. The system was deployed 1.45 m above the

seabed in waters of depth 77 m on the New England Mud Patch as part of the May 2022 Sediment Characterization Experiment. Broadband noise measurements (15i–500 Hz) of the ship are made during which the ship range varies from 10 to 1.5 km at closest point of approach. Observations are compared to an interpretative model based on the normal modes and a geoacoustic model consistent with studies conducted at this location; the defining characteristic being a low-speed mud layer constituting the upper ~10 m of the seabed that transitions with increasing sand content. Emphasis in this presentation is on parameterization of the deeper seabed structure involving compressional sound speed gradients and sediment shear rigidity.

Contributed Papers

4:20

2pUW9. Impacts of seafloor characteristics on sound propagation in a seamount environment. Brendan King (Ocean Eng., Univ. of Rhode Island, 215 South Ferry Rd., Narragansett, RI 02882, bking20@uri.edu), James H. Miller, Gopu R. Potty, Jade F. Lopez Case (Ocean Eng., Univ. of Rhode Island, Narragansett, RI), Tzu-Ting Chen (Scripps Inst. of Oceanogr., La Jolla, CA), and YT Lin (Scripps Inst. of Oceanogr., La Jolla, CA)

The New England Seamounts are located off the coast of Cape Cod. The Atlantis II Seamount, 400 miles southeast of Woods Hole, was the site of the recent New England Seamount Acoustics (NESMA) experiment during April–June of 2023. This seamount and others nearby contain complicated bathymetry with the Atlantis II peak rising roughly 3300 m from the seabed and 1700 m below the sea surface. During the experiment, broadband SUS charges were deployed in a 15 km radius around the seamount. A set of three bottom-mounted vector sensors recorded acoustic pressure and particle velocities. Data from an AUV-mounted sub-bottom profiler provided information about the sediment layers, including a sand layer atop limestone, overlying basalt. Using the RAM parabolic equation model, broadband arrivals were simulated. In addition, the Bellhop raytracing model was used to identify the arrivals. The measured data was compared to the synthetic time series produced by the models to understand the effects of the complex environment on acoustic propagation in the vicinity of the Atlantis II Seamount. [Work supported by Office of Naval Research Code 322OA and TFO.]

4:40

2pUW10. Effect of a low sound speed sediment layer on seismo-acoustic propagation in the New England Mud Patch. Jade F. Lopez Case (Ocean Eng., Univ. of Rhode Island, 215 South Ferry Rd., Narragansett, RI 02882, jadelcase@uri.edu), James H. Miller, Gopu R. Potty (Ocean Eng., Univ. of Rhode Island, Narragansett, RI), Andrew R. McNeese (Appl. Res. Lab, Univ. of Texas at Austin, Austin, TX), Preston S. Wilson (Dept. of Mech. Eng., Univ. of Texas at Austin, Austin, TX), and David P. Knobles (Sci. and Res. Corp., Austin, TX)

The New England Mud Patch (NEMP), located off the coast of Massachusetts, comprises at least four distinguishable sediment layers. The uppermost is a relatively thin, fluid-like layer (1) of mud. Below that lies more rigid mud (2) that has varying physical properties with depth. Below the mud is a sand layer (3) with a thickness of approximately 10 m. Finally, there lies a half-space layer (4) with a significantly higher density and faster compressional and shear speeds. In this paper, we explore the effect of the varying rigidity within the mud layer (2) and the sharp transition at the mud–sand interface, on forward-scattering. In the 2022 Seabed Characterization Experiment (SBCEX22), two different types of broadband sources (SUS and Rupture Disk sources) were deployed. The received signals show evidence of acoustic scattering and the generation of interface waves, along the interface between (1) and (2) and the mud-sand interface between (2) and (3). Results from normal mode modeling using Kraken and seismo-acoustic modeling using OASES suggest that the scattering can be attributed to inhomogeneities in the sediment layers. [Work supported by the Office of Naval Research Code 322OA.]

5:00

2pUW11. *In situ* measurements of sediment shear wave speed from the New England Mud Patch and shelf break areas using the acoustic coring system. Dante D. Garcia (Walker Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, dantegarcia@utexas.edu), Preston S. Wilson (Walker Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Megan S. Ballard, Kevin M. Lee (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Jason D. Chaytor (U.S. Geological Survey, USGS Woods Hole Coastal and Marine Sci. Ctr., Woods Hole, MA)

The acoustic coring system (ACS) is a probe-equipped gravity corer that provides *in situ* measurements of compressional and shear wave speed and attenuation. During an April 2022 coring survey, the ACS was deployed at 36 locations within the New England Mud Patch and New England Shelf Break areas. Data from these measurements will be presented to characterize the depth-dependent structure of the geoacoustic seabed properties as well as their spatial variability. The *in situ* measurements will be interpreted in the context of stratigraphic layering measured by a seismic survey. Historically, *in situ* shear speed measurements of the seabed have proved difficult to obtain. From the 2022 survey, depth-dependent (up to 4 meters) shear speed profiles at 800 and 1200 Hz were produced from a subset of the 36 deployments. These results, along with methods for analysis, challenges in the measurements, and supplemental laboratory experiments will be discussed. The results will also be compared to measurements from previous experiments. [Work sponsored by ONR.]

5:20

2pUW12. Features and limitations of the Weston α parameter as a descriptor of seafloor bottom loss. Adrian D. Jones (Ocean Acoust. Assoc., P.O. Box 333, Edinburgh, South Australia 5111, Australia, bearjones3@bigpond.com)

It is straightforward to prove that, for very small seafloor grazing angles, the Weston α parameter [Weston, J. Sound Vib. **18**, 271–287, 1971] provides a reasonable estimate of the slope of bottom loss in dB versus grazing angle. However, for some circumstances, the descriptions of bottom loss versus grazing angle based on the Weston α parameter are valid over a considerable span of grazing angles and to large loss values. By studying the seafloor reflection in terms of impedance, and as linked to the geoacoustic parameters, greater knowledge of the features and limitations of the Weston α parameter as a descriptor of seafloor bottom loss has been obtained. This paper covers the theory, applicable to a variety of seafloor types, each of a uniform half-space. Seafloors considered include those with either fast or slow compressional speed. For fast seafloors, shear speed is considered as one of less than, near to, or greater than the water sound speed. The effects of each of compressional and shear absorption in the seafloor are considered. Numerous sample plots of bottom loss versus grazing angle are used to demonstrate the accuracy of the analysis.

Session 3aAA

Architectural Acoustics and Signal Processing in Acoustics: Audio for Architectural Acoustics, Indoors and/or Outdoors I

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

Alexander Case, Cochair
Univ. of Massachusetts, Lowell, 25 Wilder Street, Lowell, MA 01854

Chair's Introduction—7:55

Invited Papers

8:00

3aAA1. Greater artistic and technological performance through the converged technologies of architectural acoustics, electroacoustic enhancement, and immersive audio technologies. Tim Boot (L-Acoust., 2645 Townsgate Rd., Ste. 600, Westlake, CA 91361, tim.boot@l-acoustics.com), Frederic Roskam, Phil Coleman, Simon Brown (L-Acoust., London, United Kingdom), and Julien Laval (L-Acoust., Marcoussis, France)

Producing meaningful, creative, and engaging experiences for audiences requires highly integrated acoustics and audio technologies. The fields of architectural acoustics, electro acoustic enhancement and immersive audio technologies have converged, altering the way we approach entertainment venue design. Globally, demand has increased for immersive audio in live and reproduced sound, and electro acoustic enhancement technologies are becoming more widespread. We will discuss how architecturally integrated end devices, multifunction signal processing, and architectural acoustical design should be optimized to create next-generation performance venues. We will use a project example that integrates sound system, acoustic enhancement, and architectural acoustics, illustrating how increasing clarity in audience with acoustic enhancement and supporting the immersive audio increases audience engagement. We will show what can be improved to elevate the experience in a converged approach. The paper will illustrate how a convergence of audio technologies, infrastructure, and a unified design and engineering processes is creating far greater opportunities for artistic experimentation and expression.

8:20

3aAA2. Cautionary tales at the intersection of audio and acoustics. David A. Conant (McKay Conant Hoover Inc., Westlake Village, CA) and K. Anthony Hoover (McKay Conant Hoover Inc, 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362, thoover@mchinc.com)

Sound as perceived by audiences is necessarily influenced by both the direct-arrival audio and the acoustical conditions in the space. Influences of audio and architecture upon each other can sometimes confound intended results. This presentation draws upon examples from performing arts, education, and worship spaces for interesting examples that underscore the importance of steadfast diligence at the intersection of audio and acoustics.

8:40

3aAA3. Even better musicians: Acoustic enhancement in the music classroom. Matthew Hildebrand (Wenger Corp., 555 Park Dr., Owatonna, MN 55060, matt.hildebrand@wengercorp.com)

Music educators are using electronic sound reproduction and acoustic enhancement in primary and secondary schools – and are producing better musicians as a result. The development of musical ability is inherently tied to room acoustics; a supportive variable acoustic environment accelerates the ear training and critical listening skills to proficiently play or sing in an ensemble. This presentation will review findings from the past 20 years spent with these technologies in music education. How might sound reproduction be optimized for learning clarinet? How can acoustic enhancement reduce the potential for hearing loss and other playing injuries? Is a low ceiling ever beneficial in a music rehearsal space? (Hint: no – but we'll attempt to address that, too).

9:00

3aAA4. Sound environment control using an immersive audio system—Advancing sound experience for creators and listeners using Active Field Control. Dai Hashimoto (Spatial Audio Group, Yamaha Corpration, 10-1, Nakazawa-cho, Naka-ku, Hamamatsu, Shizuoka 4308650, Japan, dai.hashimoto@music.yamaha.com), Hideo Miyazaki (Spatial Audio Group, Yamaha Corpration, Hamamatsu, Japan), Ron Bakker (Yamaha Music Europe GmbH, Rellingen, Germany), and Sungyoung Kim (Korea Adv. Inst. of Sci. and Technol., Daejeon, South Korea)

Spatial and immersive systems have gained popularity in recent years and are broadly used in various types of applications to realize natural or augmented sound environment. To create a variety of sound environment freely and flexibly, Yamaha immersive audio system, known as Active Filed Control (AFC), has two functions. One is "sound source control" based on object-based audio rendering technique and another is "sound field control" based on acoustic enhancement technology. This paper presents several unique solutions applying immersive audio systems that use either or both sound source control and sound field control. We conducted post-event acoustic measurements or survey of questionnaire supporting that intended acoustic controls had been successfully implemented.

9:20

3aAA5. If "variable acoustics" are an essential element in the long term success of a venue, how do we instill the importance of both making the appropriate alterations for different programs, and in addition, optimally maintaining these systems – both to management as well as operators? Steve Barbar (E-coustic Systems, 18 Springfield St., Belmont, MA 02478, steve@ecousticsystems.com)

The pandemic has revealed "versatility" as a significant asset for both venues, as well as performers and musicians. The ability to successfully alter and optimize venue acoustics to accommodate a wider range of programming has become an important consideration in the design of performing arts venues. The responsibility of making the changes, however, falls to the venue management—who may not have been included in meetings that determined how such systems should be utilized. Even when management and operators are trained on best deployment, personnel can change over time. If the "worth" of these systems is not conveyed to incoming personnel, they may be underutilized, thus compromising design goals. We will discuss some real world examples of both good and "not so good" deployments of variable acoustics.

9:40

3aAA6. Making a room ready and ensuring success for active acoustics systems. Steve Ellison (Meyer Sound Labs, Inc., 2832 San Pablo Ave., Berkeley, CA 94702, ellison@meyersound.com), Pierre Germain, and Roger Schwenke (Meyer Sound Labs, Inc., Berkeley, CA)

Active Acoustics systems can be thought of as reducing the effective absorption of a room and/or increasing its effective volume and depend upon well-designed acoustic treatment and room shaping. Because Active Acoustics systems cannot reduce HVAC noise or improve isolation, these associated acoustical properties depend on the room's acoustical design. Therefore, a successful Active Acoustic system installation relies on coordination with the acoustical consultant, from conceptual design to scheduling initial rehearsals with the various performance groups that utilize the room. Installation examples from around the world, including Australasia, are provided to illustrate lessons learned for developing successful projects.

10:00–10:20 Break

10:20

3aAA7. Integrating spatial audio and active acoustics systems. Steve Ellison (Meyer Sound Labs, Inc., 2832 San Pablo Ave., Berkeley, CA 94702, ellison@meyersound.com) and Roger Schwenke (Meyer Sound Labs, Inc., Berkeley, CA)

Rooms incorporating Spatial Audio and Active Acoustic systems are being built that facilitate immersive audience experiences of unprecedented musical performances. Composers are writing new pieces, and new versions of existing works are being produced to take advantage of these technologies. Spatial Audio enables pre-recorded, synthesized, and live reinforced sounds to arrive at the listener from any direction and allows the listener to move around in the space and explore the sonic experience. Active Acoustics provides early reflections and reverberance for the performers and audience throughout the room and can be adjusted for each composition. The physical acoustical design of the space needs to provide an appropriate nominal acoustic for both systems and accommodate performance audio that does not come from a fixed position and orientation to listeners who may not be in a fixed location and orientation. Both systems incorporate distributed loudspeakers that are typically integrated into the architecture yet have different electroacoustic objectives. How do these systems coexist? Installation examples that use Spacemap, a multichannel panner that originated in Australia, and Constellation, an Active Acoustics System that uses VRAS technology developed in New Zealand, illustrate design considerations when integrating these systems.

10:40

3aAA8. SAFE Credit Union Performing Arts Center audio enhancement to support architecture. David Manley (DLR Group, 6457 Frances St., Omaha, NE 68106, dmanley@dlrgroup.com), Jonathan Hopkins (DLR Group, New York City, NY), Anat Grant (DLR Group, Los Angeles, CA), and James Krumhansl (DLR Group, Cleveland, OH)

Opened in 1976 and without a major renovation since, the 98,000-sf SAFE Credit Union Performing Arts Center was in dire need of transformation. The building's aging infrastructure needed a comprehensive modernization to meet accessibility requirements, the escalating requirements of contemporary performance, and to present a new face to the public as the premier cultural institution in the state capital. The existing theater's acoustics represented a challenge to effectively accommodate the wide range of performance types including Broadway roadshows, the Sacramento Choral Society, and the Sacramento Philharmonic and Opera. Major renovations work

included a complete exterior renovation; new mechanical, plumbing, electrical, and theatrical systems; and substantial updates to the interior finishes throughout the building. Providing an accessible venue to all patrons and performers was the primary goal of the project. Significant modifications to the external and internal circulation included removal of the entire audience chamber floor and new seating and aisle ways throughout. An electro-acoustic enhancement system was designed to provide an acoustically tunable audience chamber and electronic shell on stage to meet all the needs of the resident companies. Design and implementation of the electro-acoustic enhancement system will be discussed, including challenges of performer buy-in and audio orchestra shell integration.

Contributed Paper

11:00

3aAA9. Time reversal acoustic focusing in a reverberant room: Elongating a focus, and the effect of head-scattering. Peter J. Beringer (Sydney School of Architecture, Design and Planning, The Univ. of Sydney, Darlington, New South Wales 2232, Australia, pber8384@uni.sydney.edu.au), Densil Cabrera, and Shuai Lu (Sydney School of Architecture, Design and Planning, The Univ. of Sydney, Darlington, New South Wales, Australia)

Time reversal acoustics (TRA) has tremendous potential for focusing, or sound concentration, in reverberant architectural environments. While TRA provides an inherently simple means of achieving a high degree of spatiotemporal focusing, its appropriateness for listening applications in the built environment is challenged by the need to accommodate a person within a sufficiently large focal area. At its simplest, a reciprocal time reversal focusing system for audible range sound will produce a focus with a spatial spread corresponding to one-eighth of the wavelength, which is inconveniently small for speech frequencies. However, an elongated focus is introduced by deploying a directional impulse response, which can be created with a directional microphone, or synthesized by time-delay-mixing of multiple omnidirectional impulse responses. This paper reports on results of physical experiments conducted in a reverberant room, complemented by finite-different time-domain simulations in which the effect of introducing a head-sized scatterer was investigated.

Session 3aAB

Animal Bioacoustics: Acoustic Ecology and Biological Soundscapes I

Laura Kloepper, Chair
Biological Sciences, University of New Hampshire, 38 Academic Way, Durham, NH 03824

Contributed Papers

8:40

3aAB1. Assessing Northern Gulf of Mexico Sperm Whale's (*Physeter macrocephalus*) time budget using passive acoustic monitoring. Naomi Mathew (Phys., Univ. of Louisiana, 219 Rosemary Pl, Lafayette, LA 70508, naomi.mathew1@louisiana.edu) and Natalia Sidorovskaia (Phys., Univ. of Louisiana at Lafayette, Lafayette, LA)

A time budget empirically measures species-specific behavior regarding the time periods spent engaging in a certain behavior. Sperm whales emit phonations categorized as either foraging or socializing, making it straightforward when creating their time budget based on acoustic observations alone. Understanding regional time budgets can help assessing habitat uses, prey availability, as well as identifying short and long-term trends across the population. Furthermore, sperm whales exhibit both high sexual dimorphism as well as sexually regulated group structures. Individualized multi-pulse echolocation click structure allows total body length as well as sex to be estimated. Apart from a case study conducted by McDonald *et al*., 2017, a long-term time budgets for sperm whales have not been investigated in the Northern Gulf of Mexico (NGoM). The analysis of multi-year archival acoustic data from the NGoM using a sperm whale click detector as well as CABLE software [Beslin *et al*., *JASA* (2018)] is presented to infer sperm whale's time budget and group structures. [Data acquisition for this research was made possible in part by grants from The Gulf of Mexico Research Initiative and BOEM.]

9:00

3aAB2. Using the Soundscape Code to compare coastal marine habitats in the Gulf of Maine. Grant A. Milne (Univ. of New Hampshire, 14 Chesley Ave., Somersworth, NH 03878, grant.milne@unh.edu), Jennifer Miksis-Olds, Alyssa Stasse, Bo-Young Lee, Dylan Wilford (Univ. of New Hampshire, Durham, NH), Shaurya Baruah (The Peddie School, Hightstown, NJ), and Bonnie Brown (Univ. of New Hampshire, Durham, NH)

Identifying similarities and differences in soundscape properties among coastal marine habitats is valuable for determining indicators of habitat composition, assessing functional connectivity among habitats, and informing management decisions regarding the soundscapes of these habitats. The "Soundscape Code," proposed and developed by Dylan Wilford, enables rapid calculation of values for four salient soundscape properties: amplitude, impulsivity, periodicity, and dissimilarity. This enables multivariate statistical analyses to quantitatively compare soundscapes in different habitat types and geographic regions. The objective of the current work was to determine whether geographic region or habitat type accounts for more variability in coastal soundscape properties. Passive acoustic recordings were acquired in three different habitat types (sand, macroalgae, and eelgrass dominated substrates), in each of four different geographic locations along the New Hampshire/Maine coastline, to compare the soundscapes of habitats with varied biological and geophysical substrate composition. Results indicate that geographic location accounted for more variability in the soundscapes than habitat type, suggesting that habitats' local connectivity outweighs acoustic and biological uniformity of the same habitat type over broader spatial scales. Future analyses will incorporate metagenomic data for predictive modeling of habitat composition through the combined use of passive acoustic monitoring and metabarcoding of seawater samples.

9:20

3aAB3. Examining environmental conditions that influence long-term Antarctic blue whale migration patterns in the Southern Hemisphere using an 18-year passive acoustic dataset. Gary Truong (Evolution and Ecology Res. Ctr., UNSW Sydney, E26 Biological Sci. UNSW, Botany Rd., Kensington, New South Wales 2052, Australia, g.truong@unsw.edu.au) and Tracey L. Rogers (Evolution and Ecology Res. Ctr., UNSW Sydney, Kensington, New South Wales, Australia)

Antarctic blue whales (*Balaenoptera musculus intermedia*) are difficult to study due to the low numbers and cryptic behaviour. The largest subspecies of blue whale migrates from Antarctic feeding areas to temperate and tropical breeding grounds every year. Their use of these remote areas makes it difficult to detect them using visual survey methods. However, blue whales are vocal animals and sing throughout the year especially during their annual migration. By using passive acoustic monitoring, we can study the long-term migration patterns of Antarctic blue whales across the Southern Hemisphere. Using an 18-year passive acoustic dataset from the CTBTO, we examined the migration patterns of Antarctic blue whales across four different locations spread across all three major ocean basins. We modelled blue whale acoustic detections with environmental variables including El Niño Southern Oscillation, the Southern Annular Mode, and yearly Antarctic sea-ice extent. We also examined krill abundance estimates from Krill Base in our models. We found that the whale call detections at the four sites were correlated with different variables. The variables were also correlated with various degrees of lag. Our study provides evidence that these whales vary their migration patterns from year-to-year with respect to changing oceanic, and climatic conditions.

9:40

3aAB4. Aerodynamic sound production from flying beetles. John S. Allen (Mech. Eng., Univ. of Hawaii, 2540 Dole St., Holmes Hall 302, Honolulu, HI 96822, alleniii@hawaii.edu) and Kevin O'Rourke (Adaptive Res., Las Vegas, NV)

Extensions of previous research on the aerodynamic sound production by flying beetles are presented. The topic is of interest in terms of both fundamental acoustics and the applications such as classification and tracking of invasive insect species. High speed video with synchronous acoustics recordings of the Coconut Rhinoceros Beetle (Oryctes rhinoceros) and the Oriental Flower Beetle (Protaetia orientalis) provide experimental data for the analytical and numerical models. Three dimensional, unsteady flow simulations were conducted using compressible flow software (CAESIM, Adaptive Research, Inc.) with a TVD methodology. Flapping wing motion requires mesh deformation with a rotation with a prescribed bending and also coupled rotation and translation of the wing's hinge position. Contributions of wing beat frequency and vorticity to sound generation are examined with respect assumptions of the spatial dimensions (two or three dimensional), bending and fluid structure interactions. Underlying sound generation mechanisms at the wing tips are examined in higher fidelity. Both

hovering and forward flight are investigated and compared in terms of the different flapping motion of the two species. In addition, relationships of the tones to the temporal lift and drag lift coefficients are highlighted.

10:00–10:20 Break

10:20

3aAB5. Suspected evening fish chorusing sounds detected in deep waters off Kauai, Hawaii. Stephen W. Martin (Res., National Marine Mammal Foundation, 2240 Shelter Island Dr., Ste. 200, San Diego, CA 92016, steve.martin@nmmf.org), Gabriela C. Alongi (Res., National Marine Mammal Foundation, San Diego, CA), Tyler A. Helble, Cameron R. Martin (NIWC Pacific, San Diego, CA), Brian Matsuyama (Res., National Marine Mammal Foundation, San Diego, CA), and Erin E. Henderson (NIWC Pacific, San Diego, CA)

Noise analyses are crucial to understanding passive acoustic monitoring for marine mammals and their behavioral responses to natural and anthropogenic noises. Suspected biological sounds in the ambient noise data from hydrophones at the Pacific Missile Range Facility (PMRF), Kauai, Hawaii have been found. The sounds are suspected to be chorusing associated with the Deep Scattering Layer (DSL) upward daily vertical migration given they peak around 2 hours after sunset. They have been observed occurring daily including a period in January 2017 when high ambient noise levels during a storm was associated with baleen whale cessation of calling. The sounds occur from 1.4 to 1.8 kHz with the peak near 1650 Hz. They are detectable on multiple broadband hydrophones (21 to 68 km offshore) in deep water (1.5 to 4.7 km), suggesting a large spatial extent. The maximum spectrum level in the 1.6 kHz 1/3rd octave band observed to date was 71 dB re 1 μPa^2/Hz. Individual sounds have not been identified to date given the geometries involved and the large spatial extent sensed using deep water hydrophones. These results have similarities to related chorusing research which has suggested a potential link with Myctophidae fishes.

10:40

3aAB6. Voices from the deep: Odontocete occurrence patterns relative to oceanographic conditions in the northern California Current System. Marissa Garcia (K. Lisa Yang Ctr. for Conservation Bioacoustics, Cornell Univ., 159 Sapsucker Woods Rd., Ithaca, NY 14850, mg2377@cornell.edu), Rebecca Cohen (K. Lisa Yang Ctr. for Conservation Bioacoustics, Cornell Univ., Ithaca, NY), Craig Risien (College of Earth, Ocean & Atmospheric Sci., Oregon State Univ., Corvallis, OR), Dawn Barlow, Solne Derville (Marine Mammal Inst., Oregon State Univ., Newport, OR), Melanie Fewings (College of Earth, Ocean & Atmospheric Sci., Oregon State Univ., Corvallis, OR), Leigh Torres (Marine Mammal Inst., Oregon State Univ., Newport, OR), and Holger Klinck (K. Lisa Yang Ctr. for Conservation Bioacoustics, Cornell Univ., Ithaca, NY)

Odontocetes play an important ecological role as apex predators in the northern California Current System (CCS), which is characterized by seasonal rises in primary productivity fueled by wind-driven upwelling of cold nutrient-rich water. This productivity sustains higher trophic-level prey optimal for deep-diving odontocetes including beaked and sperm whales. Due to the cryptic ecology of these whales, there are few systematic studies of their occurrence across seasons in northern CCS waters. The Holistic Assessment of Living marine resources off Oregon (HALO) Project—a collaboration between Oregon State University and Cornell University—addresses this gap via quarterly vessel-based visual surveys and year-round passive acoustic monitoring (PAM). Here, we detect and classify species-specific odontocete acoustic signals using long-term spectral averages, automated click detection, and unsupervised clustering of semi-continuous PAM data from 10/2021 to 12/2022. Site-specific occurrence of beaked and sperm whales is revealed on hourly to seasonal temporal scales at three recording sites along a depth gradient spanning the continental slope (300 and 630 m water depth) and abyssal plain (2860 m). By relating odontocete detections to dynamic oceanographic conditions from nearby profiling moorings, we identify likely drivers of prey availability influencing predator distribution. These results will inform conservation efforts of cryptic odontocetes in the face of rapid environmental change.

11:00

3aAB7. Underwater soundscape observations and associated demersal fish movement during a seismic airgun survey in coastal Oregon. Kaustubha Raghukumar (Integral Consulting, Inc., 200 Washington St., Ste. 201, Santa Cruz, CA 95060, kraghukumar@integral-corp.com), Katherine Heal (Pacific Northwest National Labs., Eugene, OR), Lauren Borland, Sarah Henkel, Taylor Chapple, and Scott Heppell (Oregon State Univ., Corvallis, OR)

Sound is a crucial aspect of the underwater environment for fishes—various species use sound to communicate, identify predators, navigate, and many other activities needed for survival in their habitat. In the summer of 2021, a seismic survey passed nearby Southern Oregon to map the Cascadia Subduction Zone using an array of airguns. To evaluate the effect of the seismic survey on the behavior of demersal fish, acoustic pressure and particle motion measurements were collected within a marine protected area, accompanied by tagging of two rockfish species, Black Rockfish (*Sebastes melanops*) and China Rockfish (*S. nebulosus*). Acoustic measurements were obtained using a vector sensor array, deployed in 25 m deep water. It was found that while cumulative sound exposure levels were dominated by wind noise, other metrics such as peak sound pressure, kurtosis, acoustic complexity index, crest factor and acoustic entropy showed clear signals associated with the seismic survey. Animal behavior and spatial use by the two rockfish species were also evaluated. Overall, results indicate that there are slight differences in movement and spatial use during times of seismic survey noise presence, but observed differences are reduced after only a few days.

11:20

3aAB8. Model updating of flowering snapdragon (*Antirrhinum litigiosum*) biomechanical responses to vibro-acoustic stimuli. Can Nerse (Ctr. for Audio, Acoust. and Vib. (CAAV), Univ. of Technol. Sydney, 123 Broadway, Ultimo, New South Wales 2007, Australia, can.nerse@uts.edu.au), Abhishek R. Mohapatra, Sebastian Oberst (Ctr. for Audio, Acoust. and Vib. (CAAV), Univ. of Technol. Sydney, Ultimo, New South Wales, Australia), David Navarro-Payá, Jone Etxeberria, José T. Matus (Inst. for Integrative Systems Biology (I2SysBio), CSIC-Universitat de Valencia, Valencia, Spain), Lorenzo Bianco, Maria R. Tucci, Elena Cumino, Luca P. Casacci, and Francesca Barbero (Dept. of Life Sci. and Systems Biology, Univ. of Turin, Turin, Italy)

The combined variation in gene expression and environmental conditions during flower development can result in phenotypic differences in shape, size, and material composition. Biomechanical responses in flower organs due to external stimuli can be mechanically measured at various levels. Here, we investigate snapdragon (*Antirrhinum litigiosum*) response to vibro-acoustic stimuli by an interdisciplinary model updating framework. In a climate-controlled setup, sweep signals and artificial signals representative of plant pollinator species were given as excitation input through a loudspeaker to a set of plants; vibrations of the flower organs were measured by laser Doppler vibrometry. Geometric features of the plants were identified using LiDAR combined with photogrammetry, while the density distribution in the flower organs and internal dimensions were estimated using micro-computed tomography scans. A computer model using finite element method was used to identify material properties of the flower organs by combining time domain measurements and dimensional classification. Results demonstrate density and stiffness gradient in the corolla contributing to a modal activity that is adaptive to local conditions and pollinators, but resilient against external noise. The framework outlined herein may give clues to which pollinators induce early-plant responses. [The authors acknowledge the support of the Human Frontier Science Program (HFSP) grant RGP0003/2022.]

11:40

3aAB9. Natural soundscape variation appears to influence the structure of stereotypic calls and repertoires in killer whales. Harald Yurk (Ecosystem Sci. Div. Pacific, Fisheries and Oceans Canada, 4160 Marine Dr., West Vancouver, BC V7V 1N6, Canada, harald.yurk@dfo-mpo.gc.ca), Caitlin O'Neill (Ocean Sci. Div., Fisheries and Oceans Canada, SIDNEY, BC, Canada), Lucy S. Quayle (Ecosystem Sci. Div. Pacific, Fisheries and Oceans Canada, West Vancouver, BC, Canada), Svein Vagle (Ocean Sci. Div., Fisheries and Oceans Canada, SIDNEY, BC, Canada), and Holly T. LeBlond (Ecosystem Sci. Div. Pacific, Fisheries and Oceans Canada, West Vancouver, BC, Canada)

Underwater soundscapes show dynamic variations in natural ambient sound levels at different locations and water depths and levels can vary over a course of a day. This soundscape dynamicity may pose a vocal challenge for highly mobile marine mammals whose habitat overlaps with many dynamic soundscapes. The most prominent communication tool of killer whales is a pulsated often multicomponent call which can be detected at more than 15 kilometers under quiet conditions. Social learning of call structure is a driver for stereotypic long lasting group- or population-specific call repertoires. Here we describe the potential vocal adaptations of killer whales to reduce the influence of sound propagation loss (PL) on their calls. Reliable propagation to identify groups and populations at considerable distances is important. In experimental field trials, the PL of killer whale calls and that of tones and sweeps was examined to determine if call component PL differs among soundscapes. PL is positively correlated with spatial and temporal ambient noise level variation. Furthermore, the use of burst pulses, a common feature of calls increased propagation distances at higher noise levels. We conclude that noise variation is one of the drivers of call structure and may also influence the call repertoire size.

Session 3aAO

Acoustical Oceanography, Underwater Acoustics and Physical Acoustics: Observing the Ocean Acoustically using Submarine Cable Systems I

Shima Abadi, Cochair
University of Washington, 1501 NE Boat St, Seattle, WA 98195

Ying-Tsong Lin, Cochair
University of California, San Diego, Scripps Institution of Oceanography, La Jolla, CA 92037

Bruce Howe, Cochair
Ocean and Resources Engineering, University of Hawaii at Manoa, 2540 Dole Street, Holmes Hall 402, Honolulu, HI 96822

Haris Kunnath, Cochair
Environment, CSIRO, GPO Box 1538, Hobart, 7001, Australia

Invited Papers

8:00

3aAO1. Interpreting phase shifts of basin-scale arrivals observed in the North Pacific. David R. Dall'Osto (Acoust., Appl. Phys. Lab. at Univ. of Washington, Seattle, WA) and Kay L. Gemba (Naval Post Graduate School, Monterey, CA93943, kay.gemba@nps.edu)

The 75 Hz Kauai Beacon is well situated for observing the North Pacific Ocean acoustically. Somewhat conveniently, three ocean observing networks, with acoustic receivers located within the deep sound channel: two observatories 4 mm to the East (RCO: Regional Cabled Observatory and MARS: Monterey Accelerated Research System), and the International Monitoring Station at Wake Island 3.5 mm to the West. During April–June of 2023, the Beacon transmitted a 37.5 Hz bandwidth m-sequence signal several times a day every 4 days. Upon processing, a persistent arrival structure is identified, which for which each arrival exhibits a unique, yet relatively steady phase over the course of a 20-min transmission. Leveraging acoustic propagation models to better understand the effect of the fluctuating ocean environment, we address the stability of the arrivals over the 20-min transmissions, from hour to hour, and over the course of the 3-month study. This includes examination of potential causes for the slow-phase shifts observed at all of the stations, and how it relates to Doppler. This talk concludes with a discussion on the future utility and research topics of interest, including long-range geo-positioning (underwater GPS), communications, thermometry, and tomography.

8:20

3aAO2. Basin scale propagation modeling of MLS signals from the Kauai Beacon source. Nicholas C. Durofchalk (Phys., Naval Postgrad. School, 1 University Circle, Monterey, CA 93943, Spanagel Hall, Rm. 141, Monterey, CA 93943, nicholas.durofchalk@nps.edu), Kay L. Gemba, Paul Leary, and Kevin Smith (Phys., Naval Postgrad. School, Monterey, CA)

Long distance underwater acoustic propagation is of interest for a variety of applications including underwater navigation, yet modeling such propagation is challenging due to the large degree of environmental uncertainty. In this presentation, the propagation of 75 Hz center frequency maximum length sequence (MLS) signals emitted from a submerged source near Kauai and received at various sites throughout the Pacific Ocean Basin is modeled with the Bellhop ray tracing [1] and Monterey-Newport Parabolic Equation [2] models. Sites such as the Monterey Accelerated Research System (MARS) observatory and the Comprehensive Nuclear Test Ban Organization (CTBTO) hydroacoustic monitoring station near Wake Island are included in the discussion. The range-dependent environment is based on historical profiles from the World Ocean Atlas database and reanalysis data from the hybrid coordinate ocean model (HYCOM) [3]. Bathymetry profiles along the axis of propagation are interpolated from the General Bathymetric Chart of the Oceans (GEBCO) 2021 database [4]. Simulated broadband channel impulse responses transmission loss levels in the modeled environment are compared to selected receptions of the signal at the hydrophone sites. Additionally, fluctuations of the modeled transmission loss in the presence of an internal wave model [5] are discussed.

Contributed Paper

8:40

3aAO3. Detecting the Kauai source beacon with ocean observatories innitiative hydrophones. John Ragland (Univ. of Washington, 185 W Stevens Way NE, Seattle, WA 98195, jhrag@uw.edu), Nicholas C. Durofchalk, Kay L. Gemba (Phys., Naval Postgrad. School, Monterey, CA), and Shima Abadi (Univ. of Washington, Seattle, WA)

In March of 2023, a sound source off the coast of Kauai began regularly transmitting a 75 Hz signal into the ocean, six times a day, every fourth day. The regular transmission of this signal allows for the long-term measurements of ocean basin scale temperature changes using the technique of acoustic tomography, as well as further investigations of long-distance acoustic propagation. The Ocean Observatory Initiative (OOI) has, as part of its suite of oceanographic sensors, 11 hydrophones that are continually recording ambient sound. These hydrophones are in various diverse environments in the north-east pacific including bottom mounted in shallow environments, deep environments, on top of seamounts, and moored at 200 m below the ocean surface. In this talk, we present preliminary results of acoustic propagation modeling and data analysis towards receiving the Kauai signal with the OOI hydrophones. The diverse environments of the hydrophones allow us to investigate how different levels of ambient sound and environmental factors, such as proximity to the sound channel affect the ability to detect a positive reception of the Kauai source across the ocean basin. [Work supported by ONR.]

Invited Papers

9:00

3aAO4. A tale of whale and fiber: monitoring baleen whales with Distributed Acoustic Sensing (DAS). Léa Bouffaut (K. Lisa Yang Ctr. for Conservation Bioacoustics, Cornell Univ., Lab of Ornithology, 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) converts fibers in existing underwater telecommunication fiber-optic cables into dense listening arrays of strain sensors. Recent advances in DAS interrogating technology enable increasing data quality, spatial coverage, and bandwidth, sparking interest in applied environmental sensing. Initially focused on seismic data collection, e.g., for earthquake monitoring, DAS has demonstrated abilities in detecting waterborne acoustic sources, including the low-frequency sounds of blue and fin whales (*Balaenoptera musculus* and *B. physalus*) over tens of km of fiber. DAS is instrumented from land, collects data in near-real time, and offers region-wide spatial coverage. Thus, this technology provides a unique opportunity for robust monitoring of endangered and threatened migratory baleen whales at ecologically meaningful spatial and temporal scales. This talk will introduce our research on using DAS to monitor low-frequency baleen whales, including detection and tracking, and the open-access DAS4Whales python package, developed to analyze terabytes of spatio-temporal data collected with DAS. We will also address the potential limitations of this new monitoring technology and provide a roadmap for further research and testing needed to exploit the technology's full potential.

9:20

3aAO5. On the potential of using Distributed Acoustic Sensing (DAS) to build teleseism based ocean-tomography network. David R. Dall'Osto (Acoust., Appl. Phys. Lab. at Univ. of Washington, 1013 N 40th St., Seattle, WA 98105, dallosto@apl.washington.edu)

The sound from marine teleseisms rank among some of the loudest in Earth's Oceans, typically radiated into the ocean near the epicenter and propagating across the entire ocean basin. Acoustic sensing on ocean observatories provide a critical path to better understand origins of these low-frequency sound waves known as T-phases, and structure of the ocean over the megameter propagation through the ocean. Of the emerging technologies implemented on these observatories, distributed acoustic sensing (DAS) has the potential to revolutionize marine seismoacoustics. DAS can be implemented on unused oceanic fibers on the continental slopes surrounding an ocean basin could readily enable a teleseism based ocean-tomography network with surprisingly little new infrastructure. In November 2021, a DAS system installed on two runs of fiber optic cable that connect to an ocean observatory measuring offshore volcanic activity, detected a few examples of teleseismic T-phases. Combined with other hydroacoustic ocean observatories in the Pacific Ocean, the T-phase origins are localized to bathymetric features over 100 km from the epicenter, an important consideration when utilizing these T-phases as sources of opportunity to study the ocean.

9:40

3aAO6. Deep clustering analysis for data exploration and anomaly detection in distributed acoustic sensing (DAS) systems. William F. Jenkins (Scripps Inst. of Oceanogr., UC San Diego, 9500 Gilman Dr., La Jolla, CA 92093, wjenkins@ucsd.edu), Chih-Chieh Chien (Scripps Inst. of Oceanogr., UC San Diego, La Jolla, CA), and Peter Gerstoft (Scripps Inst. of Oceanogr., UC San Diego, San Diego, CA)

Distributed acoustic systems (DAS) offer a unique opportunity to sense seismic and acoustic waves at high sampling rates and spatial resolutions using fiber optic cables. The prevalence of fiber optic cables crossing seabeds around the world make them particularly attractive sensors of opportunity for acoustical and geophysical applications. However, the high spatial and temporal resolutions of these systems pose significant challenges to efficient data processing and analysis. To overcome this, we propose a two-step exploratory analysis procedure in which the dimensionality of the data is first reduced into a latent space containing its salient information, and then perform clustering analysis of the data in the latent space. Spectrograms of detected events are encoded into the latent space using a convolutional autoencoder, and clustering is performed using Gaussian mixture model clustering. We demonstrate our technique on experimental data collected from a DAS array deployed at a geothermal energy research site. With this technique, we show a data-driven approach to exploratory data analysis, enabling identification of dominant types of detections in the data and identification and removal of anomalous detections due to instrument noise.

10:00–10:20 Break

Contributed Papers

10:20

3aAO7. Performance of distributed acoustic sensing relative to co-located hydrophone measurements. Alexander S. Douglass (Oceanogr., Univ. of Washington, 1501 NE Boat St., MSB 206, Seattle, WA 98195, asd21@uw.edu) and Shima Abadi (Univ. of Washington, Seattle, WA)

Distributed acoustic sensing (DAS) is a technology that transforms telecommunication fiber optic cables into dense sensor arrays by continuously transmitting pulses of light down the cable and measuring backscattering from natural inhomogeneities within the fiber cable. The technology can densely sample the acoustic field over long ranges (up to 100 km), providing a means for large scale passive acoustic monitoring. The DAS Calibration Experiment 2022 (DASCAL22) provides 9 days of co-located hydrophone and DAS data in Puget Sound, WA. The DAS data were recorded with a 2 kHz sampling rate over $\sim$3.5 km of cable in a channel reaching close to 100 m depth and with DAS channels spaced 6.38 m. A combination of an active source broadcast from 1, 5, and 10 m depths, and passive signals (primarily boat noise) are used to investigate the capabilities of DAS to sense a range of frequencies and the impacts of the environment on those capabilities.

10:40

3aAO8. Assessment of long-range distributed acoustic sensing (DAS) capabilities in controlled environments. Evgenii Sidenko (Ctr. for Marine Sci. and Technol., Curtin Univ., Curtin Univ., B301 Hayman Rd., Bentley, Perth, Western Australia 6102, Australia, evgeny.sidenko@curtin.edu.au), Olivia Collet (Ctr. for Exploration Geophys., Curtin Univ., Kensington, Western Australia, Australia), Alexander Gavrilov (Ctr. for Marine Sci. and Technol., Curtin Univ., Bentley, Western Australia, Australia), Alec J. Duncan, Christine Erbe (Ctr. for Marine Sci. and Technol., Curtin Univ., Perth, Western Australia, Australia), Maxim Lebedev, Konstantin Tertyshnikov (Ctr. for Exploration Geophys., Curtin Univ., Kensington, Western Australia, Australia), Boris Gurevich (Ctr. for Exploration Geophys., Curtin Univ., Perth, Western Australia, Australia), Robert McCauley (Ctr. for Marine Sci. and Technol., Curtin Univ., Perth, Western Australia, Australia), and Roman Pevzner (Ctr. for Exploration Geophys., Curtin Univ., Kensington, Western Australia, Australia)

Distributed Acoustic Sensing (DAS) is an emerging technology enabling the recording of disturbances along fiber-optic (FO) cables with a wide range of applications, such as monitoring marine fauna, anthropogenic activities, and ocean climate. Despite the extensive research on DAS technology, its long-range capabilities have not been structurally studied yet. We present results of a sequence of purposefully designed experiments of DAS capabilities in (1) a laboratory, (2) a pool, and (3) a marine facility. To simulate long cables, we use combinations of FO coils 10–200 km in length. In the laboratory, realistic strain signals are simulated by an FO stretcher. In the pool, realistic acoustic signals are transmitted by an underwater speaker. At the marine facility, a combination of playback and real sound sources is used. Simultaneously deployed hydrophones allow DAS calibration and performance assessment. In this series of experiments, we measure internal noise, compare different DAS equipment, and successively optimize acquisition parameters for each interrogator performing at different, long sensing ranges in increasingly realistic environments. We show that long-range capabilities are strongly dependent on DAS settings (optical power, pulse repetition frequency, gauge length, pulse width) and vary between different interrogators.

Invited Paper

11:00

3aAO9. SMART submarine cable technology can facilitate acoustics on the global scale. Bruce Howe (Ocean and Resources Eng., Univ. of Hawaii at Manoa, 2540 Dole St., Holmes Hall 402, Honolulu, HI 96822, bhowe@hawaii.edu)

The submarine cable industry is beginning to share their infrastructure for ocean observing. Science Monitoring And Reliable Telecommunications (SMART) Subsea Cables is working to integrate temperature, pressure, and seismic acceleration sensors into commercial cables ($\sim$70 km spacing) to support climate monitoring and disaster risk reduction on the global scale. The seismic and pressure sensors are expected to have sensitivity at low acoustic frequencies, (e.g., $<$50 Hz and $<$5 Hz, respectively); future systems could include hydrophones and other sensors. Further, telecom rated branching cables supporting multipurpose "nodes" are becoming a reality (per existing dedicated science cable systems). These can support low frequency transceivers and autonomous undersea vehicles (AUVs). With cabled power, these would be part of the fixed/mobile acoustic tomography system measuring ocean heat content and more generally for transporting energy, data, and acquiring multidisciplinary data throughout a large volume of the ocean. A 3700 km SMART ring system in Portugal will be ready in 2026 and one connecting Vanuatu and New Caledonia is starting. Others in planning stages that could include these concepts are: Far North Fiber connecting Norway/Finland/Ireland with Japan, the NSF proposed SMART cable connecting New Zealand with Antarctica, New Zealand-Chatham Islands, and Lisbon-Egypt through the Mediterranean.

Session 3aBA

Biomedical Acoustics and Physical Acoustics: Bridging Preclinical and Clinical Acoustics I

Misun Hwang, Cochair
Radiology, Children's Hospital of Philadelphia, 3401 Civic Center Blvd, Philadelphia, PA 08057

Shashank Sirsi, Cochair
Bioengineering, Univ. of Texas at Dallas, Richardson, TX 75080

Chair's Introduction—9:15

Invited Paper

9:20

3aBA1. Toward translating novel imaging biomarkers from preclinical to clinical settings. Misun Hwang (Radiology, Children's Hospital of Philadelphia, Philadelphia, PA, Hwangm@email.chop.edu)

Microvascular imaging of the neonatal brain is now feasible with the advent of ultrasound technologies that enable detection of slow flow in smaller vessels. As an advancement from conventional Doppler, microvascular imaging (MVI) technique has been applied in neonatal brain ultrasound scans to better understand how anatomic and functional changes of cerebral microvessels relate to disease diagnostics and prognosis in the neonatal brain. This lecture will introduce about the technique, current and emerging applications of MVI, as well as challenges and future directions of advancing cerebral microvascular markers of neonatal brain health. Some of the invasive methods used to study cerebral physiology and biomarkers in high-fidelity preclinical models are difficult to directly test and validate in vulnerable neonates. As such, this lecture will open up the forum for future directions on how the scientific and medical fields can come together in advancing the clinical validation and integration of novel imaging biomarkers to improve neonatal brain health.

Contributed Papers

9:40

3aBA2. Noninvasive investigation of dysphagia by auscultation of acoustics of deglutition by high impedance earphones placed in the ear canals. Amitava Biswas (CDS, SUNY Fredonia, 118 College Dr. #5092, USM-CHS-SHS, Hattiesburg, MS 39406-0001, Amitava.Biswas@usm.edu) and Kevin Kearns (CDS, SUNY Fredonia, Fredonia, NY)

This ongoing investigation is utilizing basic communication hardware originally distributed by Radio Shack. The earphones in these devices can work as microphones, receiving vocal utterances through the pharyngeal tissue. We are investigating the potential of such systems to monitor normal and abnormal swallowing sounds just like medical personnel monitor the heart and the lungs by auscultating with stethoscopes. Compared to regular passive stethoscopes, our setup permits stable location, signal amplification, noise filtration, digital storage, post-processing, and statistical analyses of the acoustics of deglutition. Initial findings of this study will be presented. The limitations of this study and future course of action will be discussed.

10:00

3aBA3. Attenuation study with hemoglobin microbubbles for acoustic blood oxygen level dependent imaging. Nasrin Akter (Bioengineering, The Univ. of Texas at Dallas, 800 W. Campbell Rd., Richardson, TX 75080, nxa190008@utdallas.edu), Sugandha Chaudhary, and Shashank Sirsi (Bioengineering, The Univ. of Texas at Dallas, Richardson, TX)

A new microbubble-based biosensor for BOLD imaging applications was developed and acoustically characterized using ultrasound. The biosensor design was based on the use of hemoglobin protein as a shell component of microbubbles, which undergoes changes in structure upon oxygen binding, leading to changes in the mechanical properties of the shells. The acoustic characterization was performed using an ultrasound acoustic transducer with a center frequency of 1 MHz, which was used both as a source and a receiver. The transducer transmitted 5 us pulses at a PRF of 10 Hz at peak-negative pressures ranging from 0 to 0.1 MPa. The microbubbles were then excited, and their signals were measured and transformed to frequency domain using Fast-Fourier transform, from which attenuation was computed. The oxygen-saturated hemoglobin microbubbles were added to either oxy or deoxy PBS at a concentration of 4.5×6 mB/ml to collect the microbubble signal. The results showed that the biosensor has potential for use in BOLD imaging applications. The results showed that hemoglobin microbubbles were displaying distinct attenuation properties between oxy and deoxy conditions, thereby showing response to different levels of oxygen.

10:20–10:40 Break

Invited Papers

10:40

3aBA4. Detection of cerebral ischemic injury during extracorporeal membrane oxygenation using microvascular flows measured by ultrasound localization microscopy. Zeng Zhang (Dept. of Mech. Eng., Johns Hopkins Univ., b31 Latrobe Hall, 3400 North Charles St., Baltimore, MD 21218, zeng@jhu.edu), Misun Hwang, Colbey Freeman, Laith Sultan, Sophie Haddad (Dept. of Radiology, Children's Hospital of Philadelphia, Philadelphia, PA), Todd Kilbaugh (Dept. of Anesthesiology and Critical Care Medicine, Children's Hospital of Philadelphia, Philadelphia, PA), and Joseph Katz (Dept. of Mech. Eng., Johns Hopkins Univ., Baltimore, MD)

Pediatric Extracorporeal Membrane Oxygenation (ECMO) is a life-saving therapy. Affecting 8–50% of patients, brain injury is among the most frequent complications. Non-invasive monitoring of brain perfusion during ECMO using contrast-enhanced ultrasound imaging can potentially improve the outcome. Here we utilize an in-house developed Ultrasound Localization Microscopy (ULM) procedure, which involves a super-resolution deep learning approach for subpixel localization of microbubbles and Kalman filter-based bubble tracking, to measure the brain blood perfusion in a pediatric pig model during ECMO. Parameters, including the mean velocity magnitude in large (>1mm) and medium (0.2–1 mm) vessels, as well as the cerebral microcirculation for micro (<0.2 mm) vessels, are analyzed and compared with the histological readings of brain ischemic injury levels. Data show significantly reduced micro perfusion in the cortex and thalamus with increasing levels of ischemic injury. Compared with large and medium vascular flows, microvascular flows are more sensitive in not only detecting injury but also evaluating injury levels. These results suggest that the management of ECMO patients could be guided by and benefit from the non-invasive monitoring of cerebral microvascular flows.

11:00

3aBA5. Cerebral microvascular markers of intracranial pressure in acute and subacute porcine hydrocephalus model. David Q. Le (Radiology, Children's Hospital of Philadelphia, 3615 Civic Ctr. Blvd, Philadelphia, PA 19104, led1@chop.edu), Zeng Zhang (Mech. Eng., Johns Hopkins Univ., Baltimore, MD), Mrigendra B. Karmacharya, Laith Sultan (Radiology, Children's Hospital of Philadelphia, Philadelphia, PA), Todd Kilbaugh (Dept. of Anesthesiology and Critical Care Medicine, Children's Hospital of Philadelphia, Philadelphia, PA), Joseph Katz (Mech. Eng., Johns Hopkins Univ., Baltimore, MD), and Misun Hwang (Radiology, Children's Hospital of Philadelphia, Philadelphia, PA)

Hydrocephalus affects 1.1% of infants, resulting in abnormal accumulation of cerebrospinal fluid and elevated intracranial pressure (ICP). Elevated ICP can occur prior to detectable morphological changes using computed tomography, resulting in delayed diagnosis and treatment. To improve early diagnosis, we propose vascular imaging using ultrasound localization microscopy (ULM) and ultrafast Doppler to measure cerebral blood flow changes due to acute to subacute elevations in ICP. We performed a pre-clinical study on a neonatal porcine model with 30 min, 8 h, 12 h, and 24 h ICP elevations while performing brain ultrasound measurements in hourly intervals. Ultrafast plane-wave ultrasound imaging was implemented using the Verasonics Vantage-256 and GE IC59D curved array probe at 400 Hz frame rate. Lumason contrast agent was infused into each animal, providing tracers for ULM and enhancing the ultrafast Doppler imaging. Based on preliminary analysis, ultrafast Doppler showed reduced total cerebral blood volume after acute ICP elevation. ULM showed highly correlated parabolic reductions in cortical microcirculation with increasing ICP. Drastic reductions in cerebral microcirculation but less so in microcirculation accompanied microdialysis evidence of brain ischemia. Additionally, there were spatiotemporal changes in cerebral microvascular flow with varying ICP durations that warrant additional investigations using ultrafast ultrasound imaging.

Contributed Papers

11:20

3aBA6. High-frequency ultrasound imaging of left-ventricle vortex flow in a mouse model. Daniel H. Gross (Dept. of Radiology, Weill Cornell Medicine, New York, NY), Colin K. Phoon (Div. of Pediatric Cardiology, Hassenfeld Children's Hospital at NYU Langone, New York, NY), Glenn I. Fishman (Leon H. Charney Div. of Cardiology, NYU Langone Health, New York, NY), and Jeffrey A. Ketterling (Dept. of Radiology, Weill Cornell Medicine, 416 E 55th St. MR-008, New York, NY 10022, jek4011@med.cornell.edu)

Vector flow (VF) ultrasound is available for human cardiac imaging to quantify advanced flow parameters such as vorticity, an important component of healthy cardiac function. Similar methods have yet to translate to murine models because of the need for fine temporal and spatial resolution. We evaluate the ability of a 30-MHz high-frequency probe to quantify vorticity in the murine left ventricle (LV) using a Verasonics Vantage 256. Acoustic beam properties were characterized with an 18-um wire. A 1-cm diameter rotation phantom and a flow channel were used to validate vector estimates. Transthoracic LV image data were acquired from wild-type mice. Flow parameters were obtained with custom software that implemented a multi-angle least-squares VF method. Data were processed to obtain VF estimates and local time histories of vorticity. Velocity and vorticity estimates had more bias as the axial velocity component decreased. Aliasing was observed and a basic dealiasing approach was implemented prior to final velocity estimates. Vorticity peaked during diastole at roughly 300 rad/s, consistent with past measurements at lower frequency. Results show VF can be implemented in murine models with HF probes, but that aliasing must be addressed to obtain reliable estimates. [Work supported by NIH EB032082.]

11:40

3aBA7. Developing a focused ultrasound treatment for tendinopathies in a large animal model. Grace Wood, Jacob Elliott (Graduate Program in Acoust., The Penn State Univ., University Park, PA), and Julianna Simon (Graduate Program in Acoust., The Penn State Univ., 201E Appl. Sci. Bldg, University Park, PA 16802, jcs516@psu.edu)

Current treatments for tendinopathies produce mixed success rates, with 0-84% of patients showing improvement. Our prior work in rats suggests mild microdamage induced by focused ultrasound (fUS) may perform better than traditional treatments for tendinopathies, like dry needling. Here, we work to scale our work in rats to larger animal models and develop treatment monitoring. *Ex vivo* bovine digital flexor tendons were obtained and injected with collagenase to induce chronic tendinopathy. One day later, tendons were treated with focused ultrasound at 1.1–3.68 MHz, with 1–10 ms pulses at pressures up to $p+ = 127$ MPa/$p- = 35$ MPa repeated at 0.2–1 Hz for up to 20 min. Treatments were monitored with passive

cavitation detection (PCD; Sonic Concepts Y-107) and Philips/ATL L7-4 controlled by the Verasonics research ultrasound system in passive cavitation imaging (PCI) mode. Samples were collected for histological analysis. Preliminary results suggest that lower pulse repetition frequencies produce mild mechanical microdamage with no evidence of thermal necrosis. PCI and PCD results show no consistent trends to indicate the intended mechanical microdamage bioeffect has been achieved. Work is continuing to determine the treatment parameters that produce the intended bioeffects in the smallest time and to develop a means to monitor treatment progress. [Work supported by NIHR01EB032860.]

WEDNESDAY MORNING, 6 DECEMBER 2023 ROOM C2.6, 8:00 A.M. TO 10:20 A.M.

Session 3aCA

Computational Acoustics and Physical Acoustics: Innovations in Computational Acoustics I

Danielle Moreau, Cochair
UNSW Sydney, Sydney 2015, Australia

Steffen Marburg, Cochair
Technical University of Munich, Boltzmannstr. 15, Garching 87548, Germany

3a WED. AM

Contributed Papers

8:00

3aCA1. Advanced numerical techniques for predicting frequency response functions. Kuangcheng Wu (Signatures Dept., Naval Surface Warfare Ctr. - Carderock Div., 9500 MacArthur Blvd, West Bethesda, MD 20817, kcwu@msn.com)

Numerical methods have been widely used to predict Frequency Response Functions (FRF) of structures. The FRF can be used to characterize structural dynamics and support sound and vibration control. Generally, frequency sweep is conducted in the FRF calculations to identify damping and resonant frequencies of structures under excitations. However, as numerical models are getting complex and larger, the demands for computational resources (e.g., CPU time, memory, disk space) are greatly increased. Lately, high performance computing (HPC) at the DoD HPC center has been used with advanced numerical techniques to reduce the overall computational time. Those advanced numerical techniques include Krylov subspace and Galerkin Projection (KGP) and Pade Approximation. Significantly CPU time reduction has been demonstrated for relatively smaller FE models. This paper will further discuss adaptive KGP by automating the frequency sweep process via calculated relative residues, then combine with Finite Element Tearing and Interconnecting (FETI) to solve for FRF of a large FEM model in the order of 60M DOF. A flat plate with viscoelastic layer will serve as an example to demonstrate accuracy and efficiency of AKGP with FETI. The benefits of using HPC with advanced numerical techniques to solve for large FEM models are clearly displayed.

8:20

3aCA2. Fast and accurate boundary element methods for large-scale computational acoustics. Elwin van 't Wout (Inst. for Mathematical and Computational Eng., Pontificia Universidad Católica de Chile, Av Vicuña Mackenna 4860, Santiago, Chile, e.wout@uc.cl), Reza Haqshenas, Pierre Gélat, and Nader Saffari (Univ. College London, London, United Kingdom)

The boundary element method (BEM) is a powerful algorithm to solve the Helmholtz equation for harmonic acoustic waves. The explicit use of Green's functions avoids domain truncation of unbounded regions and accurately models wave propagation through homogeneous materials. Furthermore, fast multipole and hierarchical compression techniques provide efficient computations for dense matrix multiplications. However, the convergence of the iterative linear solvers deteriorates significantly when frequencies are high or materials have large contrasts in density or speed of sound. This talk presents several algorithmic improvements of the BEM. First, a preconditioner based on on-surface radiation conditions drastically reduces the iteration count of linear solvers at high frequencies. Second, a novel boundary integral formulation remains well-conditioned for high-contrast transmission problems. We used our fast and accurate BEM implementation to simulate focused ultrasound propagation in the human body, which can be translated to important biomedical applications such as the non-invasive treatment of liver cancer and neuromodulation of the brain. We validated the methodology within the benchmarking exercise of the International Transcranial Ultrasonic Stimulation Safety and Standards (ITRUSST) consortium. As a second application, we simulated the collective resonances of water-entrained arrays of air bubbles. Finally, we implemented all functionality in our open-source Python library, OptimUS.

8:40

3aCA3. Multi-fidelity models based on Gaussian processes in acoustic applications. Caglar Gurbuz (Chair of Vibroacoustics, Tech. Univ. of Munich, Boltzmannstr. 15, Garching 85748, Germany, caglar.guerbuez@tum.de), Martin Eser, and Steffen Marburg (Chair of Vibroacoustics, Tech. Univ. of Munich, Garching, Germany)

Accurate predictions are generally accompanied with high costs and expenses. As a conesquence, system observations can only be generated for a few samples. The idea of multi-fidelity modeling aims to merge observations from different models of varying complexities and costs. This contribution introduces multi-fidelity models applied to acoustic problems. Therefore, we deploy models with different fidelity levels to treat the frequency-dependent Helmholtz equation. While the spatial solution is acquired with the boundary element method, Gaussian processes are used as surrogates to approximate the system's response in the frequency dimension. This way, multi-fidelity models are adopted to efficiently approximate the solution of the Helmholtz equation for a certain frequency range. For validation purposes, the low frequency booming noise problem occurring in vehicle cabins is treated. Our findings demonstrate that multi-fidelity models yield an accurate approximation. In addition, the computational costs are reduced when compared with the high-fidelity solution at each frequency. In the sense of a Bayesian technique, multi-fidelity models based on Gaussian processes allow to consider uncertainties. Beyond fast and accurate predictions, the proposed method paves the way for accelerated decision-making processes in early design stages, where uncertainties due to limited information on the model or simplifying assumptions are omnipresent.

9:00–9:20 Break

9:20

3aCA4. Investigating sound absorption in rail tunnels using wave decomposition. Shahrokh Sepehrirahnama (Univ. of Technol. Sydney, 32-34 Lord St., Techlab, Botany, New South Wales 2019, Australia, shahrokh.sepehrirahnama@uts.edu.au), Sebastian Oberst (Ctr. for Audio, Acoust. and Vib., Univ. of Technol. Sydney, Ultimo, New South Wales, Australia), Briony E. Croft, and David Hanson (Acoust. Studio Pty Ltd, Stanmore, New South Wales, Australia)

Acoustic noise in trains is a more prevalent problem in tunnels as compared to open track scenarios. This is mainly due to less acoustic radiation relative to increasing contributions of reflections and reverberation. Sound absorbing panels on a tunnel wall and in the track four-foot perform better above 1 kHz with absorption coefficients larger than 0.8. To investigate sound absorption below 1 kHz, we employ wave decomposition into incidence, reflection and absorption components for a section of a given underground tunnel design. A Finite Element (FE) 2D model of a carriage and tunnel is developed, representing a tangent portion of a rail track and including the noise power spectra from the rail-wheel interactions for three different roughness scenarios. The FE model, compared to a Ray Tracing one, provides precisely imposed boundary conditions and the pressure field of the entire tunnel interior. Our results can identify the performance of current panels, absorbing significantly less noise power in the lower frequency range, especially within the 0.3–0.5 kHz interval. The insights from wave decomposition analysis can lead to solutions to increase absorption by changing the reflection pattern below 1 kHz band, improving the passenger comfort during a longer train ride.

9:40

3aCA5. Tangent linear approximations for split-step Padé solutions of the parabolic-equation method in two dimensions. Brandon M. Lee (Mech. Eng., Univ. of Michigan, 1231 Beal Ave. Ann Arbor, MI 48109, leebm@umich.edu) and David R. Dowling (Naval Architecture and Marine Eng., Univ. of Michigan, Ann Arbor, MI)

The parabolic-equation (PE) method is a popular technique in underwater acoustics for obtaining 2D and 3D acoustic-field predictions in spatially-dependent environments. Tangent linear models for these PE solutions relate perturbations in the environmental properties (inputs) to corresponding changes in the acoustic field-solutions (outputs), enabling efficient schemes for approximately transferring parametric uncertainties through to acoustic-field uncertainty or computing the sensitivity of the acoustic-field to these parameters for adjoint-based acoustic inversions. Previously, first-order and higher-order tangent linear models have been developed with respective square-root operator splitting schemes for use with finite-difference and split-step Fourier PE solutions. This presentation describes the findings of an investigation into using these previously developed tangent linear models with the popular split-step Padé method in 2D problems. Additionally, two new tangent linear models are proposed based on directly perturbing: (1) the combined exponential operator in the analytical solution of the one-way wave equation and (2) the split-step Padé solution itself. The forward errors of these tangent linear models are compared analytically and across a variety of computational examples. The numerical stability of each approach is also important and of interest. Finally, the computational savings (if any) from using each approximation over the original split-step Padé model are discussed.

10:00

3aCA6. Weakly nonlinear ray tracing approximations for focused ultrasound propagation. Matt Foster (Comput. Sci., Univ. College London (UCL), 90 High Holborn, Fl. 1, London WC1V 6LJ, United Kingdom, matthew.foster.20@ucl.ac.uk), Marta Betcke (Comput. Sci., Univ. College London (UCL), London, United Kingdom), Ben Cox (Medical Phys., Univ. College London (UCL), London, United Kingdom), and Bradley E. Treeby (Medical Phys., Univ. College London (UCL), London, London, United Kingdom)

Focused ultrasound is used in a therapeutic treatment (HIFU) and uses ultrasound waves to non-invasively destroy malignant cells inside the human body. The technique works by sending a high-energy beam of ultrasound into the tissue using a focused transducer. Numerically modeling HIFU presents a problem due to nonlinear effects leading to the formation of harmonics of the source frequency. Each harmonic requires a finer grid to resolve, rapidly increasing computational complexity. This work will focus on the derivation and benefits of two ray tracing methods using weakly nonlinear ray theory formed by different asymptotic expansions of the governing acoustic equations. The first has the ray equations identical to those from linear ray theory while the amplitude equation is a nonlinear transformation of the Burgers' equation. In the second method the Eikonal and transport equations are coupled which results in ray trajectories which depend on the amplitude.

Session 3aEA

Engineering Acoustics, Signal Processing in Acoustics and Physical Acoustics: Sound Field Manipulation for Personal Audio

Jiaxin Zhong, Cochair
Pennsylvania State University, Graduate Program in Acoustics, College of Engineering, University Park, PA 16802

Sipei Zhao, Cochair
Centre for Audio, Acoustics and Vibration, University of Technology Sydney, 32-34 Lord Street, UTS Tech Lab, Botany, 2019, Australia

Jing Lu, Cochair
Nanjing University, 504 Acoustical Building, 22th Hankou Road, Nanjing 210093, China

Chair's Introduction—7:55

Invited Papers

8:00

3aEA1. Sound field navigation with virtual higher-order sound sources using complex greedy pursuit algorithm. Shaoheng Xu, Thushara Abhayapala (Audio & Acoust. Signal Processing Group, Australian National Univ., Canberra, Australian Capital Territory, Australia), and Jihui (Aimee) Zhang (Inst. of Sound and Vib. Res., Univ. of Southampton, Bldg. 13, Highfield, Southampton SO17 1BJ, United Kingdom, Aimee_jihui.zhang@soton.ac.uk)

As the Virtual Reality (VR) and metaverse industries continue to expand rapidly, there is a growing need to capture and recreate real-world experiences in immersive audio-visual scenes. Sound field translation, which activates equivalent virtual sources to recreate spatial sound fields, enables users to navigate through these scenes seamlessly. However, combining recordings from different microphones and processing mixed fields of exterior and interior sources present significant challenges. This paper introduces a novel method for virtual navigation by sparsely decomposing complex sound fields using distributed virtual higher-order sound sources and an iterative complex greedy pursuit algorithm. By combining spatially separated microphone recordings, the technique identifies its sparse representation with several higher-order virtual sources. This process effectively decomposes the complex sound field into a grid of higher-order sources, leveraging the power of the complex greedy pursuit algorithm. Through extensive experimentation, we demonstrate the suitability of our method for applications such as VR navigation and sound field reproduction with binaural devices. The results showcase enhanced realism and accuracy, offering users a truly immersive audio experience in virtual environments. As the demand for realistic audio-visual scenes grows, this innovative approach holds promise for advancing VR technology and enriching user experiences in the metaverse.

8:20

3aEA2. A grating lobes suppression method for a steerable parametric array loudspeaker. Fanyi Fan, Yunxi Zhu (Inst. of Acoust., Chinese Acad. of Sci., Beijing, China), and Jun Yang (Inst. of Acoust., Chinese Acad. of Sci., No. 21 Bei-Si-Huan-Xi Rd., Beijing 100190, China, jyang@mail.ioa.ac.cn)

A parametric array loudspeaker (PAL) is a highly directional sound source with small aperture size. To achieve accurate sound projection, the steerable PAL is implemented by the use of the phased array technology. In this paper, the sound field model based on the Westervelt equation of a practical steerable PAL is presented, and a quasi-linear approximation is adopted to obtain the spatial sound pressure distribution. By applying the phased array technology, the audio beam can steer to the target direction. However, the grating lobe arises when the transducers are spaced more than half a wavelength, which disturbs the effect of beam steering. We further propose the grating lobes suppression method and apply it to the steerable PAL. Simulation results validate the performance of the proposed method.

8:40

3aEA3. Experimental evaluation of bilateral Ambisonics-based binaural room transfer function synthesis with application to personal sound zones. Yue Qiao (Mech. and Aerosp. Eng., Princeton Univ., H101 von Neumann Hall, Princeton, NJ 08540, yqiao@princeton.edu) and Edgar Choueiri (Mech. and Aerosp. Eng., Princeton Univ., Princeton, NJ)

This study experimentally evaluates the bilateral Ambisonics method for synthesizing binaural room transfer functions (BRTFs) and explores its application in generating personal sound zones (PSZs) around listeners' ears. Bilateral Ambisonics is proposed for improving the spatial sound reproduction accuracy at a limited Ambisonics order, by shifting the origin of Ambisonics representation from the head center to the two ears. While numerical simulations have demonstrated its superiority over traditional Ambisonics, little attention has been given to validating its usability for reproduction in realistic environments with loudspeakers. Experiments are conducted wherein the BRTFs of a dummy head listener are first synthesized by combining the spatial room transfer functions (SRTFs) measured with a spherical Ambisonics microphone array at the ear position and the corresponding head-related transfer functions (HRTFs), and then compared with the actual *in situ* measured ones. Furthermore, the quality of the synthesized BRTFs is examined in the context of an in-house PSZ system, by objectively evaluating the performance of PSZ filters designed with such BRTFs in terms of acoustic isolation metrics. The application of synthesized BRTFs to PSZ systems circumvents the need for *in-situ* BRTF measurements, making it highly feasible to deploy such systems while achieving high acoustic isolation.

9:00

3aEA4. Identification of the parametric array loudspeaker system using differential Volterra filter. Yunyao Ma (Inst. of Acoust., Chinese Acad. of Sci., No. 21 Bei-Si-Huan-Xi Rd., Beijing 100190, China, mawenyao@mail.ioa.ac.cn), Jun Yang, Yunxi Zhu (Inst. of Acoust., Chinese Acad. of Sci., Beijing, China), and Zheng Kuang (Audfly Technol. Corp., Suzhou, China)

The self-demodulation property of a parametric array loudspeaker (PAL) system can peel off audible sound modulated on ultrasonic wave and deliver it with high directivity, but also introduces nonlinear distortion. The PAL system can be modeled by the Volterra series expansion in any acoustic field region. However, measurements of Volterra kernel using simple frequency sweep excitation signals may not provide adequate information about the nonlinear system, and the use of wideband signals such as speech signals makes convergence difficult. Based on the Westervelt model, we propose a PAL system identification method utilizing the differential Volterra kernel. The results demonstrate superior fitting performance for real swept frequency output data and speech signal output data compared to the existing one-dimensional Volterra filter (ODVF). Additionally, the proposed method exhibits fewer parameters and faster convergence speed than the traditional volterra kernel.

9:20

3aEA5. Optimizing acoustic contract control target for sound zones in an enclosed space. Qiaoxi Zhu (Ctr. for Audio, Acoust. and Vib., Univ. of Technol. Sydney, UTS Tech Lab, 32/34 Lord St., Botany NSW 2019, Botany, New South Wales NSW 2019, Australia, qiaoxi.zhu@uts.edu.au) and Sipei Zhao (Ctr. for Audio, Acoust. and Vib., Univ. of Technol. Sydney, Botany, New South Wales, Australia)

Acoustic contrast control is fundamental to achieving the highest sound energy difference between listening and quiet zones within a space. While extensively studied in simulations and laboratories, its real-world application has been confined to engineered applications, for example, acoustically treated rooms. However, for a broader application, we notice significant degradation in acoustic contrast performance from the laboratory prototype to the actual room application. To mitigate this disparity, we propose a physically optimized target to enhance room acoustic contrast control. The proposed method is evaluated by offline simulations using real-world room impulse responses.

9:40

3aEA6. Physics-informed neural network assisted spherical microphone array signal processing. fei ma (Ctr. for Audio, Acoust. and Vib., Faculty of Eng. and IT, Univ. of Technol. Sydney, 15 Broadway, Ultimo, Sydney, New South Wales 2007, Australia, feima1024@gmail.com), Sipei Zhao (Ctr. for Audio, Acoust. and Vib., Faculty of Eng. and IT, Univ. of Technol. Sydney, Botany, New South Wales, Australia), and Thushara Abhayapala (School of Eng., The Australian National Univ., Canberra, Australian Capital Territory, Australia)

Thanks to their rotational symmetry that facilitates three-dimensional signal processing, spherical microphone arrays are the common array apertures used for spatial audio and acoustic applications. However, practical implementations of spherical microphone arrays suffer from two issues. First, at high frequency range, a large number of sensors are needed to accurately capture a sound field. Second, the accompanying signal processing algorithm, i.e., the spherical harmonic decomposition method, requires a variable radius array or a rigid surface array to circumvent the spherical Bessel function nulls. Such arrays are hard to design and introduce a scattering field. To address these issues, this paper proposes to assist a spherical microphone array with a physics-informed neural network (PINN) for three-dimensional signal processing. The PINN models the sound field around the array based on the sensor measurements and the acoustic wave equation, augmenting the sound field information captured by the array through prediction. This makes it possible to analyze a high frequency sound field with a reduced number of sensors and avoid the spherical Bessel function nulls with a simple single radius open-sphere microphone array.

10:00–10:20 Break

10:20

3aEA7. Acoustic radiation characteristics according to the vibration area of OLED panel speakers. Hyungwoo Park (ICT, Dong-Seoul Univ., 76, Bockjeong-ro, Sujeong-gu, Seongnam-si, Gyeonggi-do 13117, South Korea, michael.park@du.ac.kr), Sungtea Lee, and Kwanho Park (LG Display, Paju, South Korea)

Previous studies have introduced the potential of flat OLED panels to be utilized as multiple-channel speakers. Subsequent research has also focused on improving sound quality and implementing multi-channel audio system in OLED TV. For personal displays using OLED panels find diverse applications, offering comfortable and accessible viewing for OTT services with top-notch video and audio quality. Moreover, they excel in delivering crucial audio information during various gaming experiences. In this study, we present a film-based Cinematic Sound OLED (CSO) technology, directly vibrating a single flat OLED panel, which extends the vibrational area compared to previous studies and can reproduce two or more independent sounds. In particular, the proposed method transmits information by generating sound directly on the screen. This method is as sophisticated, accurate, and realistic as the method using headphones. Headphones or earphones generate sound close to the human auditory system, potentially harming auditory health. However, the proposed method aligns the focus of the display and sound, maintaining a certain distance from the user's auditory system. In this paper introduced the advantages of the proposing method and discuss a personal high-performance display device and an appropriate advanced audio generation method.

10:40

3aEA8. Active attenuation of free field noise in a spatial domain using neural networks inferred controller. Ramy Kila (Mech. Eng., San Jose State Univ., 1778 Hamilton Ave., San Jose, CA 95125, ramy.kila@sjsu.edu) and Feruza Amirkulova (Mech. Eng., San Jose State Univ., San José, CA 95192-0087, CA)

In this talk, we explore the application of deep neural networks (DNNs) to infer the decoupling of an incoming acoustic array, distinguishing between the signal of interest and noise components. The objective is to leverage this decoupling information to develop an effective controller that can attenuate the noise, thereby improving the quality of the desired signal. The decoupled noise is then fed into a specially designed controller to attenuate the noise while preserving the desired signal. The controller leverages the DNN's predictions to adaptively adjust its parameters, allowing it to adapt to changing noise conditions and improve noise reduction performance. The approach is evaluated using simulations and real-world acoustic array measurements. The results demonstrate the performance of the DNN-based decoupling and noise attenuation system in reducing noise while preserving the quality of the desired signal. The proposed system offers effective adaptive solutions for enhancing the quality of desired signals in the presence of noise, with potential applications in areas such as speech recognition, audio communication systems, and environmental noise suppression.

11:00

3aEA9. A study on improving the sound radiation capability of stepped-plate parametric array loudspeakers. Beomseok Oh (Mech. Eng., Pohang Univ. of Sci. and Technol., Bldg. 5, Pohang 37673, South Korea, bs.oh@postech.ac.kr), Chayeong Kim, Dongwoo Lee, Junsuk Rho, and Wonkyu Moon (Mech. Eng., Pohang Univ. of Sci. and Technol., Pohang-si, Gyeongsangbuk-do, South Korea)

It is well known that Parametric Array Loudspeakers (PALs) produce highly directional audible sounds. However, their widespread adoption is hindered by their high cost, which is primarily attributable to the use of an array of ultrasonic radiator units. To address this issue, our research group introduced stepped-plate parametric array loudspeakers (SPPALs) in 2018. SPPALs utilize a single-structure directional ultrasonic radiator, featuring steps of half-wavelength thickness on the radiating plate. This design mitigates the problem of poor directivity, especially at high frequencies, caused by destructive interference resulting from the plate's flexural mode. Moreover, SPPALs can offer cost advantages compared to commercial PALs due to their single structure. However, understanding the vibration characteristics of SPPALs in relation to sound radiation performance is quite challenging due to their various design parameters. In this study, we present an analytical approach that offers a quantitative method for enhanced sound radiation capability. We present a simplified vibration model (SVM), which enables an efficient design process based on a comprehensive analysis of vibration characteristics. By minimizing reliance on finite element method (FEM) simulations, our method expedites the design of various directional speaker systems.

11:20

3aEA10. Head-related transfer functions upsampling with physics-informed spherical convolutional neural network. Xingyu Chen (Australian National Univ., sw2012 74 Chendler St., Belconnen ACT, Australian Capital Territory 2616, Australia, u6256034@anu.edu.au), fei ma (Ctr. for Audio, Acoust. and Vib., Faculty of Eng. and IT, Univ. of Technol. Sydney., Canberra, Australian Capital Territory, Australia), and Prasanga N. Samarasinghe (Australian National Univ., Canberra, Australian Capital Territory, Australia)

Head-related transfer functions (HRTFs) play a crucial role in virtual acoustics and spatial audio applications. However, obtaining personalized, high-resolution HRTFs remains challenging due to the time-consuming and expensive measurements. Recently, deep learning-based methods have shown promise in predicting high-resolution HRTFs from sparse measurements. Nevertheless, most of these methods often treat HRTF upsampling as an image super-resolution task, which overlooks critical spatial information and acoustic principles, leading to overfitting on datasets. This paper proposes a physics-informed spherical convolutional neural network for HRTF upsampling. First, spherical convolutional layers are used to capture spatial features on the sphere, allowing efficient handling of spherical sampled HRTF data. Second, in the upsampling process, the proposed method incorporates the Helmholtz equation as a constraint, adhering to the physics of the acoustic system. This method ensures the generation of physically feasible HRTF interpolations, thus promoting better generalization.

11:40

3aEA11. Subjective speech enhancement performance of a hearable with transparency and spatial frequency-weighted dereverberation. Reza Ghanavi (School of Elec. and Information Eng., The Univ. of Sydney, Bldg. J03, Mailbox 53, Sydney, New South Wales 2006, Australia, reza.ghanavi@sydney.edu.au) and Craig Jin (School of Elec. and Information Eng., Univ. of Sydney, Sydney, New South Wales, Australia)

Preserving natural binaural cues combined with directionality and robust dereverberation is crucial for successful sound enhancement via hearables. In this study, we subjectively evaluate the performance of a real-time hearable prototype that aims to enhance frontal speech and spatial cues in complex environments using a new dereverberation algorithm and acoustic transparency. Our real-time algorithm exploits the spatial frequency-weighted coherence-to-diffuse ratio estimated between the conchal microphones and produces directional binaural gains using a Wiener filter for sound enhancement. The new hearable system can be customized or adapted to a given environment for optimum performance in different frequency bands and preserves natural binaural cues for all individuals by symmetrical spatial filtering. We use the real-time prototype to evaluate the perceived speech intelligibility in noise by a Matrix framework for normal-hearing subjects in a reverberant classroom. We also use a multi-stimulus with hidden reference and anchor-like framework to evaluate the perceived speech quality and listening comfort of several stimuli simulated for HATS in the same environment. Results show that speech intelligibility and listening comfort in noise can be significantly improved for the real-time hearable using the new algorithm with optimized parameters compared with several state-of-the-art coherence-based algorithms implemented in a similar device.

Session 3aMU

Musical Acoustics: Indigenous Musical Instruments

Andrew A. Piacsek, Cochair
Physics, Central Washington University, 400 E. University Way, Ellensburg, WA 98926

Kimi Coaldrake, Cochair
Pan Pacific Technologies, 9 Church Road, Adelaide, 5075, Australia

Chair's Introduction—7:55

Invited Papers

8:00

3aMU1. Investigating the sonic characteristics of New Zealand's Indigenous instruments: *taonga pūoro* and *miheke oro* (Māori and Moriori musical instruments). Jennifer Cattermole (Univ. of Otago, 155 Union St. East, Dunedin 9016, New Zealand, jennifer.cattermole@otago.ac.nz)

This paper investigates the sonic characteristics of *taonga pūoro* from a performer's perspective, as well as on how the instruments' construction and materials affect their sound. It also presents findings from two recent projects that involved making 3D printed models of Indigenous instruments. The first included *taonga pūoro* from the Otago Museum, and the second the only two known Moriori flutes (held in the Bishop and Canterbury museums). These were CT scanned, and that data used to create digital models, which were printed. The first project utilized one type of print media, whereas the second utilized several. Because playing the museum instruments is not permitted, other instruments have been used as controls to test the prints' sonic properties – a *kōauau* (a type of Māori flute) held by the University of Otago in the first project, and replicas of Moriori flutes made from albatross bone (the flutes' original material) in the second. Recorded.wav files of both the original and printed replica instruments were compared using the application Izotope Rx5. This application enables sound files to be viewed as spectrograms, allowing for the harmonic and resonant content of the recordings to be compared visually.

8:20

3aMU2. Didjeridu acoustics: A review. Joseph Wolfe (Phys., UNSW, Sydney, New South Wales, Australia) and John Smith (Phys., UNSW, Phys., UNSW, Sydney, New South Wales 2052, Australia, john.smith@unsw.edu.au)

The didjeridu is the principal musical instrument of the world's oldest continuous culture and its sound is an acoustic icon of Australia. This lip-valve instrument usually produces a single, sustained, low note. The musical interest comes from rhythmic changes in timbre, including those associated with 'circular breathing', wherein exhalation from lungs to instrument with raised velum is alternated with expulsion of air from the inflated cheeks while air is quickly inhaled through the nose, with the velum lowered. This paper reviews research from our lab. Measurements of the acoustic impedance spectrum in the mouth during playing show that impedance peaks produce antiformants or minima in the output sound spectral envelope. The instrument's slight flare and irregular bore geometry produce resonances that do not fall in harmonic ratios. Instruments judged good by players have particularly weak resonances in the formant frequency region so that the vocal tract impedance peaks can dominate the sound spectrum more easily. In one technique, players phonate at frequencies above that of the instrument, producing complicated heterodyne components. Lips are low Q oscillators and, as in other lip-valve instruments, the energetics of lip oscillation depends on the phase difference between the lips' longitudinal and lateral motion.

8:40

3aMU3. The history and prehistory of Asian free reed instruments. James Cottingham (Phys., Coe College, 1220 First Ave. NE, Cedar Rapids, IA 52240, jcotting@coe.edu)

The free reed pipes and mouth organs are among the most significant indigenous musical instruments of East and Southeast Asia. The origin and development of these instruments involve the history and prehistory of a multitude of ethnic groups. The basic principles of operation and construction of the free reed are simple, and it seems likely that similar instruments may have had multiple places and times of origin. A number of examples of these instruments, ranging from very simple to fairly complex. These include mouth resonated lamellophones, a single free reed coupled to a bamboo pipe or buffalo horn resonator, a reed pipe with multiple s finger holes, and multi-pipe mouth organs. These instruments have traditionally been constructed with very simple tools using natural materials, primarily bamboo and wood. Bamboo is often still used for the reeds as well as the pipes, although metal reeds have been common for some time.

9:00

3aMU4. Acoustical affordances and challenges with indigenous and orchestral wind instruments. Jonas Braasch (School of Architecture, Rensselaer Polytechnic Inst., Greene Bldg, 110 8th St., Troy, NY 12180, braasj@rpi.edu)

All our orchestral wind instruments evolved over many centuries from indigenous prototypes. The latter were developed using found hollow objects like animal horns for early brass instruments, vulture bones for flutes, and termite-infested eucalyptus branches for the Australian didjeridu. Already over 40,000 years ago, the concept of finger holes was conceived that allowed musicians to play various scales on instruments. During the Rennaissance and early Baroque, wind instrument ranges were extended to 1 1/2 octaves and beyond on a diatonic scale and extended cross-fingering capabilities, but it was during the romantic period that our orchestral instruments reached their final form, with chromatic extended tonal ranges and known well balanced timbral qualities. Built on an industrial scale, these instruments became a commodity without real alternatives for subsequent music styles, including jazz, rock, and classical avant-garde. These styles often sought a new timbral expressiveness that can go beyond the intended design of orchestral instruments. In this paper, it will be discussed how different wind instrument designs offer(ed) unique opportunities for music genres over time and how the original design of indigenous instruments better meets some of the more recent requirements than our commonly used orchestral instruments.

9:20

3aMU5. Reassessing the anomalous low frequencies in the Japanese koto. Kimi Coaldrake (Pan Pacific Technologies, 9 Church Rd., Adelaide, South Australia 5075, Australia, coaldrake@panpacific.com.au)

A previous study reported on a finite element model of an indigenous Japanese 13-stringed zither (the koto) based on a CT scan. Frequencies above 100 Hz were relatively easy to account for, while those below 100 Hz were more problematic. Low frequencies have also been reported for the piano, another zither of similar length. The model has now been updated in a range of critical areas, including damping. The updates have resulted in a more precise resolution of the model. It allows us to reassess the anomalous low frequencies to determine if they have other rational explanations. This work continues to expand knowledge of the koto's complex resonances and has broader applications for understanding the characteristic sounds of other wooden indigenous musical instruments.

9:40–10:00 Break

Contributed Papers

10:00

3aMU6. The warble mechanism of the native American flute. Michael Prairie (Elec. and Comput. Eng., Norwich Univ., 158 Harmon Dr., Northfield, VT 05663, mprairie@norwich.edu) and Joshua Beeghley (Elec. and Comput. Eng., Norwich Univ., Northfield, VT)

The "warble" is a peculiar sound in many old Native American flutes. It is perceived as a spontaneous oscillation between two octaves of the flute's lowest note at a rate of a few Hz, and spectrograms reveal amplitude modulation of all partials during the warble period. The modulation of the fundamental is assumed to be generated through interference with difference signals generated from detuned upper partials. We propose the detuned partials are temporarily locked with natural resonance frequencies that are inharmonic due to the construction characteristics of the flute, namely the bore length and the chimney configuration at the sound hole. Natural resonances of several flutes were measured, and then the fundamental note was played using a mechanical blower that slowly increased pressure to produce the warble. Spectrograms showed the harmonic spectrum increase in frequency until the onset of the warble as the third partial locked to the third natural resonance frequency. The results were analyzed using the Alder equation to describe locking and unlocking of the third partial with the natural resonance frequency to establish conditions for warbling. Proper placement of the third resonance frequency is essential to enable the warble, and the necessary construction details are discussed.

10:20

3aMU7. Control of natural resonance frequencies in warbling native American flutes. Joshua Beeghley (Elec. and Comput. Eng., Norwich Univ., 158 Harmon Dr., Northfield, VT 05663, jbeeghle@gmail.com) and Michael Prairie (Elec. and Comput. Eng., Norwich Univ., Northfield, VT)

The natural resonances in Native American flutes are inharmonic, and in many instances the arrangement of these frequencies can support production of a traditional warble. The bore dimensions and the configuration of a chimney around the sound hole affect those resonance frequencies. In this work the passive frequencies of flutes configured with five bore lengths and 13 chimney depths were investigated to characterize how the parameters control the placement of those frequencies, with emphasis on conditions that enable the warble. Each configuration was excited by a pure tone from an external speaker in a simple anechoic chamber. A microphone capsule was positioned near pressure antinodes and the excitation frequency was adjusted to maximize peaks and measure the first five resonant frequencies. Open tubes were tested as well. End corrections for each resonant frequency were determined and plotted versus frequency and wavelength. Analysis revealed nonlinearly decreasing end corrections versus frequency at the mouth as well as at the open foot, with the relative placement of the inharmonic resonance frequencies relative to the fundamental note being a function of bore length and chimney depth. Finally, parameters that favored warbling were identified, informing the relationship between flute design for warble production.

10:40

3aMU8. Why bundengan musicians always drench their instruments prior to playing. Gea O. Parikesit (Performance Arts and Visual Arts Studies | Dept. of Nuclear Eng. and Eng. Phys., Universitas Gadjah Mada, Jalan Grafika 2, Yogyakarta, D.I. Yogyakarta 55281, Indonesia, gofparikesit@ugm.ac.id) and Indraswari Kusumaningtyas (Dept. of Mech. and Industrial Eng., Universitas Gadjah Mada, Yogyakarta, D.I. Yogyakarta, Indonesia)

Bundengan is an indigenous musical instrument from Indonesia. It evolved from a bamboo-woven dome originally used by duck herders as a portable shelter during rain and sun. By embedding strings and bars into the dome, the herders turned this shelter into a unique instrument, under which they sit and play music. Bundengan musicians have learned from experience that the instrument sounds better when played under the rain. Nowadays, they use various ways to drench their instruments prior to playing, such as soaking it in fish ponds or putting it under the shower. We explore the physics of why wetter bundengan sounds better. First, we analyze the construction steps of the bamboo dome: the weaving of the bamboo splits, the bending of the woven splits, the layering of bamboo culm sheaths on the woven splits, and the deformation of the sheaths during wetting. Second, we analyze the coupling between the mechanical vibrations in the dome and the

acoustical waves from the instrument. Certain conditions of the coupling may lead to an evanescent acoustical field dominating over the radiating acoustical field, which may explain why the best bundengan sound is confined only to under the dome, where the musicians sit.

11:00

3aMU9. A comparative analysis of Violin and Erhu: Examining the differences and similarities through statistical analysis of multiple musical excerpts. Wenyi Song (Dept. of Comput. Sci. and Eng., The Hong Kong Univ. of Sci. and Technol., Clear Water Bay, Kowloon, Hong Kong NA, Hong Kong, wsongak@cse.ust.hk), Zeyu Huang (Dept. of Comput. Sci. and Eng., The Hong Kong Univ. of Sci. and Technol., Sai Kung, New Territories, Hong Kong), and Andrew B. Horner (Dept. of Comput. Sci. and Eng., The Hong Kong Univ. of Sci. and Technol., Hong Kong, Hong Kong)

Recent work has compared the different emotional characteristics between the violin and erhu on *the Butterfly Lovers Concerto*. In this study, we investigate whether the previous studies' results hold generally in Chinese and Western classical music. Building upon previous research, we hypothesize that the violin conveys more positive emotions, while the erhu is perceived as sadder. We also expect the violin to be better at conveying high-arousal excerpts. To test these hypotheses, 46 subjects were presented with 14 excerpts, with each excerpt represented by at least four different performances of both instruments. For each performance, subjects were asked 4 binary questions: whether it was happy, sad, agitated, and calm. Results show that the erhu is consistently perceived as sadder than the violin, while the violin is perceived as happier, calmer, and more agitated when significant differences appear. Linear regression analysis suggests that instrument is a more significant factor in the emotional perception of low-arousal excerpts than high-arousal ones. Additionally, performance was a less important factor than instrument on affect perception, and made more of a difference on low-arousal than high arousal excerpts. Meaning, there was more variation between different performances of the same except on low-arousal excerpts than high-arousal excerpts.

11:20

3aMU10. On the sound of a large 3D-printed and assembled musical instrument: The case of the yi?aki. Gea O. Parikesit (Performance Arts and Visual Arts Studies | Dept. of Nuclear Eng. and Eng. Phys., Universitas Gadjah Mada, Jalan Grafika 2, D.I. Yogyakarta 55281, Indonesia, gofparikesit@ugm.ac.id), Jon McCormack, Jing Fu, Yeonuk Kim (Monash Univ., Melbourne, Victoria, Australia), Anthea Skinner (Univ. of Melbourne, Melbourne, Victoria, Australia), John Carty (South Australian Museum, Adelaide, South Australia, Australia), Will Robertson (Univ. of Adelaide, Adelaide, South Australia, Australia), Brian Djangirrawuy Gumbula-Garawirrtja (Univ. of Melbourne, Melbourne, Victoria, Australia), and Aaron Corn (Univ. of Melbourne, Melbourne, Victoria, Australia)

Nowadays 3D printing technology has allowed humanities scholars and cultural materials conservators to create replicas of indigenous musical instruments, with which they can perform various studies. However, the size of the 3D printed objects are limited by the size and the range of movements of the 3D printer. Hence, large musical instruments have to be fabricated in parts and then carefully assembled prior to playing. Here we investigate the yi?aki, a large pipe-like indigenous Australian wind instrument that is played by vibrating the lips while breathing circularly on the mouthpiece. Traditionally, a yi?aki is made from the trunk of hardwoods hollowed by termites. We scanned the internal and external shape of a traditionally-made yi?aki, built a digital model from the scanning data, printed the model in several parts, and assembled the parts into a replica yi?aki. This replica was printed using sintered nylon and assembled using a two-part epoxy. Both the original and the replica yi?akis have been played by musicians and the resulting sound data have been analysed. Our results show that, even though the replica generates yi?aki-like sounds as expected, there are subtle differences between the sound of the original and the replica.

Session 3aNSa

Noise and Physical Acoustics: Duct Noise Control- New Physical Mechanisms and Structures

Lixi Huang, Cochair
Mechanical Engineering, Univ. of Hong Kong, Hong Kong

Paul Williams, Cochair
University of Technology Sydney, 15 Broadway, Ultimo, 2007, Australia

Chair's Introduction—7:55

Invited Paper

8:00

3aNSa1. Broadband noise absorption by time-varying devices. Xue Han (The Univ. of Hong Kong,Pok Fu Lam Rd., Central And Western District, Central And Western District, Hong Kong, hanxue929@connect.hku.hk), Ying Hu (The Univ. of Hong Kong. Pokfulam Hong Kong, Hong Kong SAR, Central And Western District, Hong Kong), Xiaocong Zhu (School of Mech. Eng., Zhejiang Univ., Hangzhou City, Zhejiang Province, China), and Lixi Huang (The Univ. of Hong Kong. Pokfulam Hong Kong, Hong Kong SAR, Hong Kong, China)

A resonator array can broaden the effective sound absorption bandwidth by using the structure with multiple cavities in parallel or in series in space. However, this kind of fixed mechanical structure cannot flexibly adapt to different noise sources, which severely limits the development prospect of the resonator array in practical applications. In this paper, a time-varying resonator is proposed based on a shunted electrical circuit, realizing the equivalent effect of multi-cavities in space. Based on the working principle of spatial and temporal resonators, the theoretical models are established, and the three designs for achieving broadband effects are analyzed theoretically. Through time-domain numerical calculation, the sound absorption performance of the two spatial designs and the temporal design with the same cavity volume is obtained. This study focuses on the scheme design and performance optimization of the temporal resonator. Compared with the traditional spatial resonator, the proposed temporal resonator has the advantages of tunable frequency and high adaptability to varying sources while achieving similar or even better sound absorption performance.

Contributed Paper

8:20

3aNSa2. Acoustic metaliners for insulating noise in a duct with varying cross-sectional area. Dohaeng Kim (Mech. Eng., Korea Adv. Inst. of Sci. and Technol., 291, Daehak-ro, Yuseong-gu, Daejeon 34051, South Korea, dohaeng.kim@kaist.ac.kr) and Wonju Jeon (Mech. Eng., Korea Adv. Inst. of Sci. and Technol., Daejeon, South Korea)

We propose a sound-insulating acoustic metaliner for a duct with varying cross-sectional area with consideration of the flow that varies along the direction of flow and is not a fully-developed. The metaliner comprises an array of unit cells based on two different types of Helmholtz resonators and installed on the duct wall to preserve ventilation performance. To consider the effect of flow on the sound insulation performance of the metaliner, we establish an effective impedance model in term of the friction velocity, enabling the consideration for flows that are not fully-developed. The geometrical parameters of unit cells of the metaliner are designed individually considering the flow around each unit cell. The sound insulation performance of the designed metaliners for different flow speeds with the target frequency of 900 Hz is verified experimentally.

Invited Papers

8:40

3aNSa3. Advancing flow-duct noise control: Examples of innovative silencers. Stefan Jacob (Inst. for Acoust. and Dynam., TU Braunschweig, Bundesallee 100, Braunschweig 38116, Germany, stefan.jacob@ptb.de) and Mats ébom (The Marcus Wallenberg Lab. for Sound and Vib. Res., Royal Inst. of Technol., Stockholm, Sweden)

Two examples of innovative silencers for advanced duct noise control are presented. The first example is related to using metamaterials and the so-called 'slow sound' to create a compact virtual Herschel-Quincke tube [1]. The second example revolves around a concept called the 'modal filter' [2], which is designed to selectively damp a single propagating mode. For plane waves, optimal propagational damping is achieved by a wall impedance equal to the Cremer impedance. We will demonstrate how to compute the Cremer impedance, even for low frequencies, at which the classical solution is invalid [3]. Such an arrangement efficiently dampens both the plane (0,0) and the first radial mode (0,1). To further dampen the first two circumferential modes (1,0) and (2,0), a section with the Cremer impedance can be combined with a modal filter made of micro-perforated plates. These plates are inserted parallel to the flow direction and positioned at modal velocity maxima in the duct cross-section. We anticipate that both examples can be useful in industries requiring compact noise mitigation solutions in flow ducts. [1] https://doi.org/10.1016/j.jsv.2019.115045. [2] https://doi.org/10.1121/2.0000473. [3] https://doi.org/10.1121/1.5136952

9:00

3aNSa4. Sound absorption characteristics of a compact meta-structure under grazing flow. Ying Li (Mech. Eng., The Hong Kong Polytechnic Univ., Hong Kong, Hong Kong) and Yatsze Choy (Mech. Eng., The Hong Kong Polytechnic Univ., The Hong Kong Polytechnic University, Hong Kong 999077, Hong Kong, mmyschoy@polyu.edu.hk)

This study proposes a compact meta-structure for broadband and low-frequency sound absorption under grazing flow. It comprises a microperforated panel (MPP) and resonators with embedded tunable orifices and C-shaped cavities. The sound absorption characteristics of the proposed absorber are investigated via theoretical model (acoustic impedance circuit) and numerical model (finite element method). Consistent predicted results from both models manifest that the proposed meta-structure possesses good absorption capacity under a deep-subwavelength scale, compared to the conventional MPP absorber. To simultaneously achieve better low-frequency and broadband noise damping, a parallel configuration of the meta-structure and an MPP absorber is studied numerically and experimentally. Besides, notable variation on the acoustic performance of the proposed absorber is observed in case of grazing flow, which leads to the decline of the absorption peak and makes the peak wider than that without flow. The practical significance of this study is that it proposes a promising meta-structure for wide frequency sound attenuation applications; furthermore, it elucidates the flow effects on the acoustic performance of the meta-structure, providing a route to reduce flow-related noise in a broad frequency range.

9:20–9:40 Break

9:40

3aNSa5. Mechanism of slow-wave generation and tuning in a perforation-modulated sonic black hole structure. Sihui Li (Dept. of Mech. Eng., The Hong Kong Polytechnic Univ., 11 Yucai Rd., Hung Hom, Kowloon, Hong Kong 100872, China, sihui222.li@connect.polyu.hk), Xiang Yu, and Li Cheng (Dept. of Mech. Eng., The Hong Kong Polytechnic Univ., Hong Kong, Hong Kong)

A sonic black hole (SBH) in a retarding duct can be used to manipulate sound waves and achieve sound absorption. The SBH incorporates two indispensable physical processes: wave speed reduction and energy dissipation. In this study, a perforation-modulated SBH (PMSBH) retarding structure is proposed, in which the two physical phenomena can be balanced to enhance the black hole effects by modulating the perforation parameters. To fully investigate the mechanism of slow-wave generation and the effect of perforation parameters, an analytic model is established based on the Wentzel–Kramers–Brillouin (WKB) solutions to the linear acoustic wave equation. By examining the analytical solutions and comparing them with the numerical results obtained from transient finite element simulations, the effect of several important parameters on reducing the sound speed and enhancing sound absorption performance is revealed, offering a set of criteria and physical insights for modulating the parameters to design an optimal slow-sound absorber. Finally, an experiment is conducted to confirm the theoretical studies and demonstrate the performance of PMSBH. This study brings forward the concept of tunable design to improve the performance of SBH structures, which can benefit the design of sound wave manipulation and noise control devices.

Contributed Papers

10:00

3aNSa6. Sound radiation from semi-infinite elliptical ducts with uniform subsonic jets: An analytical approach. Ruichen Wang (Peking Univ., Beijing, China, wangruichen@pku.edu.cn) and Xun Huang (Peking Univ., Beijing, China)

This paper presents an analytical method for modeling the acoustic field radiation from a semi-infinite elliptical duct with a uniform subsonic jet. Elliptical ducts are often used as inlets for turbofan engines to take full advantage of the pre-compression effect of the fuselage and improve the stealth performance of the aircraft. The method accounts for the instability wave inside the vortex sheet induced by the shearing of jet and ambient flow and its effect to sound radiation. The method uses Mathieu functions to describe the incident and scattered sound in the elliptic cylindrical coordinates. A semi-analytical Wiener-Hopf approach with low computational cost is applied to obtain the near and far field solutions. The near field is illustrated by acoustic pressure maps at different modes and circumferential angles. The far field is shown by directivity patterns at single tones. Numerical simulations based on a finite element method are conducted to validate the accuracy of the analytical method. The results show good agreement

between the analytical and numerical solutions. The proposed method can be used to analyze the acoustic characteristics of elliptical ducts with jets, which are relevant for noise control and optimization of turbofan engine inlets.

10:20

3aNSa7. Optimized muffler design and its performance evaluation to environmental noise reduction on classrooms. Dhany Arifianto, Jamiatul Firda (Eng. Phys., Institut Teknologi Sepuluh Nopember, Surabaya, Indonesia), and Muhammad A. Asyraf (Dept. of Eng. Phys., Institut Teknologi Sepuluh Nopember, Surabaya 60111, Indonesia, maasyraf.edu@gmail.com)

Generators powered by diesel engines are widely used as a backup source of electrical energy. However, the generator produces very high noise, which is around 73–110 dBA. In this study, we designed a muffler and optimized to the highest possible noise reduction. We constructed and installed the designed muffler to evaluate the performance an absorptive muffler on the exhaust pipe of the simulated muffler with the installed one. We measured the sound pressure level (SPL) at the source of exhaust gas noise before the proposed muffler installed and after installation, including inside the surrounding classrooms. A survey was also conducted to determine the annoyance response of the teachers and the students. The results show that the presence of an absorptive muffler can reduce noise by 10 dBA and attenuate more at high frequencies (<500–$16000 > 500$–16000 Hz) at the source of the noise and also in the surrounding environment. The annoyance rating for generator exhaust noise is moderate to very disturbing for listeners who are outside the room and moderate for listeners positions in the room.

10:40

3aNSa8. Topology optimization of array of split-ring resonators in two-dimensional acoustic waveguide for low-frequency transmission problems. Wai Kit Lam (Dept. of Bldg. Environment and Energy Eng., The Hong Kong Polytechnic Univ., Rm. ZS801, Block Z, Hung Hom, Kowloon, Hong Kong 999077, Hong Kong, 20039053r@connect.polyu.hk), Anton Krynkin (Dept. of Mech. Eng., The Univ. of Sheffield, Sheffield, United Kingdom), and Shiu Keung Tang (School of Eng., The Univ. of Hull, Hull, United Kingdom)

This work focuses on the topology optimization of an array of split-ring resonators to achieve nearly zero transmission in the low-frequency regime ranging from 500 Hz to 1000 Hz in 2D waveguide, under the condition of normal-incident plane wave and based on a homogenization scheme known as the Effective Medium Approach (EMA). The EMA transforms each individual resonator defined by its outer radius (rout), neck thickness (h), and the slit width (d), into a homogenous medium characterized by frequency-dependent physical parameters. The transmission coefficient (Tcoeff) of the array can thus be easily evaluated without relying on computationally expensive Finite Element (FE) software. Two optimization algorithms, namely "Sequential Quadratic Programming (SQP)" and "Genetic Algorithm (GA)," have been implemented to the abovementioned topological parameters with sum of Tcoeff being the objective function to minimize. Accuracy of the results has also been validated using software COMSOL Multiphysics® and the satisfactory level of accuracy has proven this approach to be a more efficient and time-saving method to conduct topology optimization of split-ring resonators in low-frequency regime. Moreover, it is noteworthy that the optimized dimensions of resonators are over 20 times smaller than the wavelength in the low frequency range where almost-zero transmission is achieved.

Session 3aNSb

Noise: Community and Environmental Noise

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

Contributed Papers

8:00

3aNSb1. Stable distributions for randomly positioned sound emitters in an urban environment. D. Keith Wilson (U. S. Army Engineer Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755, d.keith.wilson@usace.army.mil), Matthew J. Kamrath (U. S. Army Engineer Res. and Development Ctr., Hanover, NH), and Chris L. Pettit (Aerosp. Eng. Dept., US Naval Acad., Annapolis, MD)

Previously [D. K. Wilson, M. J. Kamrath, C. E. Haedrich, D. J. Breton, and C. R. Hart, *J. Acoust. Soc. Am.* **150**(2), 783–800 (2021)], it was suggested that the probability distribution for sound levels in an urban environment could be usefully modeled with N randomly placed, independent sources in a circular region, with the receiver at the center. The sound decays away from the sources by a geometrical spreading law. With these assumptions, the received sound power consists of the sum of N Pareto-distributed random variables. Although an analytical solution for this sum is unavailable, some positively skewed, heavy-tailed distributions, such as the exponentially modified Gaussian distribution, provide reasonable approximations. The present study is motivated by the observation that, in the limit of large N, the Pareto sum must converge to a stable distribution; in particular, for spherical spreading, the limiting distribution is a *Landau* distribution. We have furthermore found that when N is drawn from a Poisson distribution, the Landau distribution is nearly exact for as few as eight sources. Hence, the Landau distribution provides a suitable general model for urban noise and other situations involving multiple, randomly placed sources.

8:20

3aNSb2. Approximation of the power distribution from multiple sound sources in the atmosphere using sums of gamma random variables. Matthew J. Kamrath (US Army Engineer Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755-1290, matthew.j.kamrath@erdc.dren.mil), Max E. Krackow, and D. Keith Wilson (US Army Engineer Res. and Development Ctr., Hanover, NH)

A sum of gamma random variables is a mathematically tractable approach to approximate multiple sources when the power (amplitude squared) of a single source is nearly gamma distributed because the sum can be expressed analytically for a wide variety of cases. Previous work indicates that the gamma distribution is a good two-parameter empirical approximation of received power from a single, elevated sound source in a turbulent atmosphere. The gamma distribution can be conceptualized as a sum of k independent and identically distributed exponential random variables, each with mean m. Here, k and m are called the shape and scale parameters, respectively. Thus, the summation of multiple gamma random variables with the same scale parameter still yields a gamma distribution. When multiple sources can be approximated well using a gamma distribution, then those sources could be grouped together for simplicity and conceived as a single source in a turbulent atmosphere. In addition, multiple authors have derived analytic expressions for the sum of N independent gamma random variables with distinct parameters in terms of the confluent hypergeometric functions. If the gamma random variables are correlated, then the result is still analytic using recurrence relations.

8:40

3aNSb3. Providing acoustical advice for the purpose of strategic land-use planning. Katie Teyhan (Acoust., EMM Consulting Pty Ltd., Level 3, 175 Scott St., Newcastle, New South Wales 2300, Australia, kteyhan@emmconsulting.com.au)

Strategic land-use planning is the process where stakeholders work together to design future land-use to achieve more desirable economic, social or environmental outcomes. There is often uncertainty around the type of acoustical advice required for the purpose of informing strategic land-use planning decisions. There are also limitations to the level of technical detail that can be made available when little is known about the nature or locations of potential future development. EMM Consulting Pty Limited's experience in recent years has provided opportunities to work on various strategic land-use planning projects. This experience has enabled an understanding of what is required by relevant stakeholders to appropriately inform decisions as well as what is possible from a technical perspective. The importance of establishing clear expectations around technical outcomes and uncertainties is highlighted, as well as the importance of collaboration with all relevant stakeholders. A strong understanding of the planning framework the project will be approved and operate within is critical. The importance of having an appropriate authority and/or framework to facilitate the implementation of required noise measures and a plan to transition existing land use is discussed. The benefits and limitations of a Noise Management Precinct, as described in the Noise Policy for Industry (NSW EPA, 2017), are identified. Noise modeling methods and technical assumptions that could be utilized to predict and manage likely impacts are described.

9:00

3aNSb4. A risk-based approach to precinct planning. Lance Jenkin (Acoust., EMM Consulting, Level 3/175 Scott St., Newcastle, New South Wales 2300, Australia, lance@jenkin.co.za) and Rick A. Scully (Acoust., EMM Consulting, Dulwich Hill, New South Wales, Australia)

As urban populations increase more people being affected by noise. Industrial precincts are a key component of the economy, but they are placed near existing resident. Planning is an important tool to manage the noise exposure from these areas. This paper describes the method we developed to help with deciding where noisy activities, and the amount of mitigation required, to reduce the risk of annoyance from the precinct. Our method is based on modeling the impact from a grid of sources overlayed on the precinct boundary. The transfer functions for each source to all sensitive

receiver is calculated using a standard noise propagation model. An optimization algorithm is used to fairly calculate the maximum sound power for each grid point without exceeding the noise management levels. Risk maps are created using contours based on standard industrial sound power densities to define 3 zones: the central, intermediate, and interface. The risk maps are a useful tool for planners for deciding where to locate projects and the amount of mitigation required.

9:20

3aNSb5. Effects of the preference for acoustic stimuli on sleep quality. Shota Maki (Dept. of Medical Eng., Chiba Univ., 1-33, Yayoi-cho, Inage-ku, Chiba-shi, Chiba-ken 263-8522, Japan, shota813@chiba-u.jp), Sho Otsuka, and Seiji Nakagawa (Ctr. for Frontier Medical Eng., Chiba Univ., Chiba-shi, Chiba-ken, Japan)

Since pharmacological treatment for sleep disorders has a risk of addiction, developments of non-pharmacological tools to enhance sleep are needed. Colored noise sounds characterized by the power spectrum, e.g., white, pink, brown, blue, and purple noises, have been used in some non-pharmacological sleep aids. Former reports showed that white and pink noises were effective for sleep, however the mechanisms how such noises promote sleep remain unknown. In this study, we proposed a hypothesis that effects of colored noises are depending on participants' auditory impressions and examined how each participant's preference for the noises affected sleep-quality indices (sleep latency and duration of slow-wave sleep (SWS)). Sleep states of each participant were examined by EEG in three conditions: (i) the most preferred and (ii) the least preferred noises for each subject, and (iii) no stimulation. Sleep experiments were started at 3:00 PM in a soundproof room and the stimulus began to be presented at lights-out and continued for 60 min. In results obtained, there were no significant differences in the sleep quality indices among the three conditions. Since duration of the experiment may not be enough to observe the effect of the stimuli, further investigations are needed after reviewing the experimental conditions.

9:40–10:00 Break

10:00

3aNSb6. Noise dynamics in city nightlife: Assessing impact and potential solutions for residential proximity to pubs and bars. Andy Chung (ASA ESEA Regional Chapter, 20 Collyer Quay #09-01, Singapore, Singapore, ac@kvikoo.com) and Wai Ming To (Faculty of Business, Macao Polytechnic Univ., Macao)

The vibrancy of city nightlife, particularly within pubs and bars, brings with it a nuanced soundscape, with sounds ranging from operational activities to animated conversations. While these spaces serve as pivotal social hubs, the accompanying noise can become a concern for proximate residential areas. This paper provides an in-depth analysis of the noise dynamics inherent to urban nightlife spots, accentuating the implications for those living nearby. It further outlines strategic guidelines designed to alleviate noise disturbances in alcohol-serving venues, promoting a serene urban atmosphere alongside bustling nightlife.

10:20

3aNSb7. Developing a deep learning-based noise annoyance prediction tool for urban environments in Singapore. Prachee Priyadarshinee (Sci., Mathematics and Technol., Singapore Univ. of Technol. and Design, #02-07, 53 Changi South Ave. 1, Singapore 485996, Singapore, prachee@sutd.edu.sg), Jer-Ming Chen (Sci., Mathematics and Technol., Singapore Univ. of Technol. and Design, Singapore, Singapore), and Balamurali B. T. (Sci., Mathematics and Technol., Singapore Univ. of Technol. and Design, Singapore, Singapore)

In this study, we develop a noise annoyance prediction tool using deep learning in Singapore, a densely populated city-state where a significant portion of the population resides in public housing near various noise sources like MRT, highways, bus routes, and construction sites. To investigate short-term annoyance caused by surrounding noises, we created an easily accessible web-based subjective listening test. We created a noise dataset featuring typical Singaporean noise sources, including traffic (buses, highways), trains, aviation, neighbourhood activities (playgrounds, schools, hawker centers, funerals, wildlife, home renovations), and construction. Participants were exposed to 35 noise stimuli, lasting between 15 to 30 seconds, and asked to rate perceived annoyance on a 5-point scale, with 5 being the most annoying. Using these reported annoyance levels, we ranked the stimuli relative to each other and categorized them into three classes: low, medium, and high noise annoyance. Using these three labels, Long-Short Term Memory (LSTM) networks were trained to predict the perceived annoyance of new audio samples. Such a perceived annoyance assessment tool for new audio samples can help in addressing specific noise concerns of residents, leading to more effective and targeted noise control strategies, and consequently creating a quieter and more pleasant urban living environment.

10:40

3aNSb8. Environmental noise predictions using a statistical approach. John H. Wassermann (80 Williams St., Woolloomooloo, New South Wales 2211, Australia, john.wassermann@me.com) and Nic Hall (Wollongong, New South Wales, Australia)

Noise pollution from industrial premises has the potential to have an impact on neighbouring properties. In New South Wales the Environment Protection Authority has developed the Noise Policy for Industry (NPfI). The NPfI requires noise assessments to be conducted using a typical assessment procedure that results in setting project noise trigger levels (PNTLs) which are the benchmark levels against which potential noise impacts from industrial developments are judged. The PNTLs are then used in Environmental Protection Licences to regulate the premises. The NPfI requires noise assessments to consider worst-case (i.e., noise-enhancing) meteorological conditions. In addition, regulators and consent authorities often require noise assessments to include a range of other conservative assumptions such as worst-case source levels and locations. This process appears to be very conservative, particularly considering that the PNTL in each assessment period (day, evening and night) are based on lowest 10th percentile background noise levels. This paper, through a statistical approach of using random probability distributions, will identify the likelihood that modelled noise levels and associated impacts are likely to occur and comment on the appropriateness of the conservative assessment process.

11:00

3aNSb9. Sources of anthropogenic sound in the Svalbard environment. Janusz Piechowicz (Mech. Eng. and Robotics, AGH Univ. of Sci. and Technol. Krakow, Al>Mickiewicza 30, Kraków 30-059, Poland, piechowi@agh.edu.pl), Jerzy Wiciak, Dorota Mlynarczyk, and Pawel Malecki (Mech. Eng. and Robotics, AGH Univ. of Sci. and Technol. Krakow, Krakow, Poland)

Svalbard is one of the few sparsely populated areas of the Earth. Settlement is not favored by the climate, permafrost and difficult weather conditions. But even here, settlements inhabited by people are developing. And when there are people, there is also noise. In our research, we addressed the issue of identifying sources of noise generated in human settlements; traffic noise, industrial noise and tourist noise. We have carried out a number of measurement sessions to determine the sound power of individual sources in the city, as well as the history of temporal changes in A-weighted sound levels in 24-hour periods in Longyearbyen, the capital of Svalbard.

11:20

3aNSb10. Battery energy storage systems environmental noise emission. Linnea Eriksson (WSP Australia, Canberra, Australian Capital Territory, Australia) and Zhang Lai (WSP Australia, Level 2 121 Marcus Clarke St., Canberra, Australian Capital Territory 2601, Australia, zhang.lai@wsp.com)

The use of Battery Energy Storage Systems (BESS) in the electricity grid is rapidly growing due to its ability to bridge the gap between times of energy needs and when certain renewable sources are not generating. The use of battery storage helps the grid to remain stable due to its ability to respond quickly to changes in energy demand. Grid-scale battery storage has the potential to significantly assist in the renewable energy transition. Noise has emerged as a key environmental impact challenge in the

development of BESS. But why? In our work with BESS, the noise is commonly associated with the battery and inverter modules' heating and cooling systems, with the use of fans and compressors being the main emitters. However, the noise levels emitted are highly variable and depend on several factors, including operating conditions, ambient temperatures, and speed drives. We will explore the noise emissions of BESS, and key challenges like: —pathways for mitigating noise, including discussion of options at different project stages, ranging from site selection to commissioning. —challenges in accurate noise emission modeling —explore lessons learnt from a project from the design stage to post construction commissioning testing. Including the noise modeling predictions and their accuracy.

WEDNESDAY MORNING, 6 DECEMBER 2023 ROOM C3.1, 8:55 A.M. TO 12:00 NOON

Session 3aPA

Physical Acoustics and Biomedical Acoustics: Multiphase Flow and Acoustics II

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

Richard Manasseh, Cochair
Mechanical and Product Design Engineering, Swinburne University of Technology, John Street, Hawthorn, Melbourne, 3122, Australia

Joseph (Yeo Cheon) Kim, Cochair
Mechanical Engineering, UNSW, Sydney 2052, Australia

Chair's Introduction—8:55

Invited Papers

9:00

3aPA1. Acoustic cavitation detection in biomedical and underwater systems. Jacob Elliott, Eric Rokni (Graduate Program in Acoust., The Penn State Univ., University Park, PA), Paul Trzcinski, Michael Krane, Jeff Harris (Appl. Res. Lab., The Pennsylvania State Unviersity, University Park, PA), and Julianna Simon (Graduate Program in Acoust., The Penn State Univ., 201E Appl. Sci. Bldg, University Park, PA 16802, jcs516@psu.edu)

In the past decade, significant progress has been made in detecting and localizing cavitation for treatment monitoring in biomedical acoustics. Here, we compare passive cavitation imaging (PCI) and passive cavitation detection (PCD) during histotripsy in tissue-mimicking polyacrylamide (PA) hydrogels; PCI and bubble Doppler ultrasound are also used to evaluate flow-induced cavitation in a water tunnel. A Philips/ATL L7-4 transducer driven with a research ultrasound system was used for both PCI and Doppler imaging; a Sonic Concepts Y-107 transducer was used for PCD. PA hydrogels were treated with 1.5 MHz focused ultrasound (10-ms pulses with $p+ = 127$ MPa/$p- = 35$ MPa repeated at 1-Hz for 60 s). In the 12-in. water tunnel, cavitation on a 1-in. diameter steel cylinder was imaged through a 0.5-inch-thick acrylic window while flow increased from 30–35 ft/s. High-speed cameras were also used in both experiments. In PA hydrogels, cavitation was observed with both PCI and PCD, although signal trends differed over the treatment. In the water tunnel, both PCI and Doppler ultrasound detected and localized cavitation events such as the horseshoe vortex, with measured amplitudes increasing with flow speed. These results show that cavitation imaging can be applied to multiple areas of acoustics. [Tissue work supported by NIHR01EB032860].

9:20

3aPA2. High-speed observation of bubble and droplet dynamics under irradiation of 1-MHz ultrasound. Nobuki Kudo (Faculty of Information Sci. and Technol., Hokkaido Univ., Fac Info Sci Tech, Hokkaido University, N14W9, Kita-ku, Sapporo, Hokkaido 060-0814, Japan, kudo@ist.hokudai.ac.jp)

Simulation techniques have resulted in significant progress in elucidating phenomena that are too fast and small for direct observation; however, there has also been considerable progress in the legacy techniques of direct observation. We have been studying applications of microbubbles for ultrasonically enhanced drug delivery, and in this presentation, we will introduce our results of high-speed observations carried out to elucidate the roles of microbubbles injected into the capillaries of living tissues. Micron-sized bubbles stabilized by a lipid shell were introduced into a limen of 10 microns in diameter created in acrylamide gel, and their dynamics under the conditions of exposure to 1-MHz short-pulsed ultrasound was observed using a high-speed video camera (HPV-X2, Shimadzu, Japan) equipped in a microscope (ECLIPSE Ti, Nikon, Japan) with an objective lens (Plan APO IR 40x, Nikon, Japan). The camera captured 256 frames at a framing rate up to 10 Mfps to visualize bubble dynamics inside a narrow capillary that will lead to ultrasound-enhanced extravasation. Oscillation of a water droplet of a few millimeters in size in air, which is the opposite situation of bubble oscillation in water, will also be discussed in this presentation.

9:40

3aPA3. Bioeffects of microbubbles in tissue engineering: Modeling collapsing jets and microstreaming. Kausik Sarkar (Mech. and Aerosp. Eng., George Washington Univ., 800 22nd St. NW, Ste. 3000(MAE), Washongton, DC 20052, sarkar@gwu.edu)

Ultrasound contrast agents are micron-sized bubbles coated with lipids/proteins to stabilize them in the bloodstream. Apart from enhancing the contrast of the image, they have been implicated in numerous harmful and beneficial bioeffects. I will present an overview of our research emphasizing recent efforts on microbubbles-based non-invasive pressure estimation and ultrasound-assisted bone and cartilage tissue engineering in 3D printed scaffolds. Low-intensity pulsed ultrasound (LIPUS) in conjunction with microbubbles has been shown in our lab to facilitate bone and cartilage formation from mesenchymal stem cells. We will discuss nonlinear shape oscillations of microbubbles and acoustic microstreaming that are responsible for such bioeffects. We studied them using boundary element (BEM) simulation and perturbative analysis of an encapsulated microbubble near a vessel wall. For the BEM solution, the coating of the microbubble is modeled as a viscoelastic interface using an in-house developed strain-softening model (exponential elasticity model). The influence of the shell model on the stability of the numerical simulation during the microbubble jet formation has been investigated. The effects of ultrasound excitation parameters and mechanical properties of the coating, i.e., shell viscosity and elasticity, on the bubble behaviors and the velocity and pressure in the surrounding fluid, have been studied.

10:00

3aPA4. Bubbles as high-frequency rheometers for soft matter characterization. Guillaume Lajoinie (Phys. of Fluids Group, TechMed Ctr., Univ. of Twente, Enschede, the Netherlands, g.p.r.lajoinie@utwente.nl), Alex Oratis, Ali Rezaei, Kay Dijs (Phys. of Fluids, Univ. Of Twente, Enschede, the Netherlands), Michel Versluis (Univ. of Twente, Enschede, the Netherlands), and Jacco Snoeijer (Phys. of Fluids, Univ. Of Twente, Enschede, the Netherlands)

Precise knowledge of bubble dynamics in tissue is essential for optimizing both ultrasound imaging and ultrasound therapy. Great efforts have therefore been made, over the past decade, to model the behavior of (micro)bubbles in viscoelastic materials. This has led to a series of models based on modified Rayleigh–Plesset equations. Last to date, we propose a novel generalized model build on finite strain theory and relaxation functions. At this stage however, the bottle neck remains our limited knowledge of tissue viscoelasticity at high strain rates, which is a necessary input to any theoretical model. Indeed, traditional rheometers cannot reach frequencies above 1 kHz, which is 3 orders of magnitude below the desired MHz frequencies. We will show how ultrasound-driven bubbles can be used as microscale rheometers. More specifically, we use the volumetric oscillations of (monodisperse) phospholipid-coated microbubbles embedded in hydrogels to measure the storage and loss moduli in the MHz frequency range. In addition, we can bridge the gap between kHz and MHz using the surface modes of cylindrical microbubbles trapped in micropits. These oscillations can be both modelled and simulated, and provide access to the effective surface tension and viscosity in the range of 50 to 500 kHz.

10:20–10:40 Break

10:40

3aPA5. The relevance of bubble dynamics in ultrasonic/sonochemical processes. Muthupandian Ashokkumar (School of Chemistry, Univ. of Melbourne, Victoria 3010, Australia, masho@unimelb.edu.au)

Acoustic cavitation bubble dynamics has been extensively studied by physicists and mathematicians. Process intensification is primarily studied by chemical engineers. Chemists tend to focus on bubble dynamics in a multibubble field to optimize the physical and chemical forces generated during acoustic cavitation with an intention to maximise/intensify chemical processes. To achieve process intensification successfully a multidisciplinary approach is required. Our early work on multibubble cavitation unveiled various factors that contribute to process optimization. For example, while single bubble dynamics can estimate the maximum bubble temperatures, they may not be relevant to achieving higher chemical yield, which in turn depends on varus other factors such as average bubble temperatures, number of active cavitation bubbles, etc. Other factors critical for process intensification include the generation of a homogeneous cavitation field, control over mass transfer effects caused by bubble oscillations, reactor design, etc. The presentation will provide an overview on the relevance of single and multibubble bubble dynamics in ultrasonic/sonochemical processes.

11:00

3aPA6. Preliminary study to scale up microbubble generation with acoustic wave. Minuk Jung (Minerals and Energy Resources Eng., Univ. of New South Wales, Unit 477, 83-93 Dalmeny Ave., Sydney, New South Wales 2018, Australia, minuk.jung@unsw.edu.au), Joseph (Yeo Cheon) Kim, Ghislain Bournival, and Seher Ata (Minerals and Energy Resources Eng., Univ. of New South Wales, Sydney, New South Wales, Australia)

This research focuses on the utilization of acoustic wave to control the generation of microbubbles from a capillary tube, aiming for mass production with porous media. Initially. ultrasonic standing wave is employed to generate small-size microbubble since high amplitude is required for the pipette with smaller pore size which has high resistance. However, for large bubble generation, secondary acoustic radiation force between bubbles is increased and induces bubble coalescence. Pulsing wave can reduce the force strength by reducing wave cycles (apparent frequency) and increasing the distance between bubbles by controlling bubble generation frequency. Finally, maximum bubble generation frequency is calculated at each bubble size based on calculation of limiting factors such as gas permeability and secondary acoustic radiation force. The research will contribute to generate wide size range of multiple bubbles in aqueous solution and set up its limitation regarding bubble generation frequency through theoretical calculation. The data will be utilized for multiple bubble generation with porous media.

11:20

3aPA7. Numerical investigation of bubble pinch-off dynamics. Chinthaka Jacob (Mech. and Product Design Eng., Swinburne Univ. of Technol., John St., Hawthorn, Victoria 3122, Australia, chinthaka.ravinatha@gmail.com), Richard Manasseh (Mech. and Product Design Eng., Swinburne Univ. of Technol., Melbourne, Victoria, Australia), Andrew Ooi (Dept. of Mech. Eng., Univ. of Melbourne, Melbourne, Victoria, Australia), Filippo Nelli (Mech. and Product Design Eng., Swinburne Univ. of Technol., Hawthorn, Victoria, Australia), and Shaung Jie Zhu (Dept. of Mech. Eng., Royal Melbourne Inst. of Technol., Bundoora, Victoria, Australia)

While several mechanisms have been conjectured for the bubble pinching-off process, the underlying mechanism is still poorly understood, leaving the connection between the observed kinematics and the amplitude of the sound generated during the process unclear. Despite some experimental studies reporting the kinematics and the corresponding amplitude of the bubble sound emissions simultaneously (Nelli *et al*., AFMC, 2022), minimal numerical studies have been conducted to supplement these experiments. We report a numerical study by reproducing incompressible bubble pinch-off interface kinematics. Numerical simulations were performed by the volume-of-fluid (VoF) method, which keeps track of each phase fraction using a solution of the interface advection equation. Simulations are in good agreement on the kinematics at pinch-off despite several challenges associated with high-speed, high-distortion-rate interfacial dynamics.

11:40

3aPA8. Higher-order statistical moments of predicted sonic boom waveforms through turbulence. Joel B. Lonzaga (NASA Langley Res. Ctr., 2 N. Dryden St. (MS 463), Hampton, VA 23681, joel.b.lonzaga@nasa.gov)

Sonic booms generated by supersonic aircraft are affected by turbulence in the atmospheric boundary layer through which they propagate. Turbulence effects lead to random variability of the sonic boom waveforms measured on the ground, complicating the prediction of such waveforms. As an initial effort to predict the waveform variability, the solution of the coherent or mean sonic boom waveform has previously been formulated by the author. The current paper extends the formulation to derive an expression of the second-order statistical moment necessary to calculate the variance of the spectral amplitudes. Since the derivation uses a full wave equation, results using the derived expression will be compared with those obtained using a parabolic approximation. In order to fully quantify the uncertainty of our predictions, the higher order statistical moments are also formulated and are shown to be approximately zero. Consequently, the probability density function (pdf) of the spectral amplitudes is predicted to be Gaussian. The formulation is further extended to determine the pdf of the loudness of sonic booms and quantify the uncertainties associated with the loudness prediction. Results from the formulation are compared with available flight test data.

Session 3aPP

Psychological and Physiological Acoustics: Auditory Sensory Augmentation

Craig Jin, Cochair
School of Electrical and Information Engineering, University of Sydney, Maze Crescentm Bldg, J03, Sydney 2006, Australia

Anastasia Devana, Cochair
3110 NE 27TH ST Unit 201, Ft Lauderdale, FL 33308

Chair's Introduction—7:55

Contributed Paper

8:00

3aPP1. Challenges in providing augmented hearing for individuals with hearing loss. Douglas S. Brungart (Walter Reed, 4401 Holly Ridge Rd., Rockville, MD 20853, dsbrungart@gmail.com), Virginia Best (Speech, Lang. and Hearing Sci., Boston Univ., Boston, MA), and Alyssa Davidson (Walter Reed, Bethesda, MD)

Recently there has been an emphasis on developing augmented reality systems that can enhance the sensory signals listeners normally perceive in the real world. An early example of a practical augmented hearing system is the hearing aid, which amplifies the signals that would normally enter the ear in order to improve their audibility for hearing impaired listeners. Many applications are now envisioning the use of virtual audio technology to add synthetic, spatialized sounds to the "passthrough" audio signal that would normally be produced by a hearing aid. One challenge for the designers of both augmented and virtual audio systems is to ensure they preserve the ability of both normal and hearing-impaired listeners to determine the spatial locations of sounds in the environment. The auditory cues required for localizing sounds are well understood for normal hearing listeners, but less so for hearing impaired listeners. Here we discuss the results from two experiments where hearing-impaired listeners performed substantially worse than expected with virtual or augmented audio cues, and present preliminary results suggesting that the unexpectedly poor results from hearing-impaired listeners may be the result of a different weighting of localization cues across frequency regions than is typically observed for listeners with normal hearing.

Invited Paper

8:20

3aPP2. A binaural room impulse response dataset and Shorelining psychophysical task for the evaluation of auditory sensory augmentation. Huiyuan Sun (School of Elec. and Information Eng., Univ. of Sydney, Sydney, New South Wales, Australia, huiyuan.sun@sydney.edu.au), Minh Nguyen, Howe Zhu, Vincent Nguyen, Chin-Teng Lin (Faculty of Eng. and Information Technol., Univ. of Technol., Sydney, Ultimo, New South Wales, Australia), and Craig Jin (School of Elec. and Information Eng., Univ. of Sydney, Sydney, New South Wales, Australia)

Sensory augmentation using spatial sound presented in augmented-reality (AR) can assist people with low vision or blindness in navigating their environment (Katz *et al*., 2012). Nonetheless, in many situations, the poor quality of the binaural sound rendered using current tool sets limits the potential capability of the assistive technology. In particular, acoustic environments with near-field sources and reflections pose significant challenges. In this work, we provide a reference binaural room impulse response (BRIR) dataset with near-field sources and reflections and an associated shorelining psychophysical task that is useful for the evaluation of AR spatial audio. The dataset consists of 12∼small loudspeakers arranged on a 3-by-4 grid in a complex acoustic environment. BRIR measurements are recorded using the Head and Torso Simulator (HATS) for 17∼different receiver positions with a 5^{o} angular resolution. Room impulse response measurements are also recorded using the Eigenmike for each of the 17 receiver positions. Using the Razer Anzu smart glasses to render the binaural AR spatial audio, we compare psychophysical performance on the shorelining navigation task using the recorded dataset and various existing binaural AR tool sets.

Contributed Papers

8:40

3aPP3. Evaluation of photogrammetry-based near-field head related transfer function estimation for immersive sound reproduction. Shayikh Hossain (School of Elec. and Information Eng., Univ. of Sydney, Maze Crescent, Bldg. J03, Sydney, New South Wales 2006, Australia, shayikh.hossain@sydney.edu.au), Huiyuan Sun, and Craig Jin (School of Elec. and Information Eng., Univ. of Sydney, Sydney, New South Wales, Australia)

Binaural reproduction algorithms are commonly applied in virtual and augmented reality applications to create immersive audio environments. Nonetheless, flexible near-field binaural rendering remains challenging. In this work, we explore the acoustic accuracy of photogrammetry-based, near-field head-related transfer function estimation. Photogrammetry measurements of the Head and Torso Simulator (HATS) are taken using a photogrammetry rig consisting of 44 synchronized Canon 1300D DSLR cameras mounted on ten aluminium poles arranged in a circle of 0.8m radius. An Apple iPhone 14 Pro mobile phone is used to acquire the ear data. Commercial photogrammetry software is then used to create a 3D mesh from the still images. Finally, numerical acoustic simulations using the fast multipole boundary element method (FM-BEM) technique are applied to the mesh to generate the near-field HRTFs. We present comparisons of acoustically-measured and photogrammetry-based near-field HRTFs.

9:00

3aPP4. Acoustically-mediated orientation and navigation in rooms. Henning Steffens (Medizinische Physik und Akustik, Universität Oldenburg, Oldenburg, Germany) and Stephan D. Ewert (Medizinische Physik und Akustik, Universität Oldenburg, Carl-von-Ossietzky-Str. 9-11, Oldenburg 26129, Germany, stephan.ewert@uni-oldenburg.de)

While the visual system provides the dominating sensory input in sighted humans for awareness, orientation, and navigation, in low-light conditions, smoke, or in visually-impaired and blind individuals, auditory perception becomes important. Acoustic information about the environment includes sounds radiated from sources, as well as reflections and reverberation in enclosed spaces in response to either external sound sources or self-produced sounds (e.g., mouth clicks), referred to as echolocation. Here, we investigated orientation and navigation in typical corridors based on acoustic cues only, using sighted humans without training and real-time virtual acoustics. Virtual sound sources and echo-location with predefined sounds as well as own vocalizations were used with the goal to identify suitable techniques for an acoustically augmented reality (AAR) based mobility aid. A hand-held acoustic pointer, rendering virtual sound sources at the closest wall in the pointing direction was best suited for orientation and navigation. For identification of the room shape, a performance similar to that obtained with a reference (visual) laser pointer could be achieved with the acoustic pointer in specific conditions. In addition to the above acoustic pointer, an AAR-based echolocation pointer with highly directed sound radiation, as achievable in the ultrasonic range, was additionally suggested.

Invited Papers

9:20

3aPP5. An evaluation of various spatial audio rendering and presentation techniques to enhance active navigation with sensory augmentation. Minh Nguyen (Faculty of Eng. and Information Technol., Univ. of Technol. Sydney, 15 Broadway, Uultimo, New South Wales 2007, Australia, minh.t.nguyen-4@student.uts.edu.au), Howe Zhu (Faculty of Eng. and Information Technol., Univ. of Technol. Sydney, Ultimo, New South Wales, Australia), Huiyuan Sun (School of Elec. and Information Eng., Univ. of Sydney, Sydney, New South Wales, Australia), Vincent Nguyen (Graduate School of Health, Univ. of Technol. Sydney, Ultimo, New South Wales, Australia), Craig Jin (School of Elec. and Information Eng., Univ. of Sydney, Sydney, New South Wales, Australia), and Chin-Teng Lin (Faculty of Eng. and Information Technol., Univ. of Technol. Sydney, Ultimo, New South Wales, Australia)

Active navigation is essential in everyday life and refers to the combination of cognition (spatial mapping, path planning, and decision making) and motor-sensory execution (moving and sensing environment). For people who are blind or have low-vision, auditory and tactile sensory augmentation is critical to active navigation. In assistive technologies, binaural spatial audio rendering is widely adopted. However, the most effective methods to support fluent spatial navigation are still being studied. For example, in a previous study, we demonstrated the feasibility of using spatialized earcons to support a shorelining task. In this work, we use the same shorelining task to explore various forms of spatial earcon presentation with a focus on standardization and effectiveness. We also explore the development of an intuitive auditory grammar for spatial and contextual cues. We conduct psychophysical experiments and present experimental measures such as performance time and accuracy, heart-rate variability, and the NASA task load index.

9:40–10:00 Break

10:00

3aPP6. Potential of focusing retroreflectors as passive acoustic beacons for flash sonar. Densil Cabrera (Sydney School of Architecture, Design and Planning, The Univ. of Sydney, Wilkinson Bldg. GO4, Sydney, New South Wales 2006, Australia, densil.cabrera@sydney.edu.au), Jonothan Holmes, Yoshimi Hasegawa (Sydney School of Architecture, Design and Planning, The Univ. of Sydney, Sydney, New South Wales, Australia), Shuai Lu (Tsinghua Univ., Shenzhen, China), Huiyuan Sun, and Craig Jin (School of Elec. and Information Eng., Univ. of Sydney, Sydney, New South Wales, Australia)

Human echolocation, in the form of flash sonar, can support spatial awareness and navigation for vision-impaired people. This paper examines the potential of acoustic focusing retroreflectors to support echolocation within an area. Parabolic dihedra and trihedra, along with other variants of focusing retroreflectors, return much stronger reflections to an arbitrarily located source than equivalent flat surfaces. A head-and-torso simulator was used to quantify the oral-binaural reflected energy for various surfaces, including physical focusing retroreflectors. The effect of focusing is most advantageous in relatively close-range scenarios (within a few metres). Suggested designs and application scenarios are presented.

10:20

3aPP7. Auditory sensory augmentation to support table tennis games for people with vision loss. Phoebe Peng (School of Elec. and Information Eng., Univ. of Sydney, Maze Crescent, Bldg. J03, Sydney, New South Wales 2006, Australia, phoebe.peng@sydney.edu.au), Huiyuan Sun (School of Elec. and Information Eng., Univ. of Sydney, Sydney, New South Wales, Australia), Alexandre Marcireau (ICNS, Western Sydney Univ., Sydney, New South Wales, Australia), Minh Nguyen, Howe Zhu, Chin-Teng Lin (Faculty of Eng. and Information Technol., Univ. of Technol., Sydney, Ultimo, New South Wales, Australia), and Craig Jin (School of Elec. and Information Eng., Univ. of Sydney, Sydney, New South Wales, Australia)

People with vision loss often face limitations in regular sports games with standard rules and equipment. For example, in current blind table tennis, conventional rules are modified so that the ball rolls along the table instead of bouncing. In this work, we propose an auditory sensory augmentation system to support traditional table tennis in three dimensions. We capture the trajectory of the table tennis ball using two neuromorphic event cameras and sonify the path of the ball using loudspeakers mounted near the left and right edges of the playing table. The two event cameras capture rapid changes in brightness allowing fast and precise ball tracking. The ball's 3D trajectory is then sonified using four lines of loudspeakers mounted at two different heights near the left and right edges of the playing table. We present a preliminary implementation and investigation of the proposed sensory augmentation system with a focus on the technical and perceptual challenges.

10:40

3aPP8. Frequency dependent emotional responses to sound: A cross-modal analysis of electroencephalogram band power and emotion perception. Manish Kumar (School of Eng., The Australian National Univ., Canberra, Australian Capital Territory, Australia), Thushara Abhayapala (School of Eng., The Australian National Univ., 115 North Rd., Canberra, Australian Capital Territory 2601, Australia, thushara.abhayapala@anu.edu.au), and Prasanga N. Samarasinghe (School of Eng., The Australian National Univ., Canberra, Australian Capital Territory, Australia)

Enabling machines to comprehend human emotions serves as the motivating factor for our research. This effort significantly contributes to advancing emotion recognition and human-computer interaction. Our study explores the fundamental associations between emotional responses and auditory characteristics, with a primary focus on frequency. To achieve this, we conducted a two-stage perception test, aiming to identify the interplay between EEG band power and dominant emotion class as a function of frequency. Musical notes, ranging between 110 Hz and 973 Hz, served as our stimuli. During the first stage, we collected participants' EEG data, categorized it into distinct energy bands (Alpha (α), Beta (β), Delta (δ), Theta (θ), and Gamma (γ)), and analyzed band power and ratios. In the subsequent stage, we had participants evaluate their emotional responses to each stimulus. Our results from this two-stage perception test suggest that an increased tangent of the α / β band power ratio corresponds to Low-Arousal emotions, while a diminishing tangent correlates with High-Arousal emotions. Furthermore, we observed a crossover point for four primary emotions within the 417–440 Hz frequency range. This finding supports the hypothesis that the 432–440 Hz range is emotionally neutral.

Contributed Paper

11:00

3aPP9. Personal authentication system utilizing voice distortion through guitar effectors. Keiichi Zempo (Inst. of Systems and Information Eng., Univ. of Tsukuba, 1-1-1 Tennodai, Tsukuba 305-8573, Japan, zempo@iit.tsukuba.ac.jp), Suzuha Harada, Takayuki Kawamura (Graduate School of Sci. and Technol., Univ. of Tsukuba, Tsukuba, Japan), Tadashi Ebihara, and Naoto Wakatsuki (Inst. of Systems and Information Eng., Univ. of Tsukuba, Tsukuba, Japan)

In this study, we propose a personal authentication system using processed speech that can only be understood by those who know the original voice. The proposed system presents non-invertible processed speech signals, including distortion, equalization, and echo, to the voice representing the words registered by the legitimate user at the time of authentication, just like a signal processed through guitar effects pedals. The strength of the effect is set within the range where the legitimate user can barely be identified, and the combination of effect parameters can be changed each time it is presented, making it harder for everyone other than the legitimate user to break through. We confirmed the system's effectiveness by utilizing a dataset of Japanese words with defined word familiarity. Through an authentication experiment in which the system was automatically adjusted to an effect strength that was barely recognizable to legitimate users, it was found that legitimate users could identify 75.6% of the processed speech signals. In comparison, non-legitimate users could decode only 38.8% of the words. These results indicate the feasibility of a personal authentication system utilizing distorted speech signals.

Session 3aSA

Structural Acoustics and Vibration, Engineering Acoustics and Physical Acoustics: Acoustic Metamaterials III

Christina Naify, Cochair
Applied Research Labs: UT Austin, 10000 Burnet Ave., Austin, TX 78758

Nathan Geib, Cochair
Applied Research Laboratories, University of Texas at Austin, 1587 Beal Ave., Apt. 13, Ann Arbor, MI 48105

Samuel P. Wallen, Cochair
Applied Research Laboratories and Walker Department of Mechanical Engineering, The University of Texas at Austin, 10000 Burnet RD, Austin, TX 78758

Chair's Introduction—7:55

Invited Paper

8:00

3aSA1. Inverse design of acoustic coatings using Bayesian inference. Karthik Modur (School of Mech. and Manufacturing Eng., UNSW, Sydney, New South Wales 2052, Australia, z5061725@ad.unsw.edu.au), Abhinav Koncherry (School of Mech. and Manufacturing Eng., UNSW, Sydney, New South Wales, Australia), Cikai Lin (School of Mech. and Manufacturing Eng., UNSW, Kensington, New South Wales, Australia), Jonas M. Schmid, Caglar Gurbuz, Steffen Marburg (Chair of Vibroacoustics, Tech. Univ. of Munich, Garching, Germany), Gyani S. Sharma, Alex Skvortsov, Ian MacGillivray (Platforms Div., Defence Sci. and Technol. Group, Melbourne, Victoria, Australia), and Nicole Kessissoglou (School of Mech. and Manufacturing Eng., UNSW, Kensington, New South Wales, Australia)

A Bayesian approach for the inverse design of acoustic coatings for underwater noise control is presented. The coating model comprises a periodic arrangement of voids embedded in a soft viscoelastic material. The viscoelastic material is attached to a steel backing with water on its incidence side and air on its transmission side. The acoustic performance of the coating is strongly dependent on the geometric properties of the voided inclusions as well as the number of layers of voids in the direction of sound propagation. The geometric optimization process using Bayesian inference proceeds inversely from a target absorption coefficient spectrum to the required number of voided layers, the geometric design parameters in each layer, and the distance between each layer. The Bayesian design process demonstrates that broadband sound absorption can be achieved using a multilayered coating with a gradient change in both void diameter and distance between each layer. Optimized designs using the Bayesian approach are validated against results obtained numerically as well as experimental results from the literature.

Contributed Papers

8:20

3aSA2. Elastic metamaterials with gradings and their structural applications. Andrea Colombi (Zurich Univ. of Appl. Sci., Gebäude MD, Tössfeldstrasse 13, Winterthur 8406, Switzerland, colo@zhaw.ch)

Graded designs allowing a progressive wave manipulation have been a cornerstone for the design of functional structures. Either based on local resonance, Bragg scattering or a mix of both, the performance of graded meta go much beyond the so-called bandgap, the original hallmark of elastic metamaterials. I will present different applications where elastic metamaterials featuring graded design have been studied and exploited for different purposes and scales. After introducing early proposals with EM and acoustic waves, applications on graded resonant metasurfaces will be showcased. These include both fundamental laboratory scale experiments but also civil and mechanical engineering-oriented applications for vibration energy harvesting and vibration suppression. I will then show the latest experimental efforts in modeling, optimising and 3D printing lightweight frame-based panels for structural engineering applications. These frames combine lightweight, bioinspired shapes with graded resonant elements and represent a promising design model for reimaging the design of structural members such as trusses, beams and shells integrating vibration control and application in structural engineering and architecture.

8:40

3aSA3. Lightweight soundproofing meta-panel for two separate broadband noises. Jiwan Kim (Dept. of Mech. Eng., Korea Adv. Inst. of Sci. and Technol., 291, Daehak-ro, Yuseong-gu, Daejeon 34141, South Korea, jiwan.kim@kaist.ac.kr) and Wonju Jeon (Dept. of Mech. Eng., Korea Adv. Inst. of Sci. and Technol., Daejeon, South Korea)

We propose a lightweight soundproofing meta-panel, with a multi-scale lattice structure sandwiched between thin membranes, effectively suppressing two separate broadband noises. By adjusting the geometrical parameters, the meta-panel is designed to induce negative effective mass density and negative effective bulk modulus for two distinct target frequency bands: 100–500 Hz and 1000–1500 Hz, thereby achieving high transmission loss that outperforms the mass density law. A meta-panel sample is fabricated via 3D printing technology and we evaluate its sound insulation performance by measuring transmission loss and insertion loss in an impedance tube and an anechoic chamber. Experimental measurements show that the meta-panel can exhibit excellent insulation performance while being significantly lighter in weight (about 5%) compared to a steel plate, and furthermore, it effectively suppresses low-frequency road noise and high-frequency motor noise from an actual electric car.

9:00

3aSA4. Deep-subwavelength phononic beams with topological interface states. Seongmin Park (Mech. Eng., Korea Adv. Inst. of Sci. and Technol., 291, Daehak-ro, Yuseong-gu, Daejeon, Daejeon 34141, South Korea, kaistian13@kaist.ac.kr) and Wonju Jeon (Mech. Eng., Korea Adv. Inst. of Sci. and Technol., Daejeon, South Korea)

We propose deep-subwavelength phononic beams that can support topological interface states (TISs) at low frequencies. To achieve this, we employ unit cells with acoustic black hole (ABH) configurations, enabling the creation of the first band gap in a low-frequency range where the unit cell size is much smaller than the wavelength of the controlled wave. Connecting two distinct beams with different topological properties results in TISs within their low band gaps. Through numerical simulations and experimental validation, we demonstrate the realization of TISs in the frequency range of 4.3–8.5 Hz, utilizing the designed phononic beams with lattice constants less than $\lambda/20$.

9:20

3aSA5. The integration of metamaterials into building elements. Andrew Hall (Mech. and Mechatronics Eng., Univ. of Auckland, Auckland, New Zealand, a.hall@auckland.ac.nz), Vladislav Sorokin, Emilio P. Calius, Gian Schmid, and George Dodd (Mech. and Mechatronics Eng., Univ. of Auckland, Auckland, New Zealand)

The building industry continues to rely on homogeneous materials for acoustic insulation despite the progress of research into acoustic metamaterials. With the inevitable densification of housing, the severity of noise pollution within residential living environments is escalating. While the insulation of high-frequency audible sound through building elements is often relatively good between 1 and 5 kHz, the overall acoustic transmission loss performance is often significantly limited by two specific frequency regions. The mass air mass resonance band, and the coincidence band. We present the results of an investigation into the use of metastructures and metasurfaces to improve transmission loss in these frequency regions with a focus on scalable implementation. These metastructures are metamaterial systems constructed from impedance change elements, surface variations and vibroacoustic resonant elements. The performance of selected systems from this research are presented. Experimental and modeling results are in good qualitative agreement and promising diffuse-field testing results indicate significant attenuation within the targeted band regions. The merits of each technique are analysed, and results indicate which methods are most effective at either mitigating or shifting the regions of poor transmission loss outside of the most important part of the audible frequency region.

9:40–10:00 Break

10:00

3aSA6. Asymmetric elastic wave scattering in elastic plates. Samuel D. Parker (Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX 78712, sdparker@utexas.edu), Daniel R. Roettgen (Structural Dynam., Sandia National Labs., Albuquerque, NM), and Michael Haberman (Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX)

Reciprocity is a fundamental principle that can be used to understand a wide range of wave phenomena. In reciprocal systems, the exchange of source and receiver between two points yields the same displacement at the receiver if the applied force at the source is the same for both configurations. The reciprocity principle enforces restrictions on the scattering matrix in an acoustic or elastic system, but these relationships are not obvious when multiple modes are present, particularly when modes are coupled due to the presence of asymmetric scatterers. In this talk, we apply the reciprocity principle in a simplified two-dimensional waveguide case to prove symmetry of the scattering matrix in passive systems with and without loss and establish requirements for asymmetric scattering. The requirements enforced by reciprocity and energy conservation are then generalized to Lamb modes of arbitrary order. Applications to acoustic and elastic metamaterial design and ultrasonic nondestructive evaluation are discussed. [Sandia National Laboratories is a multimission laboratory managed and operated by National Technology and Engineering Solutions of Sandia, LLC., a wholly owned subsidiary of Honeywell International, Inc., for the U.S. Department of Energy's National Nuclear Security Administration under contract DE-NA-0003525.]

10:20

3aSA7. Optimization of a structural Acoustic Black Hole embedded in a stringer-stiffened composite laminate panel. Anna Moorhouse (Graduate Program in Acoust., Penn State Univ., Appl. Sci. Bldg., Graduate Program in Acoust., University Park, PA 16802, aqm6654@psu.edu), Avery Brown (Eng. Sci. and Mech., Penn State Univ., State College, PA), Benjamin Beck (Appl. Res. Lab, Penn State Univ., State College, PA), and Micah Shepherd (Phys., Brigham Young Univ., Provo, UT)

Many modern airframes are made of composite laminate materials due to their low mass but strong loadbearing capability. Carbon fiber composite laminates allow for a strong and lightweight aircraft, but create unhealthy levels of disruptive and uncomfortable noise within the fuselage. This work looks at the possibility of a structural Acoustic Black Hole (ABH) as a passive vibration damping solution for a common stringer-stiffened airframe panel. A 1-D symmetrically damped ABH was integrated vertically into the cross section of a stringer stiffener of a composite laminate plate. A multi-objective evolutionary algorithm was employed to search for the optimal ABH dimensions in terms of the tradeoff between weight, axial linear buckling load, and integrated vibration response. Computational results predicted the ability for damped ABH-stiffeners of a certain dimension to have less vibrational response than others with comparable linear density and axial strength. Mass normalized, non-tapered and ABH-tapered preliminary panels were manufactured and tested. These results showed a capability of ABH stiffeners to decrease vibrational response in the attached skin, more prominently above their cut-on frequency. Stiffened panels were manufactured to optimized specifications and their experimental results validate the presence of an optimal ABH dimension to achieve passive vibration damping in a system with mass and structural constraints.

10:40

3aSA8. Analysis on sound absorption in Sonic Black Holes. Xiaoqi Zhang (School of Naval Architecture, Ocean and Energy Power Eng., Wuhan Univ. of Technol., No. 1178, Heping Ave., Wuchang District, Wuhan, Hubei 430063, China, zhangxiaoqi@whut.edu.cn), Nianwen He (School of Naval Architecture, Ocean and Energy Power Eng., Wuhan Univ. of Technol., Wuhan, Hubei, China), Li Cheng, Xiang Yu (Dept. of Mech. Eng., The Hong Kong Polytechnic Univ., Hong Kong, Hong Kong), and Linke ZHANG (School of Naval Architecture, Ocean and Energy Power Eng., Wuhan Univ. of Technol., Wuhan, China)

Exploration of Acoustic black holes (ABHs) for sound wave manipulation is referred to as Sonic black holes (SBHs). Different from the ideal SBH structure with continuously varied wave propagation boundary formed by putting extensive number of rings inside, the wave trapping and energy dissipation mechanism of practically realized SBH structure with discrete varied wave propagation boundary formed by quite limited number of rings inside are still not clear. To fill the gap, finite element simulations are performed to assess the sound absorption performance of a practical discrete SBH structure and explore the sound absorption mechanism behind. The validity of the FEM results is demonstrated through comparisons with experimental data obtained through impedance tube tests. Analyses reveal that the physical mechanism underpinning the sound absorption of practically realized SBH device is attributed to the combined effects of the ABH-induced wave speed changes, energy trapping and the spatially graded local resonances of series cavities formed by the space between two adjacent rings. While shedding light on the underlying sound attenuation mechanism, studies provide guidelines for further investigations on SBH structures.

11:00

3aSA9. Damping vibration of platform structure using modified acoustic black holes. Taehwan Son (Mech. Eng., Korea Adv. Inst. of Sci. and Technol., 291, Daehak-ro, Yuseong-gu, Daejeon 34141, South Korea, taehwan.son@kaist.ac.kr), Seongmin Park (Mech. Eng., KAIST, Daejeon, South Korea), and Wonju Jeon (Mech. Eng., Korea Adv. Inst. of Sci. and Technol., Daejeon, South Korea)

We propose waveguide absorbers (WGAs) to dampen structural vibrations of platforms supporting vibrating systems. The waveguide absorber is designed by modifing a spiral acoustic black hole (ABH) to improve its damping performance while saving installation space and weight. The proposed WGA absorbs the flexural wave propagating in a platform to its end, resulting in vibration damping in the platform. To maximize the damping capability of WGA, we analyze the structural intensity field of the platform and attach the WGA based on the intensity field. Numerical and experimental results show that large reductions of peaks in mobility are achieved using the WGAs, showing the possibility to utilize the WGA in practice.

11:20

3aSA10. The acoustic metasphere. A solution to improving speech intelligibility and acoustic measurements. Gregory Hernandez (Elec. and Comput. Eng., Duke Univ., 101 Sci. Dr., Durham, NC 27708, gregory.hernandez@duke.edu), Junfei Li (Mech. Eng., Purdue Univ., West Lafayette, IN), and Steven Cummer (Elec. and Comput. Eng., Duke Univ., Durham, NC)

The application of acoustic metamaterials has found utility in several disciplines ranging from biomedical ultrasound to architectural acoustics, and even the music industry. However, an underlying complication with these devices is their narrow bandwidth which limits the effectiveness of acoustic metamaterials in research and industry. Additionally, within architectural acoustics and noise control engineering, an issue that arises is speech intelligibility and clarity of the source. For example, within a populated and dense environment, such as an airport or shopping market, loudspeakers are placed every several feet to surmount the beaming directionality of these transducers at higher frequencies. Also, consequently, the gain must then be high enough to overcome this beaming effect and the noise floor generated by the environment and its occupants. Moreover, methods to more efficiently and cost effectively characterize the response of an acoustic space are needed to improve upon the noise control efforts within these locations. A solution to these obstacles is proposed with an acoustic metamaterial called the Acoustic Metasphere. Utilizing various twisting unit cells that fully envelop a loudspeaker, we demonstrate here that this acoustic metamaterial can improve the limitation of a loudspeaker due to its directionality, while simultaneously increasing the bandwidth of operation for this engineered structure.

11:40

3aSA11. Exploring pulse mitigation in heterogeneous granular metamaterials. Marcos Espinosa Cuartas (School of Eng., Royal Melbourne Inst. of Technol., Melbourne, Victoria, Australia, s3969967@student.rmit.edu.au), Emilio Calius (Computed Materiality, Auckland, New Zealand), Andrew Hall, George Dodd (Mech. and Mechatronics Eng., Univ. of Auckland, Auckland, New Zealand), and Raj Das (School of Eng., Royal Melbourne Inst. of Technol., Melbourne, Victoria, Australia)

Recognizing the need for effective control of vibration and sound propagation in various industries, this study investigates the potential of designing heterogeneous granular networks for vibroacoustic transmission mitigation. It introduces new models of granular systems: decorated, stepped, and tapered 2-level branching structures. The research assesses changes in particle size (5–10 mm radii) and material properties (density and Young's Modulus) to create finely-tuned composites that significantly modulate pulse waves. The discrete element method predicts wave propagation in these granular metamaterials, comparing monodispersed chains, conventional chain networks, and the proposed heterogeneous structures. Their pulse diffusion capacity is evaluated, showing how collective responses can be adjusted by altering physical parameters like particle size and composition. Preliminary findings underscore the utility of these configurations in advancing the development of elastic and acoustic metamaterials, demonstrating a peak amplitude reduction more than five times greater than an equivalent monomer system. With versatility across a wide frequency range, these metamaterials could pioneer a new direction in impact mitigation.

Session 3aSC

Speech Communication: Speech Production I (Poster Session)

Isabel S. Schiller, Chair
Institute of Psychology, RWTH Aachen University, Jaegerstrasse 17-19, Aachen 52066, Germany

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

Contributed Papers

3aSC1. On the nature of Tone 4 alternation in Taiwan Mandarin: Insights from acoustic analysis of trisyllabic right-branching word productions. Chin-Ting Liu (Dept. of Appl. English, National Chin-Yi Univ. of Technol., No. 57, Sec. 2, Zhongshan Rd., Taiping Dist., Taichung 41170, Taiwan, ctjimboliu@gmail.com)

Tone 4 (T4) alternations refer to the phenomenon where the tone values of a T4 syllable reduces from '51' to '53' when it is followed by another T4 syllable in Mandarin Chinese. It is inconclusive if the nature of T4 alternations is tone sandhi (an abstract phonological rule) (Chao, 1948; Jang, 2021) or tonal coarticulation (an articulatory phenomenon in speech production) (Lin, 20007; Shen, 1990). The current study invited 30 Taiwan Mandarin speakers to produce 10 pairs of right-branching trisyllabic words, including sequences of [T4[T4-T4]] (e.g., dà diàn shì 'big television') and [T4[T4-Tone1]] syllables (e.g., dà diàn jiā 'big store'). The T4 sandhi view predicted that the surface tone values of the first syllables in these two sequences would differ ("51i" vs. "53"), while the T4 coarticulation view predicted that the tone values would be the same ("53"). Results from the acoustic analysis on fundamental frequency (f0) contour, f0 slope and vowel length of the first T4 syllables indicated that the differences were not statistically significant. These results supported the tonal coarticulation view. Discussions pertaining to the mental grammar for the tone value distances of two adjacent T4 syllables and the typology of tonal coarticulation in Mandarin Chinese are formulated.

3aSC2. Compositionality in intonation: Evidence from Greek. Amalia Arvaniti (Modern Lang., Radboud, Ctr. for Lang. Studies, Radboud University, Nijmegen, Gelderland 6500 HD, the Netherlands, amalia.arvaniti@ru.nl) and Stella Gryllia (Modern Lang., Radboud, Nijmegen, the Netherlands)

The autosegmental-metrical (AM) theory of intonational phonology posits that tunes are composed of independent elements, pitch accents and edge tones. However, configurational approaches remain popular, while AM relies on but cannot prove compositionality. We addressed this issue with a corpus of Greek wh-questions ($N = 2135$), elicited from 18 speakers using a discourse completion task that involved communicative situations resulting in rise-fall-rises or rise-falls. The rise-fall-rises are autosegmentally represented as a $L^* + H$ pitch accent on the utterance-initial wh-word, followed by a L- phrase accent and a H% boundary tone; the rise-falls are represented as $L + H^*$ $L - L\%$. We applied functional principal component analysis (FPCA), a data-driven method that breaks down curves into components that capture independent modes of curve variation, and followed FPCA with LMEMs on the principal component coefficients. The pitch movement associated with each of the posited tonal elements was captured by a different PC: PC2 captured differences in the extent of the rise and the peak alignment of the pitch accent, PC1 differences in fall steepness, and PC4 the difference between a final rise and low, flat pitch. These results provide prima facie evidence for tune compositionality, since the differences are driven by the analysis not human annotation.

3aSC3. Individual variation in the use of non-F0 cues in the realization of accentual contrasts. Na Hu (Modern Lang., Radboud, Nijmegen, the Netherlands) and Amalia Arvaniti (Modern Lang., Radboud, Ctr. for Lang. Studies, Radboud University, Nijmegen, Gelderland 6500 HD, the Netherlands, amalia.arvaniti@ru.nl)

Previous analysis of Greek H*, $L + H^*$, and $H^* + L$ used FPCA on the f0 contours of these accents ($N = 844$; 13 speakers), and showed that they are distinguished by their PC1 and PC2 scores: PC1 captures differences in pitch height ($H^* < L + H^* < H^* + L$) and PC2 differences in accent shape. We extend this analysis by investigating the use of voice quality (H1-H2) and duration in distinguishing the accents, with a focus on individual variation in the use of these cues. Results obtained from Bayesian multivariate analysis demonstrate considerable variability in the use of voice quality and duration in distinguishing between $L + H^*$ and H*, as compared to $H^* + L$ and H*. Specifically, all speakers used F0 to distinguish $H^* + L$ and $L + H^*$ from H*. In addition, 7/13 speakers employed a breathier voice to distinguish $H^* + L$ from H*, while the use of duration as a cue was relatively infrequent. In contrast, 8/13 speakers employed voicing and 7/13 employed duration to distinguish $L + H^*$ from H*, with most speakers using at least one parameter in addition to F0 to make this accentual distinction. These results highlight the importance of investigating non-F0 cues used in encoding intonation categories, as well as individual variation in such use.

3aSC4. Identification training adapted to the learner's vowel space aids in improving native Japanese speakers' pronunciation of American English vowels. William Martens (National Acoust. Labs., Macquarie Univ., Sydney, New South Wales 2109, Australia, bill.martens@mq.edu.au) and Stephen Lambacher (Social Informatics, Aoyama Gakuin Univ., Sagamihara, Kanagawa, Japan)

Native speakers of Japanese learning English as a foreign language often have difficulty pronouncing certain American English (AE) vowels, such as the mid vowels /ae/, /ɑ/, /ˆ/, /ɔ/, and /ɚ/, which typically are assimilated to native vowel categories. It was shown that vowel identification training which was adapted to the learner's vowel space could aid in improving the native Japanese speakers' pronunciation of a set of 12 AE vowels that included the 5 more difficult mid vowels. Estimates of the vocal-tract length (VTL) for 53 learners were based upon the central tendency and range of the four lowest formant frequencies observed in their recordings of five short statements in the Japanese language. Having thus characterized the individual vowel spaces of 53 learners, 33 were placed in a "longer-VTL" group while the remaining 20 were placed in a "shorter-VTL" group. Those in the "longer-VTL" group were given periodic identification training using examples of 12 American English monophthongs in a consonant-vowel-consonant (CVC) context that sounded as if a male AE speaker with similar VTL had uttered them. The "shorter-VTL" learners heard training examples that sounded as if the CVCs were uttered by a female AE speaker with similarly short VTL.

3aSC5. Making figure models of speech articulators for speech production research. Makoto J. Hirayama (Faculty of Information Sci. and Technol., Osaka Inst. of Technol., 1-79-1 Kitayama, Hirakata 573-0196, Japan, makoto.hirayama@oit.ac.jp)

Physical 3D figure models of speech articulators including tongue, lips, upper and lower jaws with teeth, were made with regin and gel materials, for research and educational purposes of speech production mechanisms. Regin models were made using a 3D printer and gel models were made with hand crafted cray and a molding method. The models focuses on physiological organs and muscles inside a mouth to understand coordinated speech articulator motions driven by multiple muscles of musculoskeletal systems. For upper and lower jaws, and a hyoid bone, 3D models are designed using computer graphics software and transferred to a 3D printer to make regin models. For tongue and tissues inside mouth, cray was used to make handcrafted original model and molding was done using viscoelastic urethane gels to be duplicated. Intrinsic and extrinsic tongue muscles were modeled as one figure model, but some extrinsic muscles connected outside a tongue body were modeled individually. The models are used to teach speech production mechanisms in a university class. By using physical models, they are contributed to understand complex speech articulation mechanisms inside a mouth, more intuitively and easily.

3aSC6. What does it mean to sound Korean-Canadian, eh? A comparative study on Canadian English vowel space. Sylvia Cho (Linguist, Simon Fraser Univ., 8888 University Dr., Burnaby, BC V5A 1S6, Canada, sylvia_cho@sfu.ca)

Existing research on Canadian English shows evidence of dialectal differences between regions. Not only are there differences based on region, but groups (e.g., heritage, immigrant) in these respective regions are not participating in the ongoing sound changes of Canadian English (e.g., Canadian Raising) to the same degree. In respect to sound changes, studies have found that some sounds are undergoing more drastic change than others, young female groups are usually more advanced in the progress, and that the largest heritage groups in major Canadian cities (Toronto, Vancouver, Montreal) are participating in sound changes—with some reported differences in vowel quality. This study extends the existing literature on Canadian sound changes by investigating the acoustic vowel quality of Korean heritage speakers in Vancouver. The speech of 11 monolingual Canadian English speakers and 22 Korean heritage speakers were analyzed for specific sound patterns (/u/-fronting, /æ/ raising) as well as overall vowel space area. This study finds that Korean heritage speakers also participate in the ongoing sound changes (e.g., /u/-fronting) but there are also some notable differences in their vowel acoustics, when compared to their monolingual counterparts.

3aSC7. A theory of vowel dispersion based on probabilistic modeling of optimized speakers and listeners. John McGahay (Linguist., UCLA, 11692 Chenault St., Unit 308, Los Angeles, CA 90049, jmcgahay@g.ucla.edu)

Dispersion Theory (Liljencrants and Lindblom, 1972; Lindblom, 1986) holds that optimal vowel systems maximize perceptual contrast by maximizing psycho-acoustic distance between vowels, successfully predicting much of the acoustic typology of natural languages. Here we explore probabilistic models of optimized speakers and listeners to see if they might yield more accurate and principled predictions about vowel dispersion. Given a set of vowel distributions and a set of categorical perceptual boundaries optimized for those distributions, the optimal set of vowel distributions maximizing the percentage of correctly perceived tokens is not in general the same as the original set of distributions. This motivates an iterated confusion minimization algorithm that can model dispersion as an incremental diachronic process that achieves globally optimized vowel systems by locally optimizing each vowel category against its own perceptual boundaries at each generation. We show this model provides a principled explanation for several nuanced patterns observed in cross-linguistic corpora, including greater distances in F2 than F1 and even spacing of F1 in log(Hz) rather than auditory-based scales (Becker-Kristal, 2010). A new prediction of this model is that lower-probability vowel pairs should be less dispersed; survey work now in progress so far confirms this prediction.

3aSC8. Development of a web application for recording and evaluating intonation in emotionally charged speech and singing. Bozena Kostek (Audio Acoust. Lab., Gdansk Univ. of Technol., Narutowicza 11/12, Gdansk 80-233, Poland, bokostek@audioakustyka.org)

The goal of the work is to develop a web application dedicated to evaluating intonation in emotionally charged speech and singing, recorded according to a given scenario. The starting point is a systematic literature review concerning existing corpora of emotionally charged speech and singing and their accessibility. Then, based on the review performed, assumptions of the web-based application are shown, along with the choice of emotions expressed and the words/texts to be recorded. The realized corpus is characterized by its multimodality. Four modalities are used in recordings, i.e., audio, video, Facial Motion Capture (FMC) system sensors, and a high-speed camera. There are versions of recordings in the form of the audio signal and video signal, respectively: 25 fps (Canon), 120 fps (Vicon), and 200 fps (GoPro), audio and video signals combined, and recordings from the FMC sensor system (C3D files). Important is the fact that the corpus contains recordings by professional and amateur actors who expressed neutral emotion, joy, sadness, and anger. The recorded speech and singing signals are uploaded to a server and then prepared for the intonation analysis employing artificial neural networks with a deep architecture.

3aSC9. Synthesizing medical terms—Quality and naturalness of the deep Text-to-Speech algorithm. Bozena Kostek (Audio Acoust. Lab., Gdansk Univ. of Technol., Narutowicza 11/12, Gdansk 80-233, Poland, bokostek@audioakustyka.org) and Barbara Szyca (Faculty of ETI, Gdansk Univ. of Technol., Gdansk, Poland)

The main purpose of this study is to develop a deep text-to-speech (TTS) algorithm designated for an embedded system device. First, a critical literature review of state-of-the-art speech synthesis deep models is provided. The algorithm implementation covers both hardware and algorithmic solutions. The algorithm is designed for use with the Raspberry Pi 4 board. 80 synthesized sentences were prepared based on medical and everyday language employing the TTS algorithm developed. For tests, an application is built, containing a questionnaire allowing for evaluating the quality and naturalness of the synthesized speech, for both types of language. It is followed by the algorithm efficiency tests. A presentation of the performed tests, along with the results obtained from 30 respondents, is shown. The discussion consists of a statistical analysis of the obtained results and a comparison with other speech recognition solutions used as a reference. Finally, in the summary section, there is an overall conclusion of this approach and promising directions for future development. [This work is supported by the Polish National Center for Research and Development (NCBR) project: "ADMEDVOICE-Adaptive intelligent speech processing system of medical personnel with the structuring of test results and support of therapeutic process," no. INFOSTRATEG4/0003/2022.]

3aSC10. Optimizing medical personnel speech recognition models using speech synthesis and reinforcement learning. Andrzej Czyzewski (Multimedia Systems, Gdansk Univ. of Technol., Narutowicza 11/12, Gdansk 80-233, Poland, andczyz@gmail.com)

Text-to-Speech synthesis (TTS) can be used to generate training data for building Automatic Speech Recognition models (ASR). Access to medical speech data is because it is sensitive data that is difficult to obtain for privacy reasons. Speech can be synthesized by mimicking different accents, dialects, and speaking styles in a medical language. Reinforcement Learning (RL), in the context of ASR, can be used to optimize a model. A model can be trained to minimize errors in speech-to-text transcription, especially for technical medical terminology. In this case, the "reward" to the RL model can be negatively proportional to the number of transcription errors. The paper presents a method and experimental study from which it is concluded that the combination of TTS and RL can enable the creation of a speech recognition model suited to the specific needs of medical personnel, helping to expand the training data and optimize the model to minimize transcription errors. The learning process used reward functions based on Mean Opinion Score (MOS), a subjective metric for assessing speech quality, and Word Error Rate (WER), which evaluates the quality of speech-to-text transcription. [The Polish National Center for Research and Development (NCBR) supported the project: "ADMEDVOICE- Adaptive intelligent speech processing system of medical personnel with the structuring of test results and support of therapeutic process," no. INFOSTRATEG4/0003/2022.]

3aSC11. A description of the acoustic properties of rhotics in Costa Rican Spanish. Sergio J. Salazar Rodó (Modern Lang. and Linguist., Florida State Univ., 1521 Cinnamon Bear Circle, Tallahassee, FL 32311, sjsalazar@fsu.edu)

Studies into the rhotic variation in Costa Rican Spanish have previously utilized perceptive methodologies and focused primarily on phonetic/phonological descriptions of the phenomenon. More recent studies such as Dearstyne (2021) have utilized modern acoustic methodology and have found a system that includes trills, taps, assibilated fricatives and approximants. This study focuses on the latter, seeking to acoustically describe the variant rhotics in this dialect. The methodology is centered on audio-recordings of native speakers from Costa Rica performing a sentence-reading task. The sentences were created to contain rhotics in the contexts in which rhotic variation has been previously documented by Vásquez Carranza (2006). Dependent variables are predicated on the manner of articulation; formants are recorded for approximants while spectral peaks and the center of gravity is taken for assibilated fricatives. Preliminary results for three participants show acoustic values for both qualities that are consistent with the alveolar place of articulation, yet different to other languages and dialects (Quilis 1981, Edwards 1992). (Spectral peak [r̆] = 1,408 Hz, center of gravity [r̆] = 240 Hz, F2 [ɹ] = 1,755 Hz). It is expected that these data will contribute to a better understanding of the rhotic system in this dialect of Spanish.

3aSC12. Utilizing acoustic methods to investigate word-boundary vowel sequences in one Arabic-Spanish contact situation. Sergio J. Salazar Rodó (Modern Lang. and Linguist., Florida State Univ., 1521 Cinnamon Bear Circle, Tallahassee, FL 32311, sjsalazar@fsu.edu), Eden Stafstrom, Miriam Alrahil, and Julia Cambridge (Modern Lang. and Linguist, Florida State Univ., Tallahassee, FL)

In the modern world, the globalized nature of both Arabic and Spanish has led to distinct contact situations in different countries. This investigation focuses on word-boundary vowel sequences, which are typically realized as hiatuses or diphthongs in native Spanish speakers. By contrast, native Arabic speakers will not utilize diphthongs across word boundaries, instead preferring to lengthen one vowel or produce a glottal stop (Mohamed 2020).This investigation examines how these distinct strategies are reconciled in the contact situation of Moroccan immigrants to Spain by utilizing acoustic methods to identify and qualify the bilingual approach to word-boundary vowel sequences. This investigation compares monolingual Spanish speakers ($n = 3$) to two groups of L1 Arabic (Arabic-dominant and Spanish-dominant, $n = 7$) in Alicante, Spain. Participants completed a language survey, a reading task, and a sociolinguistic interview. All linguistic tasks were recorded and analyzed acoustically on Praat (Boersma & Weenink 2023) utilizing the waveform and spectrogram to identify the strategies utilized to resolve the phonological boundaries. Results from Spanish-dominant bilinguals differ from previous studies in Puerto Rico (Mohamed, 2020), suggesting that this language contact situation differs in Spain and opening up new avenues for study in this area.

3aSC13. Efficient measurement of spoken English skills: Design and validation of a 6-minute assessment. Jared C. Bernstein (Linguist, Stanford Univ., 1600 Amphitheatre Pkwy, Mountain View, CA 94043, jaredbernstein@google.com), Jian Cheng (Analytic Measures Inc., Palo Alto, CA), and Masanori Suzuki (Analytic Measures Inc., Palo Alto, CA)

Efficient measurement of spoken language skills can improve educational and commercial services. Traditional spoken language assessments often require 30–90 minutes. We report the development and validation of a prototype mobile-browser test of spoken English, "APro", that participants complete in 6–8 minutes. The APro test presents a stratified random draw of 20 interactive tasks (from pool of 377) to participants whose spoken responses are automatically scored, scaled, and reported. APro aggregates skill-specific measures from disparate task types such as open descriptions and story retellings. Each skill-specific score was independently validated, then combined into an overall spoken English score. APro scoring was trained on 30,000 responses from 354 non-native pilot-test participants who connected remotely from their nine home countries on five continents. The final APro instrument was validated on an independent sample of 79 participants resident in seven countries. In the validation set, human and machine scoring of same-test APro administrations correlated $r = 0.98$, suggesting accurate speech processing. APro scores predicted much of the score variance in traditional concurrent assessments from Oxford, $r = 0.86$, and Pearson, $r = 0.92$. Finally, we present detailed results on two new interactive tasks and share changes to the content, procedure, and scoring that yield improved accuracy at shorter duration. [Work supported in part by Google LLC.]

3aSC14. Machine recognition of non-native speech: Task-specific language models versus large language models. Jian Cheng (Res., Google LLC, Palo Alto, CA), Jared C. Bernstein (Res., Google LLC, 1600 Amphitheatre Pkwy, Mountain View, CA 94043, jaredbernstein@google.com), and Masanori Suzuki (Res., Google LLC, Palo Alto, CA)

Automatic speech recognition (ASR) has offered a reliable foundation for measurement of young children's reading skills and of second-language (L2) speaking skills. This is because well-fit task-specific language models (LMs) enable recognition that supports accurate scoring of pronunciation, fluency, vocabulary, usage, and grammar (Bernstein and Cheng, 2023). ASR works well in these measurement tasks because measurement of word production, disfluencies, and pronunciation errors is not very sensitive to moderate differences in word-error-rate (WER) accuracy, and because speech-interactive tasks appropriate for reading instruction or L2 assessment elicit relatively predictable responses, for which task-specific low-perplexity ASR systems achieve sufficiently accurate speech recognition (Cheng and Townshend, 2003). In the work reported here, we compared the accuracy of two English ASR systems on a set of 718 extended spontaneous speech recordings from 77 adult non-native speakers of English speaking from six countries under uncontrolled recording conditions. A Kaldi-based ASR system with well-fit task-specific LMs achieved WER 17%, while USM, a general-purpose mSLAM recognizer with an RNN-T decoder, achieved 11% WER, which is a 34% relative improvement. The mSLAM + RNN-T technology will be briefly described and an analysis of results in three different open-response interactive speaking tasks will be presented.

3aSC15. VSpace: A browser-based vowel synthesiser. Michael I. Proctor (Linguist., Macquarie Univ., 16 University Ave. North Ryde, New South Wales 2109, Australia, michael.proctor@mq.edu.au)

Two-formant representation of the universal vowel space is a foundational model in phonetics (Fant & Risberg 1963; Traunmüller & Lacerda 1983), but tools for rapid synthesis of vowel sounds corresponding to points in the vowel space are limited. Stand-alone applications such as the Formant Synthesizer Demo (Beskow, 2000) provide this functionality, and rich tools for formant synthesis are available in Praat (Boersma and Weenink, 2023), Matlab (Rabiner *et al.*, 2023) and other platforms, but these tools are not universally accessible, while web-based speech synthesis tools (e.g., Thapen, 2017) do not typically allow systematic manipulation or quantification of key acoustic parameters. VSpace is a browser-based formant synthesizer developed using the Tone.js Web Audio framework. A universal vowel space is represented as a trapezoid scaled to typical male, female or child speaker formant ranges. Clicking on any point generates a vowel sound corresponding to the first and second formant frequencies selected, synthesized with typical F3 and F4 frequencies excited by a selectable source signal. IPA symbols corresponding to typical formant frequencies for language-specific vowel phonemes may be superimposed on the vowel space. Key synthesis parameters are configurable through the web interface to allow exploration of the acoustic consequences for vowel perception.

3aSC16. Gestural timing in naïve listener imitation of Cantonese vowels. Jonathan Havenhill (Dept. of Linguist., The Univ. of Hong Kong, 930 Run Run Shaw Tower, Centennial Campus, Pokfulam, Hong Kong, jhavenhill@hku.hk) and Ming Liu (Dept. of Linguist., The Univ. of Hong Kong, Hong Kong, Hong Kong)

Studies of nonnative vowel production have analyzed sounds predominantly in terms of acoustics, while the underlying articulatory gestures, as well as their timing, remain understudied. This study examines nonnative production of backness and rounding contrasts in the Cantonese vowel sets /i y u/ and /ɛ œ ɔ/. 15 native Cantonese speakers and 15 native English speakers participated. Native English speakers read a list of English words containing the vowels [i u ɪ ʊ e o ɛ ɔ], and then imitated Cantonese words containing [i y u ɪ ʊ ɛœ ɵ ɔ], presented auditorily. L1 Cantonese speakers produced the same set of Cantonese words, presented orthographically. Dynamic acoustic, ultrasound, and lip video data were collected and analyzed using GAMMs. In L1 Cantonese, target tongue and lip positions are achieved at vowel onset and maintained throughout the vowel duration, yielding acoustically monophthongal vowels. In contrast, L1 English [u, o] exhibit peak rounding during the vowel offglide, while magnitude of tongue and F2 displacement vary according to consonantal context. Lip gestures show similar late timing in non-native imitations of Cantonese [y, u], yielding English-like F2 trajectories, although the two vowels differed in tongue position. These results suggest that acoustically non-native-like vowel productions may result in part from transfer of L1 gestural timing. [Work supported by Hong Kong RGC, No. 27614421.]

3aSC17. Modeling reduced speech using a simulated vocal tract and overlapping acoustic events. Tyler T. Schnoor (Linguist., Univ. of AB, 150 Assiniboia Hall, University of AB, Edmonton, AB T6G2E7, Canada, tschnoor@ualberta.ca), Brad H. Story (Speech, Lang., and Hearing Sci., Univ. of Arizona, Tucson, AZ), and Benjamin V. Tucker (Commun. Sci. and Disord., Northern Arizona Univ., Flagstaff, AZ)

Reduced utterances are characterized by the alteration or deletion of segments (i.e., syllables, words) and often occur in spontaneous or casual speech styles. These reductions are common in everyday communication and are generally understood given sufficient context. However, modeling reduced speech is not trivial, as the effects of context, articulatory limitations, idiosyncrasies, and speech rate need to be considered in order to approximate acoustic observations of reduction. We utilize a speech production model which modulates a simulated vocal tract according to acoustic events organized along a time axis. The present study focuses on validating its capacity for modeling reduced speech. Evidence of its effectiveness is provided by way of synthetic example utterances, each accompanied by a description of the steps taken to synthesize the utterance and acoustic comparison to an unreduced equivalent. Word-medial consonant reductions will be studied first before proceeding to massive reductions where entire syllables and words are affected. We iteratively synthesize reduced utterances until they are evaluated through informal listening to be comparable to real acoustic observations. The results of this study are discussed as validation of the idea that acoustic events can be arranged to model reductions.

3aSC18. Abstract withdrawn.

3aSC19. Abstract withdrawn.

3aSC20. Neural network-based measure of consonant Lenition in Parkinson's disease. Ratree Wayland (Linguist., Univ. of Florida, 2801 SW 81st St., Gainesville, FL 32608, ratree@ufl.edu), Kevin Tang (Dept. of English Lang. and Linguist., Heinrich-Heine-Universität Düsseldorf, Düsseldorf, Germany, Germany), Fenqi Wang (Linguist, Univ. of Florida, Gainesville, FL), Sophia Vellozzi (Comput. & Information Sci. & Eng., Univ. of Florida, Gainesville, FL), Rachel Meyer (Linguist., Univ. of Florida, Gainesville, FL), and Rahul Sengupta (Comput. & Information Sci. & Eng., Univ. of Florida, Gainesville, FL)

This study investigates the effects of Parkinson's disease (PD) and various linguistic factors on the degree of lenition in Spanish stops. The lenition is estimated from posterior probabilities calculated by recurrent neural networks trained to recognize sonorant and continuant phonological features. First, individuals with PD exhibit a higher degree of lenition in their stops compared to healthy controls, suggesting that PD has a significant impact on the articulatory control of stops, resulting in more pronounced lenition. Second, stress influences the degree of lenition, with stressed syllables showing less weakening compared to unstressed syllables. Thirdly, place of articulation plays a role in lenition, but the effects differ between PD and healthy control groups. PD patients demonstrate greater weakening in bilabial and velar stops compared to dental stops, while healthy controls exhibit more lenition in velar stops, with no significant difference between bilabial and dental stops. These findings suggest abnormalities in lip movement control among PD patients align with previous research on reduced lip velocities in Parkinsonian patients. Importantly, the study highlights the effectiveness of recurrent neural networks in quantifying lenition in PD patients.

3aSC21. Neural network-based measure of Consonant Lenition in L2 speech. Ratree Wayland (Linguist., Univ. of Florida, 2801 SW 81 St., Gainesville, FL 32608, ratree@ufl.edu), Kevin Tang (Dept. of English Lang. and Linguist., Heinrich-Heine-Universität Düsseldorf, Düsseldorf, Germany), Fenqi Wang (Linguist, Univ. of Florida, Gainesville, FL), Sophia Vellozzi (Comput. & Information Sci. & Eng., Univ. of Florida, Gainesville, FL), Rachel Meyer (Linguist., Univ. of Florida, Gainesville, FL), and Rahul Sengupta (Comput. & Information Sci. & Eng., Univ. of Florida, Gainesville, FL)

This study investigates the gradient phonetic variations in the lenition of Spanish voiced and voiceless stops among second language (L2) learners with different levels of proficiency (beginning, intermediate, and advanced). The degree of lenition is measured using posterior probabilities of the continuant and sonorant phonological features, estimated by the deep learning model Phonet. The findings reveal that the degree of lenition, as indicated by the sonorant posterior probability, increases with proficiency. However, no significant effects of proficiency were observed for the continuant posterior probability. Similar to native speakers of Spanish, L2 learners exhibit effects of stress, voicing, and place of articulation on lenition. These results suggest that all learners exhibit lenition of stops as a fricative, but more advanced learners also exhibit lenition as a sonorant. Additionally, lenition in L2 is found to be gradient and influenced by linguistic factors. Moreover, the posterior probabilities of the continuant and sonorant phonological features, estimated by the Phonet model, serve as reliable measures of lenition. Overall, this study reveals the role of proficiency and linguistic factors in shaping the degree of lenition and highlights the effectiveness of the posterior probabilities obtained from the Phonet model in quantifying lenition.

3aSC22. Investigating the interaction between voice quality and plosive production in Australian Englishes: Acoustic features of vowel-/t/ sequences. Debbie Loakes (School of Lang. and Linguist., The Univ. of Melbourne, Babel Bldg., The University of Melbourne, Parkville, Victoria 3010, Australia, dloakes@unimelb.edu.au), Kirsty McDougall (Theor. and Appl. Linguist., Faculty of Modern & Medieval Lang. & Linguist., Univ. of Cambridge, Cambridge, United Kingdom), and Adele Gregory (Dept. of Lang. and Cultures, La Trobe Univ., Melbourne, Victoria, Australia)

A number of researchers have posited a link between voice quality and consonant production (e.g., Keating and Esposito 2007). This work investigates the connection between the acoustics of voice quality (laryngeal behaviour) and consonant realization (supralaryngeal behaviour) in Australian Englishes. Focusing on speech produced by 52 Australian English speakers (both "mainstream" and Aboriginal English speakers), we show that in vowel-/t/ sequences "breathy" t-categories (e.g., affricates, fricatives) co-occur with breathier vowels (including those with pre-aspiration), while laryngealized t-categories (e.g., ejectives, glottal stops) co-occur with creakier vowels. Vowels preceeding breathy /t/ categories have stastistically significantly higher F0 and H1–H2 values compared to vowels preceding canonical /t/. Meanwhile, vowels preceding glottal t-categories generally have lower mean spectral tilt measures (H1–H2, H1–A1), signifying increased creakiness in the vowel prior to glottal variants. We also report on dynamic analyses of H1–A1, paying attention to acoustic features of the last 25% of the vowel where peaks/troughs occur depending on consonant

quality. A relationship between the voice quality of the vowel and the (broad) consonant type is demonstrated, however, it is not one-to-one. Rather voice quality appears to assist in creating the right conditions for particular consonantal variants. Keating, P. and C. Esposito. (2007), “Linguistic voice quality”, UCLA Working Papers in Phonetics, 105: 85-91.

3aSC23. Weakening patterns of intervocalic voiced plosives in Japanese. Weiyu Li (Sophia Univ., 7-1 Kioi-cho, Chiyoda-ku, Tokyo 1028554, Japan, w-li-5x4@eagle.sophia.ac.jp), Ai Mizoguchi (Maebashi Inst. of Technology/NINJAL, Maebashi, Japan), Maho Morimoto (Sophia University/Japan Society for the Promotion of Sci., Tokyo, Japan), and Takayuki Arai (Sophia Univ., Tokyo, Japan)

Japanese voiced plosives have been observed to exhibit weakened burst intensity, and in some instances, they may show a complete weakening with no observable burst. Maekawa (2010) investigated the weakening rate of voiced plosives in Japanese and its relationship with surrounding acoustic environments. His study revealed a notable tendency for consonant weakening, particularly at positions characterized by weak prosodic boundaries. However, despite referring to a general phenomenon of consonant weakening, there exist various pronunciation patterns. The articulatory gestures responsible for generating these diverse weakening patterns have not yet been thoroughly investigated. In this research, we conducted recording experiments involving native Japanese speakers, simultaneously capturing tongue movements using an ultrasound device. Our focus was directed towards the patterns of weakening observed in intervocalic Japanese voiced plosives and how to classify them. We identified three major patterns of plosives consonant weakening: (1) Voiced glide pattern, (2) Voiced frication pattern, and (3) Pattern with halted vocal fold vibration and weak aspiration. Variation was found in the frequency of occurrence of these patterns among alveolar, velar, and bilabial plosives. It is important to note that the rate of specific weakening patterns at different articulatory points varies among individuals. Additionally, for alveolar and velar sounds, tongue movement was analyzed using ultrasound imaging.

3aSC24. Prelateral vowel change in Australian English *pool-pull* trajectories. Tuende Szalay (Speech Pathol., The Univ. of Sydney, Susan Wakil Health Bldg., D18 Western Ave., Camperdown, New South Wales 2006, Australia, tuende.szalay@sydney.edu.au), Titia Benders (Lit. and Linguist., The Univ. of Amsterdam, Amsterdam, the Netherlands), Felicity Cox (Linguist., Macquarie Univ., Macquarie University, New South Wales, Australia), and Michael I. Proctor (Linguist., Macquarie Univ., North Ryde, New South Wales, Australia)

Australian English (AusE) pre-/l/ /ʉː/ is retracted when coarticulated with coda /l/, leading to vowel change through acoustic contrast reduction between /ʉː-ʊ/ (*pool-pull*). Younger speakers show smaller contrast in /ʉːl-ʊl/ targets than older speakers. As vowel trajectories are less well understood, we tested changes in the F2 trajectory. 200 tokens of /ʉː, ʊ/ in the /hVd, pVl/ contexts (*who'd-hood, pool-pull*), produced by eight younger (ages = 20–29) and nine older (ages = 54–80) female AusE speakers, were extracted from the audio corpus AusTalk. Formant trajectories in pre-/d/ vowels and /Vl/ rimes were extracted automatically, corrected manually, and time-normalized (0–1). F2 trajectory was fitted with a Generalized Additive Mixed Model using fixed factors Vowel, Coda, and Age (treatment-coded, comparing /ʉː/ to /ʊ/, /l/ to /d/, younger to older) with Speaker as random intercept with smooths for Vowel and Coda. Both age groups maintained significant vowel contrast in the pre-/d/ context in the entire vowel trajectory. In the /l/-context, older speakers showed a significant F2 difference throughout the rime trajectory, while younger speakers did not. Durational differences may be maintained, as duration contrast was not analysed due to time normalization. Our findings are consistent with an AusE *pool-pull* merger.

3aSC25. Two efforts towards natural speech synthesis: Incorporating disfluency and speaking style change based on the interlocutor. Akiko Mokhtari (Toyama Prefectural Univ., Kurokawa 5180, Imizu, Toyama 939-0398, Japan, a-mokh@pu-toyama.ac.jp), Nick Campbell (The Univ. of Dublin, Nara, Japan), and Toshiyuki Sadanobu (Kyoto Univ., Kyoto, Japan)

During the period 2000–2005, a Japanese female speaker recorded her everyday conversations with many different interlocutors using a head-set microphone. As a result, 600 hours of natural Japanese speech data were obtained. This study describes a DNN-based speech synthesis system which was trained on 300 hours of the data, focusing on two unique efforts to make it more expressive in a human-like way: (1) allowing for disfluencies, and (2) accounting for the category of interlocutor. Incorporating some frequently observed disfluent patterns in general Japanese speech such as fillers, phrase-final rising intonation, and word-internal prolongation or suspension, is believed to be effective in practical application as certain disfluencies are connected to a speaker’s attitude in Japanese communication. For example, having word-internal prolongations can show hesitation or politeness, and word-internal suspending can show the speaker’s surprised attitude. Interlocutors in the original data were categorized into four groups: family, friend, child and others. This information was used in the training process, and as a result, the synthesizer can generate different speaking styles according to the interlocutor setting. Being able to generate disfluent speech and change the speaking style depending on who you are talking to can make the synthesizer ever more expressive.

3aSC26. Classification of Chubu-region dialects using random forest. Kota Hattori (Faculty of Integrated Arts and Sci., Tokushima Univ., 1-1 Minamijosanjimacho, Tokushima 770-0814, Japan, kota@tokushima-u.ac.jp) and Shinsuke Kishie (Dept. of Japanese Lit., Nara Univ., Nara-shi, Nara, Japan)

The present study investigated how Japanese dialects in Chubu region are classified using random forest (RF) and an adaptive synthetic (ADASYN) sampling approach. We obtained the written-format pronunciations of 22 words from 884 Japanese speakers (average age = 71.1) who had been residing in their birthplaces. We calculated phonetic distance (ALINE distance) between the dialectal and standard Japanese pronunciations, and ran RF models with 1,000 bootstrap samples. The results of the RF models (ROC = 0.953, F1 = 0.68) demonstrated that speakers with a predicted probability greater than 50 percent (n = 415) were generally located in their residing prefectures, suggesting that each prefecture has its own dialect. Speakers with a predicted probability less than 50 percent were located in both their residing prefectures and the vicinities of the prefectures, particularly those in Aichi, Gifu, Shizuoka, Nagano, Gunma, and Niigata prefectures, suggesting that the speakers in the areas share some characteristics of the prefectural dialects. This potentially led to the poor predictions of the speakers. The expansion of these prefectural dialects seems to be limited by the Japanese Alps; dialects spoken on the eastern and western side of the mountain range are generally widespread only in the area, respectively.

3aSC27. Inner speech processes for overt language production—Comparison of native English speakers and different proficiency learners of English. Keiko Asano (Faculty of Medicine, Juntendo Univ., 1-11-2905 Kinko-cho, Kanagawa-ward, Yokohama-city, Kanagawa-pref 221-0056, Japan, keasano@juntendo.ac.jp)

Inner speech refers to an internal process as a tool for internalized thinking besides actual overt speech. It plays an important role in analyzing and planning tasks occurring outside the world. Asano (2015) found that second language learners of English used different inner speech depending on their proficiency level. The roles of inner speech by native speakers and language learners also differ (Fernyhough *et al*., 2004). In addition, all people do not commonly use representations in the internal process. The dominance, such as speech language, written language, and visual imagery (Nakayama *et al*., 2021), to prepare for overt activities depends on the person. This research aims to determine what kind of inner speech or other processes will be prepared for the overt language activities between native speakers and second-language learners of English. The subjects for both groups answered which dominant is used as the inner preparation processes for overt language production by answering the survey questionnaires. Knowing the native speakers' inner speech process before the overt production activities will help the second language learners' overt language activities and suggest ideas for their language proficiency improvement.

3aSC28. Does language matter in oral diadochokinesis (DDK) performances? Yunjung Kim (School of Commun. Sci. and Disord., Florida State Univ., 201 W. Bloxham St., Warren Bldg., Tallahassee, FL 32301, ykim19@fsu.edu), Jeffrey J. Berry (Dept. of Speech Pathol. and Audiol., Marquette Univ., Milwaukee, WI), and Seung Jin Lee (Div. of Speech Pathol. and Audiol., Hallym Univ., Chuncheon, Korea (the Republic of))

Oral diadochokinesis (DDK) typically refers to rapid repetitions of monosyllables (e.g., /pʌ/, /tʌ/, /kʌ/) or multisyllables (e.g., /pʌtʌkʌ/). DDK has been frequently included in studies of motor development and disorders and its use has increased over the past decade. One unanswered question in using DDK for investigation of speech motor control is whether the studied language serves as an important factor in DDK performances. Despite the general agreement that DDK performance (mostly DDK rates) is little affected by a speaker's native language, few data exist on language- or dialect-specific effects on DDK, which may be explained by factors such as speaking rate and rhythmic pattern of the languages. In this study, we compare DDK performances in two languages, American English (AE) and Korean, which were chosen primarily because of the difference in their rhythmic patterns (stressed-timed versus syllable-timed). A total of 80 young adults (40 AE-, 40 Korean-speakers) performed both mono- and multi-syllable repetitions and analysis was limited to the first 5 seconds of each performance. We will present data on several aspects of DDK performance focusing on prosodic patterns (syllable and intersyllable pause durations, intensity, and F0 across syllables) in addition to average DDK rates.

3aSC29. Second formant transitions for acoustic analysis to differentiate among dementia types. Richard J. Morris (School of Commun. Sci. and Disord., Florida State Univ., 201 West Bloxham, Tallahassee, FL 32306-1200, rmorris@fsu.edu), Chorong Oh (School of Rehabilitation and Commun. Sci., Ohio Univ., Athens, OH), and Parker Franklin (School of Commun. Sci. and Disord., Florida State Univ., Tallahassee, FL)

Recent evidence indicates that acoustic features associated with emotional prosody in speech may be an inexpensive, non-invasive method for differentiating among dementia types. In particular, frequency measures in the speech of people with dementia have been associated with listeners' perception of emotional prosody. The purpose of this study was to determine if second formant (F2) transition information would enhance these differentiations. Prerecorded speech samples of *Cookie Theft* picture descriptions from 10 individuals with dementia of the Alzheimer's type (DAT), 5 with vascular dementia (VaD), 9 with MCI, and 10 neurologically healthy controls (NHC) were obtained from the DementiaBank. Nine words that had initial obstruent consonants that occurred at least two times in each of the participant groups were selected for measurement. The F2 durations, extents, and slope were measured and analyzed. Across group comparisons revealed no pattern for F2 durations or extents. The fricatives had longer F2 durations than the plosives. Across word comparisons revealed significant differences across the consonant vowel combinations for all three measures. A group by word interaction occurred for the F2 slope with the VaD group exhibiting larger slopes than the other two groups.

3aSC30. Can a neurotypical actor align their speech acoustically to autistic speech patterns for roles in the media? Haley A. Todd (Linguist, George Mason Univ., 4400 University Dr., Fairfax, VA 22030, htodd@gmu.edu)

Canonically autistic characters are represented in television, but these roles have often been filled by neurotypical actors. One criterion of autism diagnoses includes differences in speech, often described as "monotonic" or "robotic." This study comparatively analyzed the speech of a neurotypical actor in an autistic role with that of autistic actors within neutral and emotional conditions. Acoustic measures included fundamental frequency (F0), duration, and intensity. Speech samples ($n = 87$) were collected from the television show *Atypical* and acoustically analyzed using PRAAT. Results showed that the F0 range, syllable duration range, and intensity range for the neurotypical actor in the autistic role was significantly more similar to neurotypical actors than autistic actors, though the change in syllable duration between conditions was significantly more similar to autistic actors. Additionally, a survey was developed in which participants listened to audio clips of the speech samples rating them on a Likert scale of "typical" to "atypical." Results showed that the neurotypical actor in the autistic role was rated as atypical. Overall, it was concluded that neurotypical actors may not acoustically align with autistic actors, but they are successful in the perceived effect of atypicality based on the perception of their speech patterns by an audience.

3aSC31. "Do I sound happy?" Acoustic characteristics of affective prosody in older adults. Chorong Oh (School of Rehabilitation and Commun. Sci., Ohio Univ., Grover Ctr. W218, Athens, OH 45701, ohc@ohio.edu), Richard J. Morris (School of Commun. Sci. and Disord., Florida State Univ., Tallahassee, FL), and Xianhui Wang (School of Medicine, Univ. of California Irvine, Irvine, CA)

It is known that older adults have difficulty comprehending affective prosody. However, it is unclear how well they manipulate affective prosody to express emotion. Five older adults were recorded when they completed three speech tasks: (1) Talking about happy and sad life events; (2) Describing the Refused Umbrella pictures; (3) Explaining how to make a peanut butter and jelly sandwich. Two independent investigators listened to the recordings and parsed them into utterances considering pauses and connectivity. Afterwards, the investigators identified the primary emotion (i.e., happy, sad, neutral) in each utterance. The recordings were acoustically analyzed using Pratt for rate, pitch, and amplitude measures. The rate measures included speech and articulation rates and the pitch/amplitude measures contained initial, final, minimum, maximum f0/dB and the difference between the maximum and minimum f0/dB. Multivariate ANOVAs with Tukey post-hoc test were conducted to analyze the acoustic data. Among the acoustic measures, all rate and pitch measures were significantly different across the three emotions. However, none of the amplitude measures differed significantly. Research on production of affective prosody in healthy older adults has been scarce and this study may serve as a catalyst for further exploration in this area.

3aSC32. Advancing hearing research with the NAL ecologically momentary assessment platform for real-world insights. Nicky Chong-White (National Acoust. Labs., 16 University Ave., Macquarie University, New South Wales 2109, Australia, nicky.chong-white@nal.gov.au), Joseph Tagudin, and Jorge Mejia (National Acoust. Labs., Sydney, New South Wales, Australia)

Ecological Momentary Assessment (EMA) is an invaluable tool for assessing people's behaviours and experiences in their natural surroundings. We have developed a smartphone-based EMA tool, called NEMA, that has contributed to over 10 research studies by providing meaningful real-world insights into how individuals with hearing loss interact with various hearing device technologies. By providing real-world perspectives that complement

traditional lab-based tests, NEMA offers a more comprehensive understanding of everyday hearing and communication challenges. Its advanced features include cloud-connectivity, customisable surveys, cross-device acoustic feature measurements from the environment, and a back-end platform enabling real-time data monitoring and visualization. To derive vital insights from this rich data stream, advanced analytical methodologies are applied. However, there exist limitations with smartphone-based EMAs that can influence their overall effectiveness. We identify these constraints and discuss strategies to alleviate participant burden and increase task relevance. These initiatives aim to enhance compliance and diversify data collected. Current work involves integrating smartwatches and additional sensor technology into our next generation NEMA platform. This progression aims to facilitate a more comprehensive capture of diverse data sets, ultimately leading to a deeper understanding of individual auditory experiences.

3aSC33. A cross-dialect study of vowel production in Parkinson's disease. Austin Thompson (Commun. Sci. and Disord., Univ. of Houston, Melcher Life Sci. 3871 Holman St. Rm. M242, Houston, TX 77204-6018, athompson23@uh.edu) and Yunjung Kim (School of Commun. Sci. and Disord., Florida State Univ., Tallahassee, FL)

The impact of dysarthria on vowel formant frequencies is well established. Additionally, it is understood that speaker dialects significantly influence formant frequencies. However, the specific interaction between dysarthria and speaker dialect remains a relatively unexplored area in research. This knowledge gap may hinder the generalizability of dysarthria findings from one regional dialect to speakers of other dialects. As an initial step in this research line, the current study investigates the interaction between regional dialects and dysarthria in Parkinson's disease (PD) on vowel production. To do this, our study analyzes acoustic and kinematic data from speakers with and without dysarthria, representing Midwestern and Southern United States dialects. We examine both single-word repetitions and passage-reading contexts, recognizing that there may be a significant task effect. The study includes 50 speakers, with 24 speakers (10 PD, 14 control) from the Southern Dialect (collected in Louisiana and Florida) and 26 speakers (13 PD, 13 control) from the Upper Midwest dialect (collected in Wisconsin). The findings will shed light on the interaction between regional dialects and dysarthria, offering valuable insights for speech rehabilitation and understanding speech variations across different populations.

3aSC34. Location and size of constriction in labiovelar, velar, and uvular sounds in French. Md Jahurul Islam (Linguist., The Univ. of BC, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, jahurul.islam741@gmail.com) and Bryan Gick (Linguist, The Univ. of BC, Vancouver, BC, Canada)

This study investigates the location and size of constriction in the labiovelar sound [w]. While there have been studies exploring constrictions in different sounds including schwa [Gick, 2002, *Phonetica* 59], /r/ [Epsy-Wilson *et al*., 2000, *JASA* 108], and clicks [Miller *et al*., 2009, *JIPA* 39], the understanding of the constriction in labio-velar sound [w] in relation to neighboring sounds is limited. We utilized an MRI corpus of French speech [Isaieva *et al*., *Scientific Data* 8] to measure the location and size of the constrictions in [w], [k], [u], and [ʁ]. The MRI video frames were manually traced to mark the upper surface of the tongue and the lower surface of the hard-palate, velum, and uvula, resulting in two contours. Using Euclidean distance, the location and size of the constriction were identified as the narrowest point and distance, respectively, between the two contours. Results confirmed that [w,u] had constrictions against the velum; [k]'s, however, had constrictions fronter than [w,u] (centroids were 0.88au apart on normalized X-Y space), making them more palatal. [ʁ]'s had uvular constrictions, as expected. Regarding constriction size, [w], despite being an approximant consonant, had a more open constriction than the vowel [u].

3aSC35. mGlif: A software tool for manual glottal inverse filtering. Parham Mokhtari (Toyama Prefectural Univ., Kurokawa 5180, Imizu, Toyama 939-0398, Japan, parham@pu-toyama.ac.jp)

Despite decades of advances in automated estimation of the voice source (glottal flow) from the speech signal, manual glottal inverse filtering remains the gold standard; yet, options are limited regarding available software tools. Here, a new software application for glottal-flow estimation is introduced, with a graphical user interface designed for manual specification and tuning of inverse-filter parameters while results are updated and displayed in real-time. The user can flexibly control the signal polarity and sampling rate, the analysis frame position and duration, and the type of window and preemphasis for vocal-tract modeling. The tunable inverse-filter parameters are the frequencies and bandwidths of vocal-tract poles and zeros, and the lip-radiation coefficient. A single window displays graphical information in several panels: the speech waveform (scrollable), the selected frame, its log-power spectrum before and after preemphasis, the model spectrum including vocal-tract and lip-radiation effects, the estimated glottal-flow waveform, its derivative, and their log-power spectra. All information pertaining to an analysis may be saved at any time, ready to be reloaded for review or analysis resumption. Key data may also be exported for further use outside the application. The software will be made freely available to the research community.

3aSC36. Consonant and vowel rounding: Same acoustics, different visuals. Baichen Du (Dept. of Linguist., Univ. of California, Berkeley, 1203 Dwinelle Hall #2650, Berkeley, CA 94720, albertbaichendu@berkeley.edu), Alexandra Pfiffner, and Keith Johnson (Dept. of Linguist., Univ. of California, Berkeley, Berkeley, CA)

The phonological feature [±round] is normally associated with vowels, but this feature is also relevant for some consonant contrasts, such as the retroflex/non-retroflex sibilant contrast in Mandarin. One might assume that contrastive lip rounding would be the same gesture for consonants and vowels within language. However, because similar acoustic results can be achieved through different articulations, we hypothesize that the exact parameters of rounding might be variable for different segment types. An audiovisual production experiment was conducted with 30 Mandarin native speakers. Subjects produced 35 words with sibilants onsets (retroflexes /ʂ, ʈʂ, ʈ^{h}/, alveolars /s, ts, ts^{h}/, palatoalveolars /ɕ, tɕ, tɕ^{h}/), and 40 words with high-front rounded/unrounded nuclei (/i/ and /y/). We found robust individual variation and that consonants and vowels were characterized with different visual properties: /y/ had small aperture in both vertical and horizontal directions and therefore smaller lip opening area than in /i/, while retroflexes had smaller horizontal but larger vertical apertures than alveolars or palatals. Acoustical consequences of rounding were similar: both spectral Center of Gravity and F1–F4 were lowered compared to unrounded counterparts, indicating that the same phonological feature may have multiple visual/articulatory correlates.

Session 3aSP

Signal Processing in Acoustics: Signal Processing Potpourri I

Erin Driscoll, Cochair
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Chair's Introduction—9:15

Contributed Papers

9:20

3aSP1. Angle domain cepstral comb and notch liftering—A method to separate concurrent gear and bearing faults under variable speed conditions. Robert Randall (Mech. Eng., UNSW Sydney, Barker St., Kensington, New South Wales 2033, Australia, b.randall@unsw.edu.au) and Wade Smith (Mech. Eng., UNSW Sydney, Kensington, New South Wales, Australia)

In recent years, in particular because of the importance of wind turbines in renewable power generation, there is now a demand for signal processing techniques to cope with the variable speed. It is also important to distinguish between gear and bearing faults which are both critical in wind turbine monitoring. This paper describes a pair of interrelated techniques, which make use of the fundamental differences between gear and bearing fault signals to separately diagnose them, even when both are present at the same time, and for speed variations up to ± 20%. The signals are order tracked to the (rotation) angle domain, after which the real cepstrum is calculated. The deterministic gear signals appear only at discrete quefrencies, which can be extracted by comb liftering, which results in signals virtually independent of speed and speed variation, and which are very close to the static transmission error (STE) of each gear. For bearing faults, the discrete quefrency components are instead removed by a comb notch lifter, leaving a signal dominated by the bearing fault information. The paper gives numerous examples from gear test rigs with concurrent gear faults (root crack and developing pitting) and bearing faults, separating and diagnosing the two types of fault sources while excluding background noise.

9:40

3aSP2. Predicting virtual acoustic source properties in a stereophonic sound-field using the coherence properties of the acoustic velocity vector field. Erin Driscoll (Elec. and Comput. Eng., Univ. of Rochester, 120 Trustee Rd., Comput. Studies Bldg., Rochester, NY 14620, edrisco5@ur.rochester.edu), Mark Bocko, and Sarah R. Smith (Elec. and Comput. Eng., Univ. of Rochester, Rochester, NY)

It is widely understood that the mutual coherence properties, including the relative phase, of the channel signals in a stereophonic sound reproduction system significantly influence a listener's perception of the resulting virtual acoustic sources. This talk presents a quantitative model that combines wavefield coherence theory, the acoustic velocity vector, and known perceptual cues to make quantitative predictions of the perceived location and spatial spread of horizontally placed stereophonic virtual sources as a function of the mutual coherence properties of the stereo channel signals. It will be seen that a small number of parameters, including the intrinsic coherence time of the source signal, the relative phase of the channels, and perceptual averaging times, account for most of the variation in the cues that determine virtual acoustic source localization. The output of the coherence model will be compared to the results of earlier experiments, including ones investigating the "phaseyness" problem and precedence effect. Additionally, the potential of this model to account for reverberation in the listening space will be demonstrated.

10:00

3aSP3. Multizone reproduction by matching the velocity vectors. Jiarui Wang (The Australian National Univ., 115 North Rd., Brian Anderson Bldg., Acton, Australian Capital Territory 2601, Australia, u5879960@anu.edu.au), Thushara Abhayapala (The Australian National Univ., Canberra, Australian Capital Territory, Australia), Jihui (Aimee) Zhang (Inst. of Sound and Vib., Univ. of Southampton, Southampton, United Kingdom), and Prasanga N. Samarasinghe (The Australian National Univ., Canberra, Australian Capital Territory, Australia)

This paper proposes a velocity-based multizone reproduction method, which reproduces the velocity vectors throughout each listening zone. Velocity vectors are related to the localization of sound below 700 Hz. Previous work considered reproducing the velocity vectors at sweet spots or on the listening zone's boundary, and usually involved measurements using multiple velocity sensors. In this paper, in each listening zone, the velocity vectors are calculated from the spherical harmonic coefficients of the pressure, which are obtained by a spherical microphone array. The calculation of the velocity vectors is based on the sound field translation formula. Moreover, by reproducing the desired velocity vectors throughout each listening zone, listener's movement within each zone is allowed. To enlarge the size of the listening zone, the sound field in each listening zone is expressed by a sparse distribution of plane waves that results in the same velocity vectors. The pressure transfer functions between each loudspeaker and each listening zone are measured by placing a spherical microphone array at multiple spatial sampling points within each listening zone. The loudspeaker driving functions are calculated by matching the velocity vectors due to the sparse distribution of plane waves in each listening zone.

10:20–10:40 Break

10:40

3aSP4. Development of a data sonification toolkit and case study sonifying astrophysical phenomena for visually imparied individuals. Kim-Marie N. Jones (Arup, 151 Clarence St., Level 5 Barrack Pl., Sydney, New South Wales 2000, Australia, kim.jones@arup.com), Mitchell J. Allen (Arup, Sydney, New South Wales, Australia), and Kashlin McCutcheon (Sydney, New South Wales, Australia)

Data is predominantly conveyed and analysed in a visual manner. Data sonification provides an alternative approach to data analysis, has a broader application (e.g., peripheral monitoring) and offers accessibility to an alternative cohort (e.g., the visually impaired). While there are countless data visualization tools available, producing data sonifications typically requires in-depth knowledge of audio software and/or computer programming. Research was undertaken into existing data sonification tools, and subsequently a Data Sonification Toolkit was developed using Abelton Live and Max for Live. The Toolkit was developed in particular as an alternative means of data analysis for use by multiple disciplines across a built environment firm. The key aims for the toolkit were that (1) it should be user-friendly and accessible to people without an in-depth knowledge of either Ableton or Max, (2) the sonifications produced be accurate and true representations of input data sets, and (3) the Toolkit should have the capability and flexibility to be expanded and customized by those with the expertise to do so. The Toolkit was used, in collaboration with astrophysicist Chris Harrison, to develop sonifications of astronomical phenomena for the visually impaired. These sonifications were presented at the British Science Festival in 2019.

11:00

3aSP5. Consistent physical frequencies in time-frequency analysis. Manton J. Guers (Acoust., Penn State Univ., PO Box 30, State College, PA 16804, mjg244@psu.edu)

Wavelet transforms have been studied extensively for a wide variety of applications such as signal compression and signal denoising. Wavelet transforms have also been examined for the detection of transient signals. However, wavelet levels (and corresponding pseudo-frequencies) are inherently dependent on the sampling rate of the analyzed data. The work presented herein examines how wavelet analysis can produce inconsistent data representations for the same analytical signal digitized at different sampling rates. Results are compared to time-octave and other time-frequency representations to identify methodologies which produce consistent characterization of physical frequencies. Similarities and difference between approaches are discussed.

11:20

3aSP6. Robust power spectral density estimation via a performance-weighted blend of order statistics. David C. Anchieta (ECE, Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., Dartmouth, MA 02747, danchieta@umassd.edu) and John Buck (ECE, Univ. of Massachusetts Dartmouth, Dartmouth, MA)

Loud transients introduce bias to background spectrum estimates based on the sample mean, like the Welch Overlapped Segment Averaging (WOSA) [1967]. The Schwock and Abadi [2021] Welch Percentile (SAWP) estimator avoids the loud transient bias by replacing the averaging of the WOSA with a scaled order statistic (OS) of the sample power spectrum. While the scaling ensures the SAWP is unbiased regardless of which OS is chosen, the 80th percentile minimizes the variance of the SAWP estimator in a scenario without loud transients. However, the 80th percentile SAWP is still vulnerable to bias and increased variance from more frequent loud transients. Also, the rate of occurrence of loud transients may change with time, requiring the SAWP to adapt which percentile is used. To approach this challenge, this talk proposes a Universal SAWP estimator which is a weighted sum across different fixed percentile SAWP estimators. At each iteration, the Universal SAWP updates the blend weights to promote the percentile with the lowest sample variance over recent observations. In computer simulations, the Universal SAWP quickly assigns higher weights to the lowest variance estimators as the occurrence of loud transients increases. Overall, the Universal SAWP achieves equal or lower variance as the best fixed percentile SAWP estimator. [Work supported by ONR Code 321US.]

Session 3aUW

Underwater Acoustics: Underwater SONAR

Tracianne B. Neilsen, Chair
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Contributed Papers

8:00

3aUW1. Modeling split-beam sonar to evaluate fish target strength accuracy. Axel Belgarde (Phys. and Physical Oceanogr., Memorial Univ. of NF, 7-39 Queens Rd., St. John's, NF A1C 2A6, Canada, abelgarde@mun.ca), Len Zedel (Phys. and Physical Oceanogr., Memorial Univ. of NF, St. John's, NF, Canada), and Mahdi Razaz (School of Ocean Sci. and Eng., Univ. of Southern MS, Hattiesburg, MS)

Split-beam echosounders enable estimation of fish size by directly measuring their target strength. This measurement is achieved using compound transducer array geometries with accurate knowledge of resulting beam patterns. The multiple transducer components and required accurate calibration can present challenges in predicting the exact sonar capabilities. We present a model of the split-beam sonar system that can be used to evaluate and optimize sonar performance of a given transducer before committing to a hardware design. The model has been used to generate beam patterns to match prototype instruments and to simulate acoustic signals based on the scattering of sound from particles in a three-dimensional domain. Different split-beam sonar algorithms have been compared to measure the position and target strength of particles in the generated signal. The model's prediction capabilities were evaluated through comparisons with field trials of a prototype system. The field trials were conducted by lowering a calibration target sphere to a range of 200 m in the acoustic beam. Both model prediction and prototype system performance show accuracy of $\sigma = \pm 0.2$ dB at 25 m range. Potential future applications of the model include exploring methods of target separation and improving accuracy when presented with complicated target structures.

8:20

3aUW2. Bubble pulse arrival time calculations to determine SUS charge detonation depths. Haley H. Green (Ocean Eng., Univ. of Rhode Island, 215 South Ferry Rd., Narragansett, RI 02882, haley.green@uri.edu), Andrew Ferguson, Sydney Johnston, James H. Miller, Gopu R. Potty (Ocean Eng., Univ. of Rhode Island, Narragansett, RI), and David R. Dall'Osto (Acoust., Appl. Phys. Lab. at Univ. of Washington, Seattle, WA)

As part of the New England Seamount Acoustics (NESMA) project in 2023, experiments were conducted over the Atlantis II Seamount. A total of 216 Signal Underwater Sound (SUS) broadband explosives (150 1.1 oz and 66 1.8 lb net explosive weight) were deployed at multiple locations and two different design depths around the seamount. An acoustic recorder (SoundTrap ST4300), with four hydrophone channels of varying sensitivities, was deployed during each of the SUS events to record the signals generated by the explosives. The recorded signals include arrivals corresponding to the shock wave, subsequent bubble pulses and reflections from the sea surface and sea floor. The time between the first arrival and the bubble pulse was used to calculate the depth at which each SUS charge detonated. The resulting SUS detonation depths from the bubble pulse calculation were compared with depths obtained from surface and bottom reflections to assess the accuracy of the bubble pulse method. A comparison of the results from the two methods shows that using the bubble pulse interval to calculate the SUS detonation depth is reasonable and can be used in propagation models. [Work supported by Office of Naval Research Code 322OA and TFO.]

8:40

3aUW3. Narrowband underwater vector sensor using michroelectromechanical systems. Fabio Alves (Phys., Naval Postgrad. School, 833 Dyer Rd., Monterey, CA 93943, fdalves@nps.edu) and Justin Ivancic (Phys., Naval Postgrad. School, Monterey, CA)

A narrowband underwater vector sensor using microelectromechanical (MEMS) systems is demonstrated. A combination of two pressure gradient MEMS sensors and an omnidirectional hydrophone allows for determination of the direction of arrival (DOA) of income sound over 360 degrees azimuth. The MEMS sensors, inspired on the hearing system of the *Ormia-ochracea* fly, consist of two wings connected by a bridge and anchored to the substrate by a tortional bean. They are operated with open back to allow a cosine dependence with the angle of incidence in the predominant bending vibrational mode. In the vector sensor, two MEMS sensors are orthogonally arranged to provide a cosine and sine directional pattern. When combined with an omnidirectional hydrophone that shares the same phase center, an arctangent estimator is used for DOA determination. The MEMS sensors are operated near resonance to filter out noise coming from undesired bands and improve the signal-to-noise ratio. A correction algorithm is applied to compensate frequency response differences. Two configurations have been demonstrated with the MEMS sensors enclosed in silicone oil and in air. The standard deviation of the measured DOA error has been computed for both configurations to be less than 5 degrees for various types of sound stimuli.

9:00

3aUW4. Shift-variant deconvolution beamforming using Implicit Neural Representation for near-field acoustic image measurement. Yifan Zhou (Key Lab. of Underwater Acoust. Signal Processing, Southeast Univ., No. 2, Southeast University Rd., Jiangning District, Nanjing, Nanjing, Jiangsu 211189, China, yf_zhou@seu.edu.cn), Shiliang Fang, Liang An, and Lijun Chen (Key Lab. of Underwater Acoust. Signal Processing, Southeast Univ., Nanjing, Jiangsu, China)

The conventional beamforming (CBF) output can be expressed as a convolution of the source distribution and the array dependent point spread function (PSF), which is defined as the response of the beamformer to a point source. The resolution and accuracy of sound source localization can be improved by deconvolution of CBF. The shift-invariant PSF assumption is only a good approximation when the source region is small compared to the distance between the array and the source. However, the PSF of the near-field underwater acoustic map measurement is shift-variant, whose mismatch with the beam could lead to performance degradation. In this paper, we proposed an optimized deconvolution method by using the neural network called implicit neural representation (INR) to solve the beam mismatch problem, due to its strong performance in learning and reconstructing spatial scenes. Rather than recompute the PSF for each pixel in the scene, INR predicts the complex-valued weight matrix in two-dimensional space that encapsulates both the spatially varying properties of the PSF and the scatter distribution of the scene. Numerical simulations and experimental results verify the effectiveness of our method. And this work can be broadly applied to arrays with shift-variant PSFs.

9:20

3aUW5. A novel bearing-time record estimation method based on α-stable distribution modeling. Ge Yu (National Key Lab. of Underwater Acoust. Technol., Harbin Eng. Univ., Harbin, Heilongjiang, China), Bingbing Tang (National Key Lab. of Underwater Acoust. Technol., Harbin Eng. Univ., Nantong St., Nangang District, Heilongjiang, Harbin 150001, China, tangbing@hrbeu.edu.cn), and Shengchun Piao (National Key Lab. of Underwater Acoust. Technol., Harbin Eng. Univ., Harbin, China)

Bearing-time record (BTR) is widely used in the field of passive sonar information processing for target tracking. Most existing BTR estimation methods of ship radiated noise are generally based on energy-varying under the assumption of Gaussian modeling in the time domain. However, some weak trajectories may exist in BTR for the decentralized energy distribution and unstable goodness of fit with Gaussian modeling caused by the spatio-temporal instability of the marine environment. In this paper, a novel ship trajectory enhancement method which is based on α-stable distribution modeling theory is presented. In order to make the statistical characteristic of ship radiated noise better reflected, the proposed method utilizes the α-stable distribution modeling in the real part of ship radiated noise Discrete Fourier Transform (DFT) coefficient instead of the Gaussian modeling in the time domain. The scale parameter γ of α-stable distribution is substituted for variance and enhanced BTR is achieved. According to the experimental results, the real part of the DFT coefficient of ship radiated noise could be fitted by α-stable distribution at the significant level of 0.05. Compared with traditional BTR estimation method, the proposed method could gain 3.1dB enhancement on peak signal-to-noise ratio (PSNR).

9:40–10:00 Break

10:00

3aUW6. Evolution of performance metrics in the sonar world. Nicholas J. Felgate (Dstl, Dstl Portsdown West, Portsdown Hill Rd., Fareham PO17 6AD, United Kingdom, NJFelgate@dstl.gov.uk), Haym S. Panesar (Dstl, Fareham, United Kingdom), and Andrew P. Holden (Dstl, Fareham, United Kingdom)

The performance of sonar technology, including passive acoustic monitoring, environmental impact assessment and the monitoring of man-made noise sources, is typically characterized by the through-water detection range. This single performance measure depends on the characteristics of the sound sources, the underwater environment and sensing capabilities. Measuring sonar performance is considered critical in the deployment, operation and development of sonar systems, otherwise one is danger of investing in the wrong technology and/or operating this technology incorrectly. It is however already known that using detection range by itself may be insufficient, for instance where the detection and classification tasks are inseparable and counter-detection needs to be taken into account. This talk will describe the different aspects of sonar performance that need to be considered and correctly assessed. In particular, ongoing work to identify better metrics for detection, classification and localization will be outlined. This work is looking initially to borrow metrics from other domains and applications, both for speed of development and retention of cross-domain compatibility, in particular from the area of Machine Learning. Finally, this talk will identify the major areas where further work may still be required.

10:20

3aUW7. An overview of the MASTODON sonar simulator. Denton Woods (NSWC PCD, 110 Vernon Ave. Panama City, FL 32407, denton.l.woods.civ@us.navy.mil)

The Modular Acoustic Simulation Toolset of the Department of the Navy (MASTODON) is a feature-rich ray-based sonar simulator derived from the Personal Computer Shallow Water Acoustic Toolset (PC SWAT) codebase, developed at the Naval Surface Warfare Center Panama City Division over several decades. This paper discusses the design of this new simulation architecture and gives an overview of its features and models used. New simulation capabilities will be highlighted that enable drastic speed improvements from the addition of graphics processing unit (GPU) simulation code into MASTODON's codebase. [Work supported by the Office of Naval Research.]

10:40

3aUW8. Aural based scene understanding for sonar applications. Daniel C. Brown (Penn State Univ., University Park, PA) and Benjamin Cowen (Penn State Univ., University Park, State College, PA 16801, ben.cowen@psu.edu)

Acoustic returns scattered from either surfaces or objects are typically interpreted using image processing techniques such as beamforming and computer vision models such as convolutional neural networks. However, image representations can be burdensome to generate, creating delayed responses in real-time scenarios, and difficult to parse from the scene-understanding perspective. This work investigates direct exploitation of the information content from streaming, near-raw sonar time series data for the purpose of scene characterization. Summary statistics drawn from the sonar performance estimation community and communities that study aural-based perception are used to identify scene characteristics such as flat sand, ripples, gravel beds, or sea grass, in side-looking synthetic aperture sonar (SAS) data. An empirical multi-class classification study is presented that explores the performance of seafloor characterization methods on time series versus imagery.

11:00

3aUW9. A study on the optimal configuration of absorbing materials of multi-layered silencers for maximizing transmission loss. Dongheon Kang (Seoul National Univ., 1, Gwanak-ro, Gwanak-gu, Seoul, South Korea, honeyheony22@snu.ac.kr), Haesang Yang, and Woojae Seong (Seoul National Univ., Seoul, South Korea)

This study focuses on the optimal configuration of absorbing materials of multi-layered silencers applied to seawater pipelines. Absorbing materials in silencers are made of poroelastic material, and acoustic wave propagation is analyzed using the Biot-Allard Model. Acoustic transmission loss is computed numerically using COMSOL Multiphysics. It is observed that among various standard polyurethane materials, those with lower density exhibit higher acoustic transmission loss. Finally, optimization is carried out with the objective function of maximizing the transmission loss at target frequencies. Low transmission loss values at specific frequencies of one-layer silencers can be meaningfully increased by optimizing the sequence of various standard polyurethane materials.

11:20

3aUW10. Coherent combination of perpendicular synthetic aperture sonar passes. Kyle S. Dalton (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, ksd5377@psu.edu), Daniel C. Brown (Penn State Univ., University Park, PA), and Thomas E. Blanford (Ctr. for Acoust. Res. and Education, Univ. of New Hampshire, Durham, NH)

Many downward-looking synthetic aperture sonar (SAS) systems form a synthetic aperture in the direction of travel (along-track) and a real aperture in the perpendicular direction (cross-track). At lower frequencies and longer ranges, the difference between the synthetic aperture resolution and real aperture resolution may become significant. Images created with such a spatially-varying resolution may provide an inaccurate depiction of important seafloor features or objects of interest. Combining two perpendicular SAS passes into one "effective" pass provides a synthetic aperture in two orthogonal directions, which should enable the region where the two passes overlap to be imaged with a spatially-consistent resolution. The difficulty in this "multi-pass coherent fusion" (MPCF) comes in quantifying and accounting for all of the sources of error that occur when forming a synthetic aperture. This presentation will outline current work towards the coherent combination of perpendicular SAS passes. We will examine factors that impact the coherence between passes and show how this work relates to existing research in repeat-pass SAS, image co-registration, and SAS micronavigation. Imagery and results from both modeled data and field data will be presented. Finally, future work and applications to automatic target recognition will be discussed.

11:40

3aUW11. Joint channel and Doppler shift estimation for underwater acoustic single carrier system. Xinran Cao (School of Information and Electronics, Beijing Inst. of Technol., No. 5 Yard, Zhong Guan Cun South St. Haidian District, Beijing 100081, China, caoxinran1999@163.com) and Lijun Xu (School of Information and Electronics, Beijing Inst. of Technol., Beijing, China)

Underwater acoustic channel is a time-varying, strong multipath channel. Due to the low velocity of sound (about 1500 m/s), Doppler shift, occurred by the relative motion between transmitter and receiver, restricts the performance of communication. In order to transmit data in underwater acoustic single carrier communication system with strong stability and high reliability, especially in high mobility scenarios, a joint channel and Doppler shift estimation method is proposed in this paper. We propose a novel data frame structure based on a Gold sequence, so the channel and Doppler shift can be estimated. For different underwater scenarios, we design Gold sequence templates with different Doppler factors. When the Gold sequence from the received data frame is extracted, with good correlation characteristics of the Gold sequence, the maximum correlation value can be calculated, and the Doppler shift corresponding to the maximum is the estimation of Doppler shift. Then Doppler shift will be compensated by resampling the communication data. For channel estimation, the Gold sequence is also used to estimate the time-domain channel impulse response (CIR). Simulation results show that the proposed method can effectively estimate channel and Doppler shift in an environment with multipath and Doppler.

WEDNESDAY AFTERNOON, 6 DECEMBER 2023 ROOM C3.4, 1:00 P.M. TO 2:20 P.M.

Session 3pAAa

Architectural Acoustics and Signal Processing in Acoustics: Audio for Architectural Acoustics, Indoors and/or Outdoors II

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

Alexander Case, Cochair
Univ. of Massachusetts, Lowell, 25 Wilder Street, Lowell, MA 01854

Contributed Papers

1:00

3pAAa1. Electroacoustic modeling of speech intelligibility for a new road tunnel. Derek Thompson (WSP Australia Pty Ltd., Melbourne, Victoria, Australia), Reymar Victoria (WSP Australia Pty Ltd., Level 27, 680 George St., Sydney, New South Wales 2000, Australia, reymar.victoria@wsp.com), and Jennifer Feng (WSP Australia Pty Ltd., Sydney, New South Wales, Australia)

This paper presents the process of electroacoustic design and modeling for tunnel public address (PA) system for the North East Link (NEL) road tunnel in Melbourne. The tunnel includes over 6km of twin-bore tunnels, with varying cross-sections along its length. The different tunnel spaces addressed in the design include mainline tunnel bores, interchanges, ramps, as well as pedestrian cross-passages and egress passages. This paper outlines how large data sets from CAD models were managed using a new automation method (Grasshopper 3D) throughout the design process to produce and maintain detailed electroacoustic modeling in Odeon software. In particular, this paper explores new and novel modeling methods adopted to address design constraints and to achieve a PA system capable of producing adequate speech intelligibility for emergency evacuation within very challenging acoustic environments.

1:20

3pAAa2. Acoustical characterization of historical performance structure of Taramati Baradari at Hyderabad, India. Syed A. Akhtar (Dept. of Conservation, School of Planning and Architecture, Bhopal, Madhya Pradesh, India), Manish Manohare (Dept. of Architecture and Planning, Indian Inst. of Technol., Roorkee, Haridwar District, Roorkee, Uttarakhand 247667, India, mmanish@ar.iitr.ac.in), and Vishakha Kawathekar (Dept. of Conservation, School of Planning and Architecture, Bhopal, India)

The Baradari, a square-shaped building with twelve open doors, was designed for entertainment purposes and featured outstanding acoustic qualities, making it ideal for music and dance performances by noble courtesans. This article focuses on the acoustic characterization of Taramati Baradari, a historical performance structure in Hyderabad, India, built during the 17th century by the Qutb Shahi (Golconda) dynasty. The research aimed to assess the space's acoustical behaviour through *in-situ* measurements in an unoccupied state. Virtual acoustical models were developed and validated using the *in situ* measurements in the same condition, later simulated and analysed for both occupied and unoccupied scenarios, revealing an average reverberation time (RT60) of 4.1 s at mid-frequencies for unoccupied conditions. After analysing various acoustical parameters, it became evident that Taramati Baradari's acoustics possessed a distinctive character, giving the structure a unique identity. Unfortunately, in the Indian context, conservation and restoration efforts have primarily focused on visual aspects, often

neglecting considerations of the space's acoustics. Therefore, characterizing the acoustic nature of performance structures becomes vital to inform sustainable interventions for conservation architects.

1:40

3pAAa3. Influence of indoor and outdoor noises on the speech intelligibility in classrooms. Seung-Min Lee (Chungbuk National Univ., 1, Chungdae-ro, Seowon-gu, Cheongju 28644, South Korea, lsm0515@chungbuk.ac.kr) and Chan-Hoon Haan (Chungbuk National Univ., Cheongju, South Korea)

There are many noise sources inside the classrooms such as heating, ventilating, and air-conditioning(HVAC) noises. Also, there are outdoor noises near classrooms such as road traffic noises. Under these situations, teachers are using loudspeaker system in order to enhance the speech transmission. The present study aims to investigate the students' subjective responses to teacher's speech under many different noise situations. in order to this, questionnaire survey was undertaken to 60 students in two different classrooms. They were asked to mark their subjective listening comprehension and speech intelligibility in 5-points scale at every cases when three conditions were changed including opening status of doors and windows, operating HVAC system, and usage of loudspeaker system. Also, sound levels were measured in each case. As a result, it was found that only 17% of subjects responded that they were prevented from noises from outdoor and HVAC system. Also, it was responded that using loudspeaker system can enhance the listening conditions 2.7 times more than normal condition. Nevertheless, there are 33% of negative responses to loudspeaker system due to the artificial tone of speech. Thus, it can be concluded that indoor and outdoor noises cause by opening windows and operating HVAC system do not affect student's speech intelligibility much in general cases, Also, using loudspeaker system in small classroom should be used cautiously considering students' preferences.

2:00

3pAAa4. Spatial audio reproduction of floor impact-borne sounds for auditory demonstrations in living environment. Youngmin Kwon (Eng Div., Samsung C&T, 26 Sangil-ro 6-gil, Gangdong-gu, Samsung GEC - Tower B, Seoul 05288, South Korea, iam0min.kwon@samsung.com) and Daewook Jung (RiverwayENG, Seoul, South Korea)

Floor impact-borne sound in the multistory residential buildings has been a serious social issue in Korea for a long time. The housing and land authority in a ministry of Korea recently amended the floor impact-borne sound evaluation method and criteria in response to the acoustic characteristics of the actual impact sources perceived in our daily lives. However, many people have not yet been convinced by the effectiveness of those evaluation method and criteria. This study discusses spatial audito reproduction of the various floor impact-borne sounds for auditory demonstrations to increase the laypersons' understanding of the acoustic features of impact sources. The impact-borne sounds generated both by the standard impact sources used in evaluation and by some actual ones were recorded at a test-bed built in a typical apartment type. Their sound levels and reverberation times were also obtained across 1/3rd octave bands for reference. A multi-channel audio playback system engaging front/rear/ceiling speakers with subwoofers was integrated in a test-bed with a typical setting of living environment to reproduce a realistic sound field. It was optimized to create the sounds close to those actually heard in the room. The impact-borne sounds of the different damping floor systems will also be presented.

Session 3pAAb

Architectural Acoustics and Architectural Acoustics: Industry-Academia Collaboration on Architectural Acoustics II

Lucky Tsaih, Cochair
Architecture, National Taiwan University of Science and Technology, 43 Keelung Rd., Sec. 4, Taipei, 106, Taiwan

Wei-Hwa Chiang, Cochair
Architecture, National Taiwan University of Science and Technology, 43 Keelung Rd., Sec. 4, Taipei, 106335, Taiwan

Contributed Papers

2:40

3pAAb1. Harmony behind bars: Unravelling the impacts of acoustic design in prisons. James Boland (Acoust. and Vib., SLR, 214 Durham St. South, Christchurch 8100, New Zealand, jboland@slrconsulting.com) and Helen Farley (Faculty of Law, Univ. of Canterbury, Christchurch, New Zealand)

Design briefs for prisons often include criteria regulating noise from mechanical plant or break-in, partition performance and reverberation times. However, criteria are often taken (without modification) from non-carceral related standards used for design of residential or health-care projects. While carceral spaces are used for sleeping and health care, this paper will show the acoustic needs (and wants) of people inside prisons can be significantly different from those of people outside. This paper provides insights from the emergent field of sensory criminology into what makes the acoustic environment inside prisons different from other residential or health care contexts. It highlights the role that acoustics plays in the lived experience for the occupants, including incarcerated individuals and corrections staff. In prisons, acoustics facilitates communication and information gathering for incarcerated individuals, and corrections staff rely on auditory cues to gauge the prevailing tension, or "heat." The intricate dynamics of prison cultures and insights from sensory criminology can readily evade the purview of acousticians. By leveraging complementary disciplines, acousticians can more effectively design acoustics which foster pro-social communication while mitigating risks associated with undesirable social dynamics. This approach ensures the acoustic design becomes well-informed and purposeful, yielding benefits for incarcerated individuals and corrections staff.

3:00

3pAAb2. Lightweight modular construction of school classrooms, reverberation time prediction versus field testing. Joel Parry-Jones (GF, 79 Mann St., Gosford, New South Wales 2250, Australia, joelpj@pka.com.au)

Lightweight modular schools are in the early phases of being designed and constructed throughout NSW as part of Schools Infrastructure "Modern Methods of Construction (MMC)." This method proposes the use of CLT (Cross Laminated Timber) panels, timber floor and wall cassettes, and/or hybrid systems. Schools Infrastructure NSW has developed, along with PKA Acoustic Consulting's acoustic input, the "Kit of Parts" catalogue which utilises predetermined set of components that can easily be assembled and constructed. The classroom layout is therefore defined and can be incorporated into the acoustic predictive model. Additionally, the absorption characteristics of lightweight constructions differ to conventional construction and therefore should be considered. This presentation discusses the significance of construction type and fitout finishes, by comparing predicted reverb times of modular lightweight classrooms with *in-situ* absorptive testing at various cold shell, warm shell, and completed fitout stages of a school classroom and open learning space.

3:20

3pAAb3. A comparative assessment of NSW educational acoustic design guideline. Jorge Reverter Garcia (JHA Engineers, Level 20, 2 Market St., Sydney, New South Wales 2000, Australia, jorge.reverter@jhaengineers.com.au)

School Infrastructure New South Wales (SINSW) Educational Facilities Standard Guidelines (EFSG) Design Guideline 11 (DG-11) is the acoustic guideline for public educational premises in NSW. A comparative assessment of the guideline against alternative guidelines is conducted in this paper. The analysis identifies shortcomings stemming from an outdated guideline, as well as gaps in design clarity. Design solutions to meet the mandatory EFSG GD-11 acoustic requirements have been proposed to address the identified shortcomings. Additionally, we thoroughly examined DG-11's design specifications to resolve ambiguity and reduce the potential costly and over design. Our findings revealed weaknesses in SINSW DG-11 when compared to other nationally and internationally recognized standards. To create innovate and sustainable educational spaces that cater the users' needs and enhance the functionality of educational spaces, fostering of a constructive dialog is required between stakeholders and acoustic designers while preparing acoustic guidelines.

3:40

3pAAb4. Refurbishment of a vacated office building to provide high-performance music teaching and performance facilities in regional NSW. Jorge Reverter Garcia (JHA Engineers, Level 20, 2 Market St., Sydney, New South Wales 2000, Australia, jorge.reverter@jhaengineers.com.au) and Joseph Milton (JHA Engineers, Sydney, New South Wales, Australia)

In 2016, a Conservatorium of Music made a significant decision to move their music teaching and performance facilities, to a nearby vacated building situated 3km away. From the outset, the move presented the design team to deliver a state-of-the-art Conservatorium of Music, with several challenges due to the spatial and building constraints. Among the most notable of these challenges was the limited volume within the existing building which is an essential requirement for optimal music performance spaces. This paper outlines the users' expectations—which were gathered through an acoustic questionnaire, design process, details the specific challenges faced by the design team, and presents the design solutions devised to overcome these problems. The Conservatorium of Music opened in early 2023 and has been well received by users and audiences.

4:00

3pAAb5. Classroom acoustics: A case study of the cost-benefit of retrofitted interventions. Coralie van Reenen (Smart Places, Council for Sci. and Industrial Res., Bldg. 2A CSIR, Meiring Naude Rd., Pretoira, Gauteng 0081, South Africa, cvreenen@csir.co.za)

It is known that suitable classroom acoustics is important for effective teaching and learning. To this end, many countries have developed classroom acoustics standards. However, research shows that these standards are often not implemented. Research assessing the implementation of classroom acoustic standards internationally concludes that successful implementation is driven by mandatory standards that are part of the building codes and that the cost of compliance is a barrier. The study presented here explores the cost of upgrading classrooms to achieve a suitable reverberation time. The paper presents a case study in South Africa, where acoustic standards are generally not implemented and cost and know-how are barriers. The objective was to optimize the cost, acoustic benefit, and accessibility of acoustic interventions. Four adjacent similar classrooms were retrofitted with different sound-absorbing ceiling panels that cover a percentage of the ceiling area. The weighted sum model was used to assess the suitability of each intervention, taking into account the cost, acoustic benefit (in terms of reverberation time), and ease of access to materials. The case study demonstrates that a noticeable improvement in acoustic conditions can be achieved without significant cost and provides a basis for further research to develop simple standardized design recommendations.

WEDNESDAY AFTERNOON, 6 DECEMBER 2023 ROOM C2.3, 1:00 P.M. TO 4:20 P.M.

Session 3pAB

Animal Bioacoustics: Acoustic Ecology and Biological Soundscapes II

Laura Kloepper, Chair

Biological Sciences, University of New Hampshire, 38 Academic Way, Durham, NH 03824

Contributed Papers

1:00

3pAB1. Arnoux's beaked whales acoustically misidentified as Type C killer whales in Prydz Bay, Antarctica. Lucinda E. Chambers (Ctr. for Marine Sci. and Innovation (CMSI), UNSW Sydney, Sydney, New South Wales, Australia, lucinda.chambers@student.unsw.edu.au), Jane McPhee-Frew, and Tracey L. Rogers (Ctr. for Marine Sci. and Innovation (CMSI), UNSW Sydney, Kensington, New South Wales, Australia)

The Southern Ocean is home to five ecotypes of killer whale, which differ in diet, morphology, behaviour, and vocal repertoire. Recently, fish-eating Type C killer whales were reported in Prydz Bay, Antarctica from sounds recorded in hydrophone data collected in March 2017, on the basis that these sounds bore similarities to acoustic behaviour of Icelandic fish-eating killer whales. Here we re-examine the structure of the sounds identified as Prydz Bay Type C killer whales and we identify the calls to be those of the Arnoux's beaked whale. The whistles are more structurally similar and fall within the same frequency range as those of the Arnoux's beaked whale, rather than the killer whale. Of the two killer whale types observed in Prydz Bay, the most common had behaviour and morphology typical of Type B killer whales, while the other had morphology typical of Type A. Although we show that the sounds are not likely to be killer whale Type C calls, this would be the first recorded occurrence of Arnoux's beaked whales in Prydz Bay. Arnoux's beaked whales are a rarely seen, cryptic whale species.

1:20

3pAB2. Passive acoustic monitoring of artificial reefs south of Long Island, New York provides novel insights on fish, marine mammal, and post-pandemic human activity occurring in nearshore coastal habitats. Melissa Leone (Stony Brook Univ., Southampton, NY) and Joseph D. Warren (Stony Brook Univ., 239 Montauk Hwy, Southampton, NY 11968, joe.warren@stonybrook.edu)

Passive acoustic recorders were deployed at two artificial reef sites (25 m water depth) south of Long Island, New York from 2020 to present. The study sites are popular recreational fishing locations but also near commercial fishing activities and shipping lanes for the Port of New York and New Jersey providing numerous types of human-generated signals which can be monitored. We investigated how human activities changed the nearshore soundscape following the global pandemic and found that periods (2020) following the pandemic shutdown were (in some regards) louder instead of quieter than soundscapes later on (2021). Peaks in power spectral densities recorded at the reef sites often corresponded to signals from idling or transitory vessels. In addition to human-produced sounds, signals from both fish and marine mammals are regularly heard on the reefs which provided the opportunity to investigate several ecological questions including: reproductive activity of commercially important cod and weakfish; residency of bottlenose dolphins through individual signature whistle analysis; and baleen whale usage of the nearshore habitat.

1:40

3pAB3. The soundscape of the southern Gulf of Mexico (Veracruz and Campeche). Carmen Bazúa Durán (Facultad de Ciencias, UNAM, Mexico, Circuito exterior s/n, Ciudad Universitaria, Mexico, CDMX 04510, Mexico, bazua@unam.mx) and Nataly Morales Rincón (Posgrado en Ecología y Pesquerías, Instituto de Ciencias Marinas y Pesquerías, Universidad Veracruzana, Boca del Río, Veracruz, Mexico)

The southern part of the Gulf of Mexico is a large area where little is known about its soundscape. We measured the environmental noise in Laguna de Términos, Campeche, México from 2004 to 2008 in stations distributed homogeneously in this lagoon during 17 sampling periods using a digital audio tape recorder sampling at 48 kHz with 16 bits for 1 min at each station. We also measured the environmental noise off Alvarado, Veracruz, México in June 2023 using a digital audio tape recorder sampling at 96 kHz with 24 bits for 50 min at four stations. The environmental noise in the recordings was measured with a semi-automatic MATLAB routine designed for this purpose. We will present preliminary results on the environmental noise in these two coastal locations using snapping shrimp sounds as a measure to depict differences in time and space within and between locations. Changes in submarine environmental noise are helping us in understanding the effects of natural phenomena and antropogenic sounds in the distribution and abundance of marine wildlife, such as snapping shrimps and bottlenose dolphins. [Work supported by CONACyT-Campeche, PAPIIT&PASPA-UNAM, SMM, and UV.]

2:00

3pAB4. Biodiversity and big data to investigate the link between biosonar sensing and flight control in bats. Rolf Müller (Mech. Eng., Virginia Tech, ICTAS II, 1075 Life Sci. Cir, (Mail Code 0917), Blacksburg, VA 24061, rolf.mueller@vt.edu), Yihao Hu (Elec. and Comput. Eng., Virginia Tech, Blacksburg, VA), Bibi Junaidi (Dept. of Mathematics, Universiti Brunei Darussalam, Bandar Seri Begawan, Brunei Darussalam), Samuel Kramer (Mech. Eng., Virginia Tech, Blacksburg, VA), Chi Nnoka (Dept. of Mech. and Aerosp. Eng., Univ. at Buffalo, Buffalo, NY), Phillip M. Parisi (Pacific Northwest National Lab., Sequim, WA), Jungsoo Park (Mech. Eng., Virginia Tech, Blacksburg, VA), Nicholas K. Rock (Dept. of Phys., Univ. of Rhode Island, South Kingstown, RI), Jared Shing (Mech. Eng., Virginia Tech, Blacksburg, VA), Udhaya Srinivasan (Dept. of Biomedical Eng., National Univ. of Singapore, Singapore, Singapore), Muhammad Syafi'ie Su'eif (Faculty of Sci., Universiti Brunei Darussalam, Bandar Seri Begawan, Brunei Darussalam), Danny Yessayan (Mech. Eng., Virginia Tech, Blacksburg, VA), Margaret Zhang (School of Mech. Eng., Georgia Tech, Atlanta, GA), Ulmar Grafe (Faculty of Sci., Universiti Brunei Darussalam, Bandar Seri Begawan, Brunei Darussalam), Chandratilak De Silva Liyanage, Owais Ahmed Malik, and Wee Hong Ong (School of Digital Sci., Universiti Brunei Darussalam, Bandar Seri Begawan, Brunei Darussalam)

Any animal with dexterous mobility in complex natural environments needs powerful systems for sensing and mobility that also have to be well-integrated with each other. Echolocating bats that hunt in dense vegetation rely on biosonar integrated with flapping flight to accomplish this. The low-dimensional nature of the biosonar inputs together with the complexity of the bats' flight apparatus make this coupling a fundamental scientific challenge. To understand how bats can control the many degrees of freedom of their wings based on external information that is conveyed by only two one-dimensional echo trains, a flight tunnel has been instrumented with synchronized high-speed video cameras and ultrasonic microphones. The array recordings can provide data volumes that are large enough to enable deep-learning analyses of the biosonar echoes and the flight kinematics. These analyses will be applied to reducing the dimensionality of ultrasonic inputs, the kinematic outputs, and discovering the relationship between the essential dimensions of biosonar and flight. The tunnel has been installed on the island of Borneo to take advantage of the local bat biodiversity as a source of variability that can be exploited in a comparative approach to identify functional traits that are essential to sensorimotor integration in echolocating bats.

2:20

3pAB5. Extraction of acoustic features of calls of five *Platypleura* species using the field recordings. IKUO MATSUO (Tohoku Gakuin Univ., Shimizukoji 3-1, Wakabayashi-ku, Sendai 9848588, Japan, matsuo@mail.tohoku-gakuin.ac.jp)

Five *Platypleura* species are distributed in the south of Amami Island in Japan. At particular, *P. albivannata*, and *P. yayeyamana* are distributed at the same area in Ishigaki island. The survey of *P. albivannata*, which is endangered species, is important problems for conservation. Although it is necessary to discriminate between these species, it is difficult to discriminate between them from the morphology, that is, visual information. It is necessary to discriminate with these species by using another cue. In this presentation, we proposed the analysis methods to extract acoustic features dependent on *Platypleura* species to the south of Amami Island in Japan. We primarily aimed to examine whether these classifications correspond to the resemblance in the time and frequency domain of calling songs. Furthermore, we tried to extract characteristics that can be used for species identification in monitoring endangered species, *P. albivannata*. We found that maximum peak frequencies are feasible for species identification, especially for the detection of *P. albivannata* in the wild.

2:40–3:00 Break

3:00

3pAB6. Localizing multiple sound sources using an array of autonomous, asynchronous receivers. Eva-Marie Nosal (Ocean and Resources Eng., Univ. of Hawaii, 2540 Dole St., Holmes Hall 402, Honolulu, HI 96822, nosal@hawaii.edu) and Tom Fedenczuk (Abakai Int. LLC, Waianae, HI)

Time-difference-of arrival (TDOA) methods are well established and commonly used to localize sound sources in bioacoustic applications. TDOA methods generally require/assume that the receivers of the acoustic array are time-synchronized. Time synchronization can be achieved in various ways, including by connecting the receivers to a common data aquisition device, or by using control/calibration sources of known location to time synchronize. In this talk, we introduce a TDOA approach that can be used with an array of autonomous, asynchronous receivers. The approach requires that multiple sources are detected over the array, and localizes the sources simultaneously in addition to solving for clock offsets between the receivers. This approach offers potential cost and time savings since it can be used with autonomous receivers and since it does not require control/calibration sources. Simulation results will be used to illustrate and explore the approach, and results from an experiment using asynchronous microphones will be presented. [Work supported in part by ONR.]

3:20

3pAB7. Real-time swarming detection in honeybees: Leveraging audio signal processing and machine learning techniques. Iman Ardekani (Computing and Information Technol., New Zealand Inst. of Skills and Technol. - Te Pukenga, 139 Carrington Rd., Mt Albert, Auckland 1025, New Zealand, imanog@yahoo.com), Soheil Pour, and Hamid Sharifzadeh (Computing and Information Technol., New Zealand Inst. of Skills and Technol. - Te Pukenga, Auckland, New Zealand)

Acoustical signals play a significant role in honey bee communication across various behavioral contexts. One notable example is queen piping, which refers to the acoustic signals emitted by young queens during the swarming process. Specifically, emerged virgin queens emit a distinct acoustic signal known as "tooting." Tooting signals typically consist of one or two pulses lasting approximately one second, characterized by an initial rise in both amplitude and frequency. Recognizing these signals is of utmost importance for beekeepers as it enables them to identify an imminent swarm, effectively manage the swarming process, and safeguard the colony. This paper proposes a novel method for online swarming detection in honey bees, leveraging audio signal collection, acoustical signal processing, and

machine learning techniques. The experimental results demonstrate the effectiveness of the proposed system in accurately detecting swarming at its early stages. Notably, the system maintains a high level of accuracy even when confronted with environmental noise and other unfamiliar audio signals that could potentially corrupt the audio signal acquired from the beehive.

3:40

3pAB8. Evolution of the highest sound pressure levels in vertebrate communication calls. Benjamin J. Walker (Evolution and Ecology Res. Ctr., UNSW Sydney, Sydney, New South Wales, Australia, ben.walker@unsw.edu.au), Neil R. Jordan (Ctr. for Ecosystem Sci., UNSW Sydney, Sydney, New South Wales, Australia), and Tracey L. Rogers (Ctr. for Marine Sci. and Innovation (CMSI), UNSW Sydney, Kensington, New South Wales, Australia)

The loudest communication calls produced by vertebrates have been proposed to have a physical limit. However, this idea is at odds with the implicit assumption that loudness is driven by body size, where each section of an acoustic system (e.g., vocal tract, pharyngeal space, lungs) increases in size as an animal gets larger. A physical limit to loudness may exist for animals with different acoustic systems, but this has not been explored using comparative analysis. We compare the loudest communication calls across 80 vertebrate species and show the assertion of a physical limit to loudness exists for animals with particular acoustic systems but breaks down where others scale with body mass. We show that for vertebrates with air sacs, loudness scales with body mass but it approaches a physical limit. We also show that air sac location can affect loudness, where louder calls are correlated with air sacs that filter (after the sound source) as opposed to air sacs that increase air volume (before the sound source). We examine ecological and evolutionary pressures which may have led to these modifications and their influence on vocal signaling.

4:00

3pAB9. Evolution of the highest sound pressure levels in mammalian echolocation calls. Benjamin J. Walker (Evolution and Ecology Res. Ctr., UNSW Sydney, Sydney, New South Wales, Australia, ben.walker@unsw.edu.au), Neil R. Jordan (Ctr. for Ecosystem Sci., UNSW Sydey, Sydney, New South Wales, Australia), and Tracey L. Rogers (Ctr. for Marine Sci. and Innovation, UNSW Sydney, Kensington, New South Wales, Australia)

Recent work has shown that for some echolocating bat species there is a physical limit to the loudness of their echolocation signals, and this idea has been extended to suggest there is a physical limit to the loudness of echolocation signals of all species. This is at odds with the assumption that the loudness of a species call is driven by their body size, where louder species are larger species. This is also at odds with the prey detection hypothesis, where for echolocation calls the successful detection of prey at size extremes (both minima and maxima) drives the loudness of echolocation calls. A physical limit of loudness in echolocation calls may exist, but this has not been explored using comparative analysis. We compare the loudest echolocation calls across 63 echolocating mammals (including almost half of all aquatic echolocators) and show the assertion of a physical limit to loudness exists for bats in terrestrial habitats but breaks down for odontocetes in aquatic habitats. For aquatic echolocators, the loudness of their echolocation calls scales with body mass. We examine evolutionary and ecological pressures which may have led to these differences, and their influence on echolocation loudness.

Session 3pAO

Acoustical Oceanography, Underwater Acoustics and Physical Acoustics: Observing the Ocean Acoustically using Submarine Cable Systems II

Shima Abadi, Cochair
University of Washington, 1501 NE Boat St., Seattle, WA 98195

Ying-Tsong Lin, Cochair
University of California, San Diego, Scripps Institution of Oceanography, La Jolla, CA 92037

Bruce Howe, Cochair
Ocean and Resources Engineering, University of Hawaii at Manoa, 2540 Dole Street, Holmes Hall 402, Honolulu, HI 96822

Haris Kunnath, Cochair
Environment, CSIRO, GPO Box 1538, Hobart 7001, Australia

Invited Paper

1:00

3pAO1. A cabled ocean acoustic array to be installed in the Gulf of Maine. Jennifer Miksis-Olds (Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824, j.miksisolds@unh.edu), Debra Brask (SubCom, Newington, NH), David Buck (Univ. of New Hampshire, Durham, NH), Art Cole (JASCO Appl. Sci., Dartmouth, NS, Canada), Leon Hashem (SubCom, Newington, NH), Jack Hennessey (JASCO Appl. Sci., Dartmouth, NS, Canada), Anthony P. Lyons (Univ. of New Hampshire, Durham, NH), Bruce S. Martin (JASCO Appl. Sci., Dartmouth, NS, Canada), Theresa Ridgeway (Univ. of New Hampshire, Durham, NH), Richard Rogers (Ocean Specialists, Inc., Stuart, FL), and David Willoughby (Ocean Specialists, Inc., Stuart, FL)

It is often noted that we know more about the planets than the processes of our oceans. A network of ocean observatories is improving our knowledge; however, measurements are still relatively scarce and do not provide enough data to validate and refine models of ocean currents, soundscapes, and biological activity. A cabled acoustic array is being developed for deployment in 2024 in the coastal waters of the Gulf of Maine to complement existing oceanographic monitoring infrastructure. The system will make measurements of the nearshore (6 km) environment and shed light on the connections between coastal measurements and the overall Gulf of Maine environment. The system is designed to synoptically collect acoustic, oceanographic, and biogeochemical data, while also remaining flexible enough in design to incorporate new sensors in the future. Measurements made with this ocean asset will serve as a baseline for pattern and trend analyses of changing environmental conditions in an area of high productivity, human use, and species diversity. The data management and visualization team are developing an infrastructure to provide public access to the data and its products. Public data access will permit researchers at all levels to advance our understanding of ocean acoustics and oceanographic processes.

Contributed Paper

1:20

3pAO2. Using ocean ambient sound to sense arrival time fluctuations due to temperature. John Ragland (Univ. of Washington, 185 W Stevens Way NE, Seattle, WA 98195, jhrag@uw.edu) and Shima Abadi (Univ. of Washington, Seattle, WA)

Ambient noise interferometry is a technique that uses coherent ambient sound measured at two separate locations to estimate the Green's function between the sensors. Specifically, if the ambient sound is diffuse, the cross-correlation between sensors converges to the Green's function between the sensors. In this talk, we apply the technique of ambient noise interferometry to two Ocean Observatories Initiative hydrophones that are separated by 3.2 km and bottom mounted at a depth of 1500 m. It has previously been shown that these hydrophones can passively estimate several multi-path propagation peaks reliably throughout the 8 years that they have been recording ambient sound. We present an algorithm that estimates the arrival time of these peaks using the empirical Green's function and compare these estimated arrivals to simulated acoustic arrivals. We demonstrate that the arrival times estimated with ambient sound show good agreement with the simulated results and have clear annual fluctuations due to seasonal temperature changes in the water column. We will discuss the future possibilities of extending this work for inversion based oceanographic measurements. [work supported by ONR.]

Invited Papers

1:40

3pAO3. Toward assimilation of acoustic travel times into an ocean state estimate. Ivana Escobar (Oden Inst., 201 E. 24th St., Austin, TX 78712, ivana@utexas.edu), Patrick Heimbach (Oden Inst., Austin, TX), and Feras Habbal (Environ. Sci. Lab., Appl. Res. Labs., Austin, TX)

Direct ocean observing systems are a scarce set of data that build the backbone for understanding the current state of the ocean. These systems are limited in areas of high variability in temperature and salinity. Pioneering work of Wunsch (1977) proposed efforts to extract detailed hydrographic information using sound propagation through the ocean, providing the scaffolding to improve our understanding of the ocean's interior. The integrated nature of such measurements makes acoustic thermometry a powerful application for monitoring regional to basin-averaged hydrographic changes in the ocean, a measurement that is difficult to achieve by individual "point" measurements (moorings, ship-borne CTD casts, or autonomous floats) alone. This work models acoustic travel times corresponding to eigenrays between a fixed source and receiver from an evolving ocean state. Inclusive computation of acoustic travel times within a dynamically evolving modeled ocean state provide a novel approach to model data comparison within a general ocean circulation model. An adjoint assimilation framework combines observations with numerical models to estimate and update oceanic state variables Forget *et al*. (2015). This process provides the necessary operators for model data comparison of ocean acoustic observable quantities within a consistent state estimation framework allowing for data assimilation from acoustic tomography measurements.

2:00

3pAO4. Energy partitioning of the underwater soundscape. David R. Barclay (Dept of Oceanogr., Dalhousie Univ., 1355 Oxford St., LSC Bldg., Halifax, NS B3H 4R2, Canada, dbarcl@gmail.com) and Najeem Shajahan (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada)

A four-element three-dimensional hydrophone array has been deployed on the Strait of Georgia node of the Victoria Experimental Network Under the Sea (VENUS) since March 2020. To the present, continuous pressure time series data from the four sensors has been streamed ashore and made available by Ocean Networks Canada. A technique for classifying and partitioning ambient noise using the three-dimensional noise coherence function is applied to the recordings. The noise coherence (directionality) due to wind generated surface noise and individual ships is analytically modelled using measured environmental inputs such as the time varying sound speed profile and sediment properties from the measurement site and compared with the observation. The theoretical coherence curve is computed by mixing the contributing sources until a best-fit is found. Since the wind generated surface noise coherence is stable and independent of the source effective sound power per unit area, the contribution of ship noise to the soundscape can be exactly determined. Applying this algorithm to the long-term data set allows the relative contribution of ship noise to the soundscape in the region to summarized succinctly. [Research supported by ONR.]

Contributed Papers

2:20

3pAO5. Enabling long-term soundscape observation via Ocean Networks Canada's acoustic Northeast-Pacific infrastructure. Lanfranco Muzi (Ocean Networks Canada, Univ. of Victoria, 2474 Arbutus Rd., 100, Victoria, BC V8N 1V8, Canada, muzi@oceannetworks.ca), David R. Barclay, Brendan Smith (Oceanogr., Dalhousie Univ., Halifax, NS, Canada), Kohen Bauer, Fabio De Leo, Martin Heesemann, Steven Mihály, Martin Scherwath, John Dorocicz, Paulo Correa, Jesse Hutchinson, and Jasper Kanes (Ocean Networks Canada, Univ. of Victoria, Victoria, BC, Canada)

Data time series covering extended periods of time in support of long-term environmental studies are particularly difficult and expensive to collect, especially at offshore locations. Ocean Networks Canada's (ONC) undersea cabled observatories in the Northeast Pacific ocean have been collecting an extensive array of oceanographic, seismic, geophysical, biological and acoustical data over periods that, in some cases, extend beyond sixteen years. With regard to soundscape studies, ONC currently owns and operates 22 hydrophones (including four 4-element, three-dimensional arrays) between the VENUS coastal observatory in the Salish Sea and the NEPTUNE offshore deep-sea observatory in the Northeast Pacific Ocean. The data, streamed in quasi-real-time to ONC's web data portal, offer a window on a number of different environments, from the busy, shallow waters of the Salish Sea to the 2200 m of depth of the Endeavour hydrothermal-vent field. This presentation gathers highlights from the latest research utilizing ONC's acoustic infrastructure and data, ranging from ambient noise to hydrothermal-vent soundscapes, seismic events, bioacoustics, and signal processing applications such as source localization.

2:40

3pAO6. Underwater noise from submarine turbidity currents. Alex E. Hay (Oceanogr., Dalhousie Univ., Halifax, NS, Canada), Matthew Hatcher (Oceans Graduate School, Univ. of Western Australia, 35 Stirling Terrace, Albany, Western Australia 6330, Australia, matt.hatcher@uwa.edu.au), and John E. Hughes Clarke (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, Durham, NH)

Submarine landslides and associated turbidity currents can be orders of magnitude larger than their terrestrial equivalents. Their prevalence globally is hugely underrepresented by *in situ* observations. In addition to their importance in mass transport and morphologic change they can pose significant threat to coastal communities and subsea infrastructure. This presentation focusses on noise measurements from turbidity currents flowing down the face of a fjord delta and how they can inform the use of passive acoustics as a tool for future observations of these under sampled phenomena. Noise was measured underwater at frequencies from 1 to 1200 kHz. The noise spectra are consistent with sound generation by collisions among sand-sized particles within the turbidity current, also known as acoustic sediment-generated noise or SGN. The spectra from the leading head of the current extend to higher frequencies than those from the trailing body, indicating that collisions were between finer-grained particles in the head and coarser-grained particles in the body. Noise intensity increased 100-fold for a two-fold increase in current head speed, consistent with the expected collision rate for granular materials in the high-flow gas-like phase and highly turbulent particle-laden flows.

3:00

3pAO7. Boundary Pass Underwater Listening Station—Data management and processing. David E. Hannay (JASCO Appl. Sci., Victoria, BC, Canada, david.hannay@jasco.com), Connor Grooms, Zizheng Li, and Jorge Quijano (JASCO Appl. Sci., Victoria, BC, Canada)

The Boundary Pass Underwater Listening Station, sponsored by Transport Canada, Port of Vancouver and JASCO Applied Sciences, has continuously operated two compact tetrahedral hydrophone arrays beneath the busy shipping lanes leading to Vancouver, Canada since June 2020. The arrays are deployed 300 m apart on the seabed at 190 m depth and are cabled 2.6 km to shore. A high sample rate of 512 kHz on all eight channels, with 24-bit samples, produces almost 400 TB/yr of raw acoustic data. Automated real-time data analysis calculates ambient noise statistics, detects marine mammal sounds, measures underwater radiated noise of individual ships, and creates spatial noise maps of the hulls of some vessels passing closely overhead. The derived data products are stored in an easy-to-maintain online database of just a few TB/yr. Results in graphic, tabular and compressed audio formats are made available to scientists and project managers through an intuitive web interface. The project follows international standards for ship noise and ambient noise calculations, so the results can be compared with measurements made elsewhere.

WEDNESDAY AFTERNOON, 6 DECEMBER 2023 ROOM C2.2, 1:35 P.M. TO 4:00 P.M.

Session 3pBAa

Biomedical Acoustics and Physical Acoustics: Biomedical Acoustics in Pulmonology

Xiaoming Zhang, Cochair
Radiology, Mayo Clinic, 200 1st St. SW, Rochester, MN 55905

Libertario Demi, Cochair
University of Trento, Via Sommarive 9, Trento, 38123, Italy

Chair's Introduction—1:35

Invited Paper

1:40

3pBAa1. Single and multiple scattering quantitative ultrasound methods in the lung. Azadeh Dashti, Roshan Roshankhah, Marie Muller (Mech. and Aerosp. Eng., North Carolina State Univ., Raleigh, NC), and Jonathan Mamou (Radiology, Weill Cornell Medicine, 416 East 55th St., B1, New York, NY 1022, jom4032@med.cornell.edu)

Lung ultrasound images are qualitative and rely on subjective interpretation for diagnosis. Ideally, quantitative ultrasound (QUS) methods applied to lung ultrasound data would provide biomarkers associated with specific diseases in an absolute fashion. While QUS methods are successful in numerous organ systems in preclinical and clinical applications, their direct application to lung ultrasound data requires taking multiple scattering (MS) into account. This presentation shows QUS methods independently exploiting MS and single scattering (SS) components of radio-frequency (RF) data, acquired from lungs from rodent models of pulmonary edema and fibrosis. SS and MS components were obtained using singular value decomposition and eigenvalue thresholding. Spectral and envelope QUS parameters were computed from the SS component. MS-based QUS parameters were obtained by extracting the diffusion constant and tracking the SS intensity decay rate. Stepwise linear discriminant analyses using only three QUS parameters yielded strong correlations between wet-to-dry ratio in healthy and pulmonary edema rats. Strong correlations were also obtained between QUS parameters and modified histological Ashcroft scores in healthy and pulmonary fibrosis rats. These results demonstrate the value of QUS parameters obtained from SS and MS approaches for assessing lung diseases with high specificity and sensitivity. [Supported in part by R21HL154156 and W81XWH1810101.]

Contributed Paper

2:00

3pBAa2. Noninvasive measurement of lung elastic properties. Xiaoming Zhang (Radiology, Mayo Clinic, 200 1st St. SW, Rochester, MN 55905, Zhang.Xiaoming@mayo.edu)

Many lung diseases are associated with dramatic changes in mechanical properties of the lung. Chest radiographs and computed tomography (CT) are the most common clinical methods for evaluating lung disease, but they are expensive, use radiation, and do not measure lung stiffness. Shear wave elastography (SWE) uses ultrasound radiation force (URF) to generate wave propagation in the tissue. However, URF should not be applied to the lung, because there are air and blood in the alveoli. Relatively high-intensity ultrasound can cause alveolar hemorrhage and lung injury. We have developed a noninvasive, and clinically feasible technology, lung ultrasound surface wave elastography (LUSWE), that is capable of measuring lung elastic properties with speed and accuracy. LUSWE uses a handheld shaker to generate a local small harmonic vibration on the chest surface. The resulting wave propagates through the intercostal muscle and propagates on the lung. The surface wave speed is noninvasively measured using an ultrasound technique. LUSWE is safe because diagnostic ultrasound is only used to detect the wave propagation along the lung duo to mechanical wave generation on the chest. LUSWE may be used to assess multiple lung disorders such as lung fibrosis or lung edema.

Invited Papers

2:20

3pBAa3. Application of lung ultrasound surface wave elastography in the evaluation of diffuse lung diseases. Sanjay Kalra (Pulmonary & Critical Care Medicine, Mayo Clinic, 200 First St. SW, Rochester, MN 55905, kalra.sanjay@mayo.edu), Brian Bartholmai, and Xiaoming Zhang (Radiology, Mayo Clinic, Rochester, MN)

The evaluation of diffuse lung disease invariably involves chest computerized tomography (CT). The detailed information obtained is often equivalent to that from lung biopsies. It, however, involves exposure to ionising radiation which limits its repeated application in monitoring and follow up. Ultrasound techniques offer potentially safer/cheaper alternatives and lung ultrasound surface wave elastography (LUSWE) is a novel, noninvasive, and clinically feasible technique to quantify lung stiffness, an important biomechanical property that changes in diffuse fibrotic and infiltrative lung diseases. We have developed a sensitive, accurate and reproducible technique to measure changes in lung surface stiffness by using surface wave transmission speed and demonstrated its value in detecting and staging the degree of fibrosis in diffuse interstitial lung diseases, specifically idiopathic pulmonary fibrosis and progressive systemic sclerosis associated lung disease, as well as changes in heart failure. The technique consists of ultrasound detection of the propagation speed of a wave generated by the vibrations from an indenting shaker in multiple intercostal spaces to provide a representative assessment of both lungs. It offers the prospect of simple, radiation-free, repetitive assessment of changes in acute and chronic diffuse lung diseases with the potential for easy and rapid bedside use, all important considerations in clinical practice.

2:40–3:00 Break

3:00

3pBAa4. State of the art in diagnostic lung ultrasound, from physics to clinics. Libertario Demi (Univ. of Trento, Via Sommarive 9, Trento 38123, Italy, libertario.demi@unitn.it)

Starting from the new international guidelines and consensus on the use of lung ultrasound, which have been published in 2023 [https://doi.org/10.1002/jum.16088], and by presenting the latest clinical results on quantitative lung ultrasound spectroscopy, this lecture will discuss the key technical challenges ahead for the field of lung ultrasound (LUS). These concerns the development of standardized imaging protocols and scoring systems, the application of AI-driven solutions to the analysis and classification of LUS data, the applicability of numerical wave propagation models, the design and fabrication of controllable and reproducible lung mimicking models, the understanding of the dominant physical mechanisms at play in the interaction between ultrasound waves and lung tissue, and the development of dedicated and quantitative solutions for the monitoring and diagnosis of lung diseases.

3:20

3pBAa5. Innovative AI techniques to aid clinical diagnosis of complex lung pathologies in ultrasound images. Maria Antico (Australian e-Health Res. Ctr., CSIRO, 296 Herston Rd., Brisbane, Queensland 4006, Australia, maria.antico@csiro.au), Khalid Moafa, Damjan Vukovic, Christopher Edwards (School of Clinical Sci., Queensland Univ. of Technol., Brisbane, Queensland, Australia), Jason Dowling (Australian e-Health Res. Ctr., CSIRO, Brisbane, Queensland, Australia), David Canty (Dept. of Surgery (Royal Melbourne Hospital), Univ. of Melbourne, Melbourne, Victoria, Australia), Marian Steffens, and Davide Fontanarosa (School of Clinical Sci., Queensland Univ. of Technol., Brisbane, Queensland, Australia)

Lung ultrasound (LUS) has recently gained increasing interest as a reliable method for point of care (POC) diagnostics and management of lung diseases. LUS is easily accessible, has no radiation-related risks and is portable, so it does not require relocating the patient, minimising the risk of further infection. This research aims to develop efficient and automated methods for lung pathology diagnosis using AI to support clinicians. We have introduced a binary classifier based on a state-of-the-art Swin Transformer to discriminate between LUS clips of healthy patients and patients with interstitial lung disease (ILD). Differently from previous approaches, this is better aligned with the current clinical assessments of ILD since it evaluates LUS clips instead of single frames; and does not require any additional annotations from clinicians, since its training is based only on the already available medical report. Furthermore, we propose an unsupervised deep learning approach (based on a generative adversarial network) to convert in real-time US volumes to MRI-like volumes of the thoracic region. This approach can be used to spatially localise the US probe with respect to the MRI in real-time and gather anatomical contextual information of the imaged region, providing thus guidance to clinicians.

Contributed Paper

3:40

3pBAa6. Automatic detection of lung ultrasound b-lines over heart cycles for assessing pulmonary edema. Ngoc Thang Bui (Radiology, Mayo Clinic, Rochester, MN), Charlie E. Luoma (Cardiology, Mayo Clinic, Rochester, MN), and Xiaoming Zhang (Radiology, Mayo Clinic, 200 1st ST SW, Rochester, MN 55905, Zhang.Xiaoming@mayo.edu)

Excess extravascular lung water (EVLW) is a common consequence for patients with congestive heart failure, inflammatory conditions, or acute respiratory distress syndrome. Computed tomography (CT) is widely used to assess EVLW, but with a higher radiation dose and logistical complexity. Lung ultrasound (LUS) using B-line artifacts demonstrates a reasonable correlation with EVLW, but user experience with the acquisition technique, equipment-dependent factors and the subjective analysis of the B-line artifacts results in significant inter-observer variability. We propose a technique to automatically detect lung ultrasound B-lines over heart cycles. The method first extracts B-mode image features from videos in a region of interest (ROI). A ResNet50-UNet network is then applied to segment regions of B-lines. This technique was performed on 8 videos (i.e., 4 patients versus 4 controls with the duration of each video 2–3 s, resolution 1024×758 pixels, and frame rate 28 fps). The training accuracy and loss value of proposed models were 96.5% and 0.05, respectively. The paired t-test statical analysis shows that there is significant difference ($p\text{-}value = 0.0035$) of total number of B-line between normal and lungs with edema. The proposed method improves the accuracy for counting the B-lines and reduces the labor work and intra-observer variability from different users.

Session 3pBAb

Biomedical Acoustics and Physical Acoustics: Bridging Preclinical and Clinical Acoustics II

Misun Hwang, Cochair
Radiology, Children's Hospital of Philadelphia, 3401 Civic Center Blvd, Philadelphia, PA 08057

Shashank Sirsi, Cochair
Bioengineering, Univ. of Texas at Dallas, Richardson, TX 75080

Contributed Papers

1:40

3pBAb1. Clinical Implementation of Combined Quasi-Static Ultrasound Elastographic and Regional Blood Flow Assessment Techniques, in Areas of Forearm Myofascial Dysfunction, Before and After Osteopathic Physical Manipulation. Chandhana Pedapati (School of Osteopathic Medicine, A.T. Still Univ., Mesa, AZ), James Keane (School of Osteopathic Medicine, A.T. Still Univ., Mesa, AZ), Timothy Durr (School of Osteopathic Medicine, A.T. Still Univ., Mesa, AZ), Deborah M. Heath (School of Osteopathic Medicine, A.T. Still Univ., Mesa, AZ), Douglas Mast (Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH), Curtis Bay (Arizona School of Health Sci., A.T. Still Univ., Mesa, AZ), and Inder Raj S. Makin (School of Osteopathic Medicine, A.T. Still Univ., 5850 E Still Circle, Mesa, AZ 85215, imakin@atsu.edu)

A fast-growing area of high-resolution ultrasound imaging is localizing painful regions termed myofascial tender points (MTP). To mitigate pain and reduce musculoskeletal nodularity and stiffness, osteopathic physicians perform loco-regional osteopathic manipulative procedures (OMT). Current MTP-diagnostic and clinical efficacy criteria using sono-elastographic, or regional flow biomarkers are poorly characterized. This research aims to develop a well-grounded approach to identify forearm MTPs and track changes post-physical interventions, using elastography combined with regional power Doppler. A 44-subject clinical study cohort, was randomly divided in OMT, mild exercise and "rest" groups: imaging was performed using an open-platform SonixTouch Q+. During controlled tissue deformation (4.5-5 Hz), ultrasound elastography was performed by computing root-main-square (RMS) strain of beamformed ultrasound echoes (13 MHz, 15 fps) for 3 s. Similarly, power Doppler maps were averaged over 5 seconds (5–12 fps), to evaluate regional blood flow. Data acquisition and processing sequence was repeated post-intervention. Changes in tissue stiffness as well as regional blood flow were compared pre- and post-intervention for all three human subject groups. With the experimental technique used in this study, a significant reduction in tissue stiffness, and increase in regional blood flow is demonstrated in the OMT group compared to the exercise and rest group.

2:00

3pBAb2. Skull bone acoustics properties in children with specific language disorders. ibrahim y. elnoshokaty (Acoust., ENOSH Sci. Ctr., 22 Villa Mohamed Mahmoud Kassem St., Elhegaz Sq., Cairo, Heliopolis 11431, Egypt, ibrahim@enoshmink.com)

The acoustic properties of the skull OF children with specific language disorders and how they might affect hearing were investigated. Broadband noise was isolated throw 2 procedures first by all tests in the audiology room and second by a tailored isolated helmet with six bone mics attached to skull gaps and spectrally analyzed using a Fast Fourier Transform and in 1/3-octave bands. Energetic peaks were found centered near 63 and 125 Hz, and all on the left side of the skull (e.g., range greater than 10 dB around 900 Hz). Acoustic patterns from each skull were subsequently compared with air and bone conduction sensory thresholds. Individual skull patterns reliably correlated with bone conduction thresholds, but not air conduction thresholds, indicating a possible mediating role of the skull to hearing.

2:20

3pBAb3. Delivering wireless ultrasound energy to remote systems: Design implications for powering medical implants requiring milliwatt versus several watts of power. Inder Raj S. Makin (School of Osteopathic Medicine, A.T. Still Univ. of Health Sci., 5850 E Still Circle, Mesa, AZ 85215, imakin@atsu.edu), Paul Jaeger, Harry Jabs, Thomas P. Ryan (Piezo Energy Technologies, LLC, Mesa, AZ), Douglas Mast (Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH), and Leon J. Radziemski (Piezo Energy Technologies, LLC, Mesa, AZ)

Current medical device technological advances either involve, miniaturization or wireless operation or both. Miniaturization enables implant placement closer to target organs, e.g., blood vessels, viscera, or neurological structures, while wireless powering of larger left-ventricular assist devices (LVAD) is desired to mitigate infection risk due to wires. Ultrasound is a viable energy modality, wirelessly propagating through material media (tissue) and either directly powers systems, or charges an integrated battery. Whether powering implantables, or non-implantable systems, such as digital devices, ultrasound power transfer approaches have to be application-specific. This talk will present data and development strategies for four distinct use-cases requiring respectively, $<$5 mW (miniaturized systems), $\sim$500 mW (implantable pulse generators, IPG), $\sim$2 W (non-implanted digital systems), and $\sim$8 W (LVADs). Choice of low-MHz frequency, planar single-element or coarse array transmit sources, as well as application-dependent receivers a few wavelengths in size (low-mW), to 45 mm diameter ($\sim$8 W), will be described. Benchtop experiments demonstrate successful ultrasound charging of a miniaturized solid-state (250 mAh) battery within 20 minutes. Results from live porcine model studies show a 200 mAh Li-ion battery within a Ti-shelled IPG, being successfully charged without thermal-effect-related tissue changes in the propagation path. [Work partially supported by NIH/NIBIB R43EB019225.]

2:40–3:00 Break

3:00

3pBAb4. Auditory brainstem response to different chirp (based on cochlear partition) stimulation rates. Sergio Mora Camargo (Eng., Tec de Monterrey, 13 Mayo, 17, Mexico City 03820, Mexico, serch_mcam@live.com), David I. Ibarra-Zarate, and Luz Maria Alonso-Valerdi (Eng., Tec de Monterrey, Monterrey, Mexico)

All through the last decades, hearing thresholds have been estimated by audiologists by different means, such as pure-tone assessments, auditory steady state response, auditory brainstem response, etc. Objective EEG-

based measurements have been applied, such as the auditory brainstem response (ABR), which measures neural synchrony along the brainstem auditory pathway. Through ABR waves' amplitude and latency due to a stimulus level, hearing thresholds can be estimated. Considered a short-latency auditory evoked potential (1–20 ms), very short transient stimulus (0.1–10 ms) is needed to evoke ABR. Chirp is a stimulus that has demonstrated to evoke better response, compared with clicks (aprox twice responses amplitude). Chirp is a transient auditory stimulus designed that considers delay in basilar membrane tonotopic gradient inside the cochlea. Chirp auditory response has been suggested to represent higher amplitudes, lower latency and lower time to identify thresholds. Characterization of ABR morphology due to chirp stimulus is something that has still been studied since multiple chirp designs have been proposed to identify changes in response due to chirp parameters differences. In the present work a description of Chirp design and detailed ABR extraction is presented, looking for a comparison between 4 Chirp stimulation rates as described in literature.

3:20

3pBAb5. A mathematical model for maximizing drug uptake in low-EPR tumors using different methods of ultrasound-mediated drug delivery. Mohammadaref Ghaderi (Bioengineering, the Univ. of Texas at Dallas, 800 W Campbell Rd., Richardson, TX 75080, mxg190012@utdallas.edu) and Shashank Sirsi (Bioengineering, the Univ. of Texas at Dallas, Richardson, TX)

Cancer remains a significant health burden and research efforts aim to improve drug delivery options. Focused ultrasound has emerged as a potential mechanism for enhanced targeted drug delivery in tumors with a low Enhanced Permeability and Retention (EPR) effect by "sonoporation". Traditional sonoporation has shown promise, but it can still benefit from improvements in drug uptake. The current study investigates the use of a new non-portative technique, called "ultrasonic tumor painting," to enhance vascular retention of nanoparticles in the tumor space to achieve targeted drug delivery and increase therapeutic efficacy. An existing mathematical model was adapted to simulate the concentration of free and liposome encapsulated doxorubicin in circulation. The simulations were performed in MATLAB and specifically evaluate doxorubicin uptake in low EPR tumors. To increase drug uptake, sonoporation was employed where efficiency of intratumoral liposomal doxorubicin uptake was increased. An alternative non-permeabilizing strategy, called "tumor painting," was utilized to improve vascular retention instead of increasing permeability. After retention, the small molecule doxorubicin is allowed to leak into the tumors space via concentration mediated Fickian Diffusion. This study demonstrates that ultrasound-based drug delivery can be improved with tumor painting. This method is a valuable tool to overcome the limitations of traditional sonoporation and has exciting potential for enhancing the therapeutic efficacy of chemotherapy.

3:40

3pBAb6. Higher education audiometric research: HEAR—Onsite Assessment of Conservatoire Students. Stephen Dance (Built Environment and Architecture, LSBU, 103 Borough Rd., London, Lambeth SE1 0AA, United Kingdom, dances@lsbu.ac.uk), Ruben Vazquez Amos (Built Environment and Architecture, LSBU, London, Lambeth, United Kingdom), and Georgia Zepidou (Acoust., AECOM, London, United Kingdom)

Through analysis of the ISO 1999 dataset it was found that half of noise induced hearing damage occurs in the first three years of adult life. As a result the Higher Education Audiometric Research project was founded in the UK with the ambition of minimizing this damage through educational awareness and on-site testing and assessment. In the UK, 50% of this age group are in higher education. Through the newly established UK Hearing Conversation Association awareness was raised through the "Love Sound – Listen with Care" campaign. Using long-term links with the Royal Academy of Music two on-site hearing screening methodologies were compared in a pilot study involving 23 students. The obvious advantage of onsite testing and assessment being convenience. The two methodologies were audiometry using Kudawave Prime based on active noise canceling headphones, and otoacoustic emissions using Hearing Coach software and Path Medical instrumentation. Presented are the results in terms of consistency, test-retest, and ease of use/practical application.

4:00

3pBAb7. Focused ultrasound-mediated delivery of anti-programmed cell death-ligand 1 antibody to the brain of a porcine model. Siaka Fadera (Washington University in St Louis, St Louis, Missouri, United States)

Immune checkpoint inhibitor (ICI) therapy has revolutionized cancer treatment by leveraging the body's immune system to combat cancer cells. However, its effectiveness in brain cancer is hindered by the blood-brain barrier (BBB), impeding the delivery of ICIs to brain tumor cells. This study aimed to assess the safety and feasibility of using focused ultrasound combined with microbubbles-mediated BBB opening (FUS-BBBO) to facilitate trans-BBB delivery of an ICI, anti-programmed cell death-ligand 1 antibody (aPD-L1), to the brain of a large animal model. In a porcine model, FUS sonication of targeted brain regions was performed after intravenous microbubble injection, which was followed by intravenous administration of aPD-L1 labeled with a near-infrared fluorescent dye. The permeability of the BBB was evaluated using contrast-enhanced MRI, while fluorescence imaging and histological analysis were conducted on *ex vivo* pig brains. Results showed a significant 4.8-fold increase in MR contrast enhancement volume in FUS-targeted regions compared to non-targeted regions. FUS sonication enhanced aPD-L1 delivery by an average of 2.1-fold, according to fluorescence imaging. *In vivo* MRI and *ex vivo* staining found the procedure did not cause significant acute tissue damage. These findings demonstrate that FUS-BBBO offers a noninvasive, localized, and safe delivery approach for ICIs in a large animal model, showcasing its potential for clinical translation.

Session 3pCA

Computational Acoustics and Physical Acoustics: Innovations in Computational Acoustics II

Danielle Moreau, Cochair
UNSW Sydney, Sydney, 2015, Australia

Steffen Marburg, Cochair
Technical University of Munich, Boltzmannstr. 15, Garching 87548, Germany

Contributed Papers

1:00

3pCA1. Hardware accelerated ray-tracing in ocean acoustics. David J. Battle (Mission Systems Pty Ltd., Unit 1, 79-85 Mars Rd., Ln. Cove West, New South Wales 2066, Australia, david.battle@missionsystems.com.au) and Vsevolod Vlaskine (Mission Systems Pty Ltd, Ln. Cove West, New South Wales, Australia)

Due to the expense and complexity of real-world data collection, machine learning applications in ocean acoustics demand ever-faster and more realistic models for training. In this talk, I discuss some recent GPU technologies which support hardware accelerated ray tracing and how they can be exploited to boost the throughput of certain types of calculations. Firstly, I describe RTX-Prop, which is a large-scale range-dependent Gaussian beam model which benefits from hardware-acceleration of bounding volume hierarchy searches and geometry intersection calculations. Designed to ingest oceanographic forecasts from the Royal Australian Navy's Bluelink system, RTX-Prop has been developed as an embedded model to support tactical decisions by autonomous underwater vehicles. In the second application, I discuss hardware acceleration in the context of high-frequency sonar simulation. By synthesising fine-scale environmental detail, we have previously shown that simulated imagery can be almost indestinguishable from real data. Simulated sidescan and SAS imagery from our S4 model has recently become the basis of a machine-learning pipeline for training detectors and classifiers for naval mine countermeasures. [Work supported by the Trusted Autonomous Systems Defence Coorperative Research Centre.]

1:20

3pCA2. An application of dereverberation and multiplicative beamforming techniques for imaging noise sources generated by boat engines. Rohit Singh (Mech., IIT Kanpur, G-201, Hall 7, Kanpur, Uttar Pradesh 208016, India, rohitme0074@gmail.com) and Akhilesh Mimani (Mech. Eng., Indian Inst. of Technol. Kanpur, Kanpur, Uttar Pradesh, India)

This paper demonstrates the application of a dereverberation algorithm for localizing noise sources emitted from two-stroke boat engines in three dimensions domain. The engines considered have a maximum RPM of 1500 and 1200 cc. It is important to note that these engines are operated with both petrol and CNG. The investigation primarily focuses on the noise generated under various loading conditions and aims to differentiate the noise characteristics between petrol and CNG engine operations. The dereverberation algorithm is applied to enhance the accuracy of noise source localization in this study, it involves filtering an approximate optimal window width to retain data related to the direct field while removing reflections through multiplication with an appropriate Hanning window. The analysis showed that the dereverberation algorithm effectively improves the imaging source map compared to conventional beamforming maps in the 3-D domain. To compute the beamforming output, data has been recorded from five different sides of the engine, and the multiplicative beamforming technique is employed. The findings highlight the significance of dereverberation and multiplicative beamforming techniques in accurately localizing noise sources in boat engine applications.

1:40

3pCA3. A comparison of background noise reduction algorithms for improving acoustic beamforming output. Rohit Singh (Mech., IIT Kanpur, G-201, Hall 7, Kanpur, Uttar Pradesh 208016, India, rohitme0074@gmail.com) and Akhilesh Mimani (Mech., IIT Kanpur, Kanpur, Uttar Pradesh, India)

This work presents a comparison of different background noise reduction algorithms available in literature at a range of signal-to-noise ratios (SNRs) for improving the conventional beamforming (CB) source maps. The algorithms include the classical background noise removal (BNR), eigenvalue identification organization and subtraction (EIOS), subspace-based background subtraction (SBS), and ensemble empirical mode decomposition (EEMD), amongst others. To assess the performance of different algorithms, three test-cases were considered: (1) localizing a loudspeaker placed in an anechoic environment in the presence of background white-noise, (2) diagnosing machine-tool noise source(s) during machining, i.e., when the workpiece and tool piece are in contact and (3) localizing airfoil trailing-edge (TE) noise sources in the presence of background tunnel noise. The analysis showed that the suitability of an algorithm depends upon the given situation – for the loudspeaker source, all algorithms deliver comparable results while for machine-tool application, the relatively simple BNR algorithm delivered the desired improvements in CB maps. For the experimental airfoil TE noise test-case, the EIOS and EEMD algorithms demonstrated substantial enhancements as compared to other methods. This investigation, therefore, suggests that for a given application, it is desirable to customize the background noise reduction algorithm to obtain optimal results.

2:00

3pCA4. Numerical simulations of drill-string responses for downhole acoustic telemetry. Ao-Song Zhao (Inst. of Acoust., Chinese Acad. of Sci., Beijing, China), Xiao He (Inst. of Acoust., Chinese Acad. of Sci., 21 North 4th Ring Rd. West, Beijing 100190, China, hex@mail.ioa.ac.cn), and Hao Chen (Inst. of Acoust., Chinese Acad. of Sci., Beijing, China)

In petroleum engineering, one major challenge for logging while drilling is that the transmission rate of downhole telemetry is too low for real-time evaluations. Downhole acoustic telemetry (DAT) uses acoustic guided waves in a drill string with periodic structures to transmit measurement data, expected to break through low-speed telemetry limitations. However, the lack of effective modeling methods makes the channel response difficult to predict, severely restricting equipment development and field applications. We propose a novel modeling approach to study the responses of the DAT channel. Excitation and propagation mechanisms of the mode waves

3p WED. PM

in the DAT channel are investigated. The 2-D modeling issue is approximated to the 1-D plane-wave propagation along the borehole, considering transmission, reflection, and interconversion of the drill-string and fluid waves. The coefficients of each wave component and the full-wave channel function are derived from the transfer matrix. The efficiency and accuracy of the proposed means are validated by comparison with the finite difference. Numerical results show that the fluid-solid coupling leads to a generalized multipath effect in the DAT channel, making the acoustic responses exhibit discrete bandgaps and nonlinear phase distortion. This study provides a basis for frequency band selection and design for DAT systems.

2:20–2:40 Break

2:40

3pCA5. Construction site environmental pollution management—Initiatives and innovation in South East Asia. Charles-Etienne Lamort (Acoem, 200 Chemin des Ormeaux, Limonest 69760, France, charles-etienne.lamort@acoem.com)

The management of pollution, and in particular of the impact of noise and vibrations, is very important for construction activities, even more so when it comes to urban construction sites. Indeed, some operations have a very significant emissive potential and it is appropriate, for the proper conduct of the activities, to adopt a certain strategy in order to control the impacts and thus ensure the good acceptance of the construction by the neighborhood. Environmental monitoring is the key to achieving this dual objective: prevent pollution and communicate. This article shows some initiatives adopted in South East Asia and illustrates how recent innovations using AI support work supervisors.

3:00

3pCA6. Applications of intrinsic coordinates for simulation of thin shocks in relaxing media. William A. Willis (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX 78713-8029, william.willis@utexas.edu), John M. Cormack (Div. of Cardiology, Dept. of Medicine, Univ. of Pittsburgh, Pittsburgh, PA), and Mark F. Hamilton (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

The augmented Burgers equation that describes nonlinear propagation in a relaxing fluid can be expressed using intrinsic coordinates [Hammerton and Crighton, *JFM* **252**, 585 (1993)]. A multivalued pressure waveform in physical coordinates can be represented as single-valued with intrinsic coordinates. Shocks can then be inserted into the multivalued pressure waveform using the equal area rule from weak shock theory to obtain a single-valued waveform solution, thus avoiding the high computational cost associated with discretizing thin shocks with conventional algorithms. By first solving the evolution equation in intrinsic coordinates, and then using the solution as the input to a conventional algorithm based on the augmented Burgers equation, a two-stage approach can be developed that achieves both high accuracy and significantly reduced computational cost when compared to solving the Burgers equation with conventional numerical algorithms alone. Approaches based on intrinsic coordinates are applied to realistic problems in relaxing media, which include air and seawater, to allow for comparisons of computational efficiency and accuracy. [WAW is supported by the ARL:UT Chester M. McKinney Graduate Fellowship in Acoustics.]

3:20

3pCA7. Estimation methodology for three-dimensional acoustic modeling: Predicting surf noise soundscapes in coastal urban environments. Simon D. Johnson (ADP Consulting, Level 16 15 Adelaide St., Brisbane, Queensland 4000, Australia, s.johnson@adpconsulting.com.au)

This research develops an estimation methodology for coastal surf noise soundscapes for implementation in three-dimensional acoustic modeling to improve communication of acoustic design and for assessment with sustainability, wellness and acoustic standards. As sustainability standards include more parameters relating to soundscapes, it is becoming increasing important to develop methods of communicating acoustics to designers in the built environment. Three-dimensional acoustic modeling provides a graphical tool for designers to engage with and understand the impacts multidisciplinary engineering and design decisions have on the soundscape of different parts of a project. As a nation with many coastal urban environments, the view, scents and sounds are all marketing and wellness points for residential and commercial projects. The methodology includes consideration of typical software-agnostic three-dimensional modeling inputs, capabilities and graphical outputs as well as the primary variables that influence measurement and prediction of coastal surf noise in relation to the constant environmental noise. Coast surf noise measurements have been collected and undertaken as part of this work and the data within the method may be updated for most locally measured data. This research aims to enhance the demonstration of soundscapes, facilitating more informed decision-making fostering sustainable development in acoustics, engineering and design.

Session 3pEA

Engineering Acoustics: Transducer Design and Evaluation

Wonkyu Moon, Cochair
Pohang University of Science and Technology, 39, Jigok-ro, Nam-gu, Pohang-si 37666, South Korea

Stefanie Gutschmidt, Cochair
University of Canterbury, Mechanical Engineering Dept., Christchurch 8140, New Zealand

Contributed Papers

1:40

3pEA1. Machine learning based flexible piezoelectric sensor. Keon Ja E. Lee (KAIST, Kangnamgu Seonleungro 221, Seoul 06276, France, keonlee@kaist.ac.kr)

This seminar introduces flexible inorganic piezoelectric membrane that can detect the minute vibration of membrane for self-powered acoustic sensor and blood pressure monitor. Herein, we reported a machine learning-based acoustic sensor by mimicking the basilar membrane of human cochlear. Highly sensitive self-powered flexible piezoelectric acoustic sensor with a multi-resonant frequency band was employed for voice recognition. Convolutional Neural Network (CNN) were utilized for speaker recognition, resulted in a 97.5% speaker recognition rate with the 75% reduction of error rate compared to that of the reference MEMS microphone.

2:00

3pEA2. A high-fidelity MEMS microphone with a polymer membrane that can detect infra-sounds. Junsoo Kim, Woongji Kim (Mech. Eng., Pohang Univ. of Sci. and Technol., Pohang-si, South Korea), Siyoung Lee, Kilwon Cho (Chemical Eng., Pohang Univ. of Sci. and Technol., Pohang-si, South Korea), and Wonkyu Moon (Mech. Eng., Pohang Univ. of Sci. and Technol., 39, Jigok-ro, Nam-gu, Pohang-si, Gyeongsangbuk-do 37666, South Korea, wkmoon@postech.ac.kr)

This study proposes a polymer MEMS microphone that operates through electromechanical amplitude modulation. We have developed a highly optimized MEMS structure with a flexible polymer material, resulting in a miniaturized and flexible device with excellent acoustic performance. The stiffness and damping characteristics of the polymer diaphragm are analyzed and it shows that the polymer material would be suitable for microphone applications. An equivalent circuit model is also developed for design purposes, so that the device could be properly designed using it and fabricated by in-house polymer thin film processes. The fabricated device is tested using a DC bias scheme to reach its signal-to-noise ratio (SNR) of 65 dBA within a bandwidth of 10 to 10,000 Hz, which is comparable to existing high-end commercial silicon MEMS microphones. We would then extend the frequency bands of linear detection down to the infrasound range by using electromechanical amplitude modulation. This operation scheme can also provide the microphone with no need for metal packaging. Its feasibility is confirmed theoretically by numerical simulations including electromechanical amplitude modulation and experimentally by implementing it at the PCB level. In addition, this approach enables the sensor stronger to the environmental noise so that metal shielding could be omitted.

2:20

3pEA3. Improved spatial-temporal sound perception using active MEMS. Stefanie Gutschmidt (Univ. of Canterbury, Christchurch, New Zealand, stefanie.gutschmidt@canterbury.ac.nz), Alexandra McKendry, Christopher Cameron, and Seigan Hayashi (Univ. of Canterbury, Christchurch, New Zealand)

Current microphone technology is based on a transducer concept that is the same for all end user groups, namely, passive, linear. While most of the technology-related challenges have been overcome by sophisticated post-processing algorithms, some of the current limitations of sound detection technology, especially as recognized by groups dealing with hearing impairments and losses, should be addressed at the transducer stage. In this work we present the dynamic behaviour and performance of an active, nonlinear MEMS sensor subject to acoustic stimuli and noise. Our work includes the presentation of a reduced-order model of the composite MEMS structure, relevant dynamic analyses including bifurcation and stability diagrams, as well as experimental investigations. Simulations and experimental results demonstrate cochlear-like amplification properties, meaning that soft signals are amplified significantly different from louder signals. Signals which initially are imbedded in a noise floor can be amplified without also amplifying the noise and thus providing another way of detecting sound patterns.

2:40

3pEA4. Acoustics of a pressurized air-filled pipe open at one-end at varying depths. Carl Howard (School of Elec. and Mech. Eng., The Univ. of Adelaide, Adelaide, South Australia 5005, Australia, carl.howard@adelaide.edu.au), Richard Craig (School of Elec. and Mech. Eng., The Univ. of Adelaide, Adelaide, South Australia, Australia), and James Forrest (Defence Sci. and Technol. Group, Fishermans Bend, Victoria, Australia)

A rigid pipe with a loudspeaker at one end and open at the other end, was submerged in a water tank at various depths. Microphones were placed along the length of the pipe to measure the acoustic response of the duct due to the acoustic excitation from an internal loudspeaker. Compressed air was injected into the pipe to equalise the internal pressure of the pipe caused by the water pressure acting on the open end of the pipe. The results were that the acoustic resonance frequencies of the duct did not alter with depth.

3:00–3:20 Break

3:20

3pEA5. Physical-model-based reconstruction of three-dimensional sound field from multi-directional measurement by parallel phase-shift interferometry. Haruka Nozawa (Dept. of Intermedia Art and Sci., Waseda Univ., 3–4–1 Ohkubo, Shinjuku-ku, Tokyo 169–8555, Japan, redmouse995@akane.waseda.jp), Mayuko Imanishi, Yasuhiro Oikawa (Dept. of Intermedia Art and Sci., Waseda Univ., Shinjuku-ku, Japan), and Keji Ishikawa (NTT Commun. Sci. Labs., Atsugi, Kanagawa, Japan)

The high-spatial-resolution nature of optical non-invasive measurement and the reconstruction of three-dimensional sound fields are important for evaluating acoustic transducers. The computed tomography (CT) method has been used to reconstruct sound fields from optical measurement results because of its contactless feature. However, since the CT method does not assume any physical property of sound, it may not be suitable for the reconstruction of sound fields. In contrast, a physical-model-based method is able to reconstruct sound fields more accurately by considering the physics of sound. In this study, we propose a scan-free and physical-model-based reconstruction of a three-dimensional sound field. A sound field is multi-directionally measured using parallel phase-shifting interferometry and a high-speed polarization camera by rotating an acoustic transducer. A physical-model-based method is applied to reconstruct the sound field from the recorded optical interferograms. We also discussed the effectiveness of the proposed method to evaluate acoustic transducers.

3:40

3pEA6. An improved lumped parameter model of the single free-flooded ring transducer using Helmholtz–Kirchhoff integral solution. Junsu Lee (Pohang Univ. of Sci. and Technol., 39, Jigok-ro, Nam-gu, Pohang-si, Gyeongsangbuk-do 37666, South Korea, junsu.lee@postech.ac.kr), Kyounghun Been, and Wonkyu Moon (Pohang Univ. of Sci. and Technol., Pohang-si, Gyeongsangbuk-do, South Korea)

The Helmholtz-Kirchhoff integral (HKI) may be considered as the boundary element method (BEM) used to calculate the acoustic pressure at any position in the 3D-radiation problem from a vibrating surface. This method provides valuable insights into the radiation characteristics of vibrating structures, which may offer significant information and tools to underwater transducer design engineers during the early stages of development. It may save considerable computational costs in the design processes for the underwater transducer such as Hull Mount Sonar. This study illustrates usefulness of the HKI-based approach by developing a simple model of the Free Flooded Ring (FFR) transducer using a piezoelectric ring model and the HKI solution on the surface acoustic pressure of the ring through their physical conditions on their connected interfaces. The FFR transducer is widely utilized as a low-frequency acoustic source in underwater environments due to its broad operating frequency bandwidth and relatively small size. The developed approach aims to construct a precise model that considers the sound-structure interactions between the piezoelectric ring and the acoustic medium around it. To validate the accuracy of the proposed method, the acoustic pressure and electrical admittance are compared with those obtained through finite element method (FEM) simulations.

4:00

3pEA7. Comparative analysis of receiving characteristics of all flextensional transducer classes. Gihyeon Kim (School of Mech. Eng., Kyungpook National Univ., 80 Daehakro, Bukgu, Daegu 41566, South Korea, f12gto5800@naver.com), Wenbo Wang, Donghyun Kim, Hayeong Shim, and Yongrae Roh (School of Mech. Eng., Kyungpook National Univ., Daegu, Korea (the Republic of))

Flextensional transducers have been widely used as low-frequency projectors, and their characteristics can be exploited to develop hydrophones with better sensitivity and wider receiving bandwidth than other existing hydrophones in the low-frequency band. In this study, we compared and analyzed the receiving characteristics of different classes of flextensional transducers to identify the most suitable class for wideband low-frequency hydrophones, particularly for broad applications. To begin, we established an initial model with the same peak receiving-voltage-sensitivity (RVS) frequency for each class of flextensional transducers. Through a thorough analysis of various structural variables, we examined their impact on the receiving characteristics of the transducers. Based on these findings, we selected the design variables with the most significant influence on the bandwidth for each class. Subsequently, we determined the optimal combination of these variables to maximize the receiving bandwidth while maintaining RVS at or above 1 kHz. Comparing the performance of the best-fit models across all classes, we found that class IV flextensional transducers exhibit the widest low-frequency receiving bandwidth, closely followed by the convex class I. The −3 dB receiving bandwidth of the class IV transducer amounts to 87.6% of the peak RVS frequency, while the convex class I transducer achieves 67.1%.

Session 3pNSa

Noise and Architectural Acoustics: Soundscapes

Sophie Gleeson, Cochair
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Contributed Papers

1:00

3pNSa1. Application of soundscape methodologies in an acoustic consulting context. Sophie Gleeson (Acoust., Arup, Wurundjeri Woiwurrung Country, Sky Park One Melbourne Quarter, 699 Collins St., Docklands, Victoria 3008, Australia, Sophie.Gleeson@arup.com) and Mitchell J. Allen (Acoust., Arup, Sydney, New South Wales, Australia)

Understanding environmental sound from subjective perspectives is important for comprehending sound qualities beyond physical parameters. Soundscape approaches are promoted as valuable frameworks for understanding subjective experiences of environmental sound. However, the application of such approaches in acoustic consulting environments is limited and the demonstrated value is unclear. This study investigates the applicability of soundscape methodologies in a corporate consulting context. In this paper we summarise available soundscape methodologies and their known challenges, and test the application of select methods, including sound mapping and soundscape descriptors, in different acoustic consulting settings at Arup. Through practical application, we explore the contexts in which soundscape approaches may be used to document, assess and describe sound environments. We review barriers and limitations experienced through application, such as training, project constraints and the need for project-specific methodology adaptation, and discuss values that emerged, including an enriched understanding of sound qualities not otherwise captured through conventional acoustic approaches. We position this research as a step towards developing a soundscape toolkit for acoustic consultants and argue that the incorporation of a soundscape perspective in acoustic consulting is necessary to expand the ways in which we understand environmental sound and its effects on those within it.

1:20

3pNSa2. Sonic environment mapping in Tunjungan road Surabaya with soundwalk method. Muhammad A. Asyraf (Dept. of Eng. Phys., Institut Teknologi Sepuluh Nopember, Surabaya 60111, Indonesia, maasyraf.edu@gmail.com), Dhany Arifianto, and Bina Artika Putri (Eng. Phys., Institut Teknologi Sepuluh Nopember, Surabaya, Indonesia)

The soundscape analysis, based on the soundwalk method, was carried out at Tunjungan Street, which is one of the historical areas and a place of traffic. A soundwalk is one method that is used to get the impression of a soundscape. The parameters used in the assessment are acoustic parameters and semantic parameters. Acoustic parameters consist of loudness level, dynamic, and IACC. Semantic parameters are subjective judgments using 16 question items that were given by the researcher to the respondent in the questionnaire. The collection of data was done by the respondent in Tunjungan Street, which was divided into five sections or five points, and each point was measured at three times: morning, daytime, and nighttime. From the results of the study, several conclusions were drawn, namely that the morning has the largest loudness level 81 dBA. The highest dynamic span about 15 dBA and the lowest IACC value is 0.51. When daytime, it has larger loudness level and dynamic value, while at the nighttime it has the largest loudness level and IACC values in particular area only. This finding suggests that there are some part that offers comfort acoustically.

1:40

3pNSa3. A research on cognition of Soundscape in daily life of university students living in Tokyo. Takeshi Akita (Dept. of Sci. and Technol. for Future Life, Tokyo Denki Univ., 5 Senju-Asahi-cho Adachi-ku, Tokyo 1208551, Japan, akita@cck.dendai.ac.jp)

In the present research, interest to soundscape that is cognized by a person in everyday life is investigated. Ordinarily, a person perceives their environment by using his sense. Using hearing sense, he perceives acoustic environment. At the same time, he perceives light environment by vision, thermal environment by sensation of warmth, and air quality by smell. In such a case, knowing how important acoustic environment is for people among various environment is seemed to be useful for creating good soundscape. To reveal this point, one investigation using a questionnaire of free description is carried out. Forty-five university students who were all twenties and lived in Tokyo participated in the investigation. They were instructed to write down perceived environment (acoustic, light, thermal, or air quality environment) and its comfortableness in their daily life at home, office, or part time job. Results showed that 7.3 environments per person were answered by questionee, and 41.9% of answer were about acoustic environment. They also showed that 49.3% of answers about acoustic environment said uncomfortable, but that answered acoustic environment and its comfortableness varied widely. It is supposed that students in Tokyo relatively have high level interest in soundscape among various environments.

2:00

3pNSa4. The soundscape method and eco-acoustic indicators in the analysis of the acoustic environment of the mountain national parks in Poland and Sweden. Dorota Mlynarczyk (Dept. of Mech. and Vobroacoustics, AGH Univ. of Krakow, Krakow, Poland) and Jerzy Wiciak (Dept. of Mech. and Vobroacoustics, AGH Univ. of Krakow, Krakow, Poland, wiciak@agh.edu.pl)

Soundscape is one of the essential environmental elements affecting people's experience in nature spaces. In the complex mechanism of landscape and soundscape perception, the acoustic characteristics of the environment are the primary influencing factor. The aim of the paper is the analysis of soundscape of two mountain national parks—Tatra National Park (TNP) in Poland and Sarek National Park (SNP) in Sweden. The first park is very frequently visited by tourists (around 4 million tourists a year) while the second park has an attendance of around a few thousand tourists. The paper presents the results of the analysis of acoustic measurements and ambisonic recordings made in the Koscieliska Valley (TNP) and the Rapa Valey (SNP, Swedish: Rapadalen) using classical method and the soundscape method. In addition, psychoacoustic parameters and soundecology indicators such as loudness, sharpness or roughness, ACI (acoustic complexity index), NDSI (normalized difference soundscape index), BIO (bioacoustic index), ADI (acoustic diversity index), and AEI (acoustic evenness index) were calculated.

2:20

3pNSa5. The puzzling soundmark of a cultural and tourism city: The case of Yogyakarta. Christina E. Mediastika (Dept. of Architecture, Universitas Ciputra Surabaya, CitraLand, CBD Boulevard, Made, Sambikerep, Surabaya, Jawa Timur 60219, Indonesia, eviutami@ciputra.ac.id), Anugrah Sabdono Sudarsono (Kelompok Keahlian Fisika Bangunan, Institut Teknologi Bandung, Bandung, Indonesia), Sentagi Sesotya Utami (Phys. Eng., Universitas Gadjah Mada, Yogyakarta, Indonesia), Yusuf Ariyanto (Dept. of Architecture, Universitas Ciputra Surabaya, Surabaya, Indonesia), Teguh Setiawan (Yogyakarta Tourism Office, Yogyakarta, Indonesia), and Ressy Jaya Yanti (Phys. Eng., Universitas Gadjah Mada, Yogyakarta, Indonesia)

Yogyakarta is a city rich in culture and heritage, making it the second tourist destination in Indonesia after Bali. However, unlike Bali, Yogyakarta losses its intangible uniqueness quicker. This paper explores societies' and stakeholders' perceptions of Yogyakarta's soundscape, particularly the vanishing soundmark. Quantitative and qualitative approaches were employed to collect data through questionnaires and focus group discussions. Quantitative data were processed using a modest statistical method to show trends, and qualitative data were processed using the Colaizzi protocol. Quantitatively, respondents stated that the sounds of gamelan (a set of traditional musical instruments) and andong (horse-drawn carriages) are two soundmarks of Yogyakarta. They also recognize a third soundmark, namely the mystical sound of a marching band heard at certain times, whose origins are still debated. It truly represents the term "intangible." Through the FGD, the discussants revealed that determining the iconic sound is intricate. It needs consent on what "icon" means because different areas have unique sounds. However, they agreed that Maliboro is still the most iconic place but has lost its soundmark due to traffic noise. The challenging part in developing soundscapes and soundmark is the varying perceptions between residents, visitors, and generations about the pleasant and memorable sound.

2:40

3pNSa6. Urban Soundscape: White-vented Myna choruses on Zhongqing road. ChihHsuan Chen (Architecture, National Taiwan Univ. of Sci. and Technol., No. 43, Section 4, Keelung Rd., Taipei City 106, Taiwan, M11013037@mail.ntust.edu.tw) and Lucky Tsaih (Architecture, National Taiwan Univ. of Sci. and Technol., Taipei, Taiwan)

Urbanization has impacted both humans and birds, and certain bird species, like the white-vented myna choruses on Zhongqing Road, have adapted to human-dominated environments. A study in April 2023 utilized cameras and sound level meters to document this phenomenon. Two phases were conducted: the first phase recorded white-vented mynas' departure and return times at dawn and dusk, along with the sound pressure level of birdsong. The second phase measured indoor sound pressure levels during peak birdsong at nearby buildings. Additionally, a questionnaire assessed people's perception of the bird choruses. Results indicated that birdsong at dawn had a lower sound pressure level (68.8 dBA) compared to dusk (75.4 dBA), but it persisted longer. The dominant frequency was around 3150 Hz. Three out of four indoor sites exceeded the regulatory standard of 60 dBA. The questionnaire showed that over 75% of nearby residents disliked the bird choruses, while 18% of passersby found them interesting and potentially attractive for visitation. In conclusion, urban bird choruses can impact indoor acoustic environments and lead to negative perceptions among residents. This study aimed to provide quantitative data and insights into residents' listening impressions of bird choruses on Zhongqing Road, offering valuable reference data for stakeholders.

Session 3pNSb

Noise, Architectural Acoustics and Physical Acoustics: Forensic Acoustics: What's that Noise?

Dana Hougland, Chair
Shen Milsom & Wilke, LLC, 1801 Wewatta, Floor 11, Denver, CO 80202

Chair's Introduction—3:20

Invited Paper

3:20

3pNSb1. What's *that* noise—Found it! Sometimes. John Baldassano (Ostergaard Acoust. Assoc., 1480 US 9, Woodbridge, NJ 07095, jbaldassano@acousticalconsultant.com) and Joseph Keefe (Ostergaard Acoust. Assoc., Woodbridge, NJ)

A few case studies involving investigation of mystery noise sources are presented. These address techniques used to identify the cause of complaints, locate noise sources amidst competing noise, and compare predicted sound levels to measured sound levels. The presentation will discuss mysteries solved, our approach to noise control recommendations, and lessons learned. Interesting cases include investigation of a tone generated by an electrical power transmission line pole, office noise control involving malfunctioning HVAC systems, and noise source identification within a quiet multi-family residence.

Contributed Papers

3:40

3pNSb2. Tackling the strange wind noise on a commercial building—Case study. Harvey Law (Mech. Eng. & Mater. Eng., Monash Univ., Australia, Bldg. 3 (Rear), 621 Whitehorse Rd., Mitcham, Victoria 3132, Australia, harvey.law@megasorber.com)

A mysterious noise was reported in a commercial building. This commercial building consists of multiple levels of car parks with one single level of office space at the top. This top-level office area is home to Renault Australia, Mulgrave, Victoria. The office staff at the Renault reported a dreadful "howling" noise around the office. It sounded like a huge truck thundering past the office, especially on windy days. Specialist engineers, acoustic and wind engineers, were summoned to investigate the root cause of this noise. The initial investigation revealed that on windy days, the exterior aluminium Z Perlin profile cladding created air turbulences, which resulted in this unwanted noise. After conducting tests in the Monash University wind tunnel, various recommendations were suggested and trialled. For example, removing some purlins to reduce the air-flow turbulence, attaching fibre cement sheets to seal the purlins, vibration damping treatment of the Z purling cladding, and absorptive infill on the metal purlins. This case study presents the analysis of the noise, the results of the various tests and the successful practical resolution.

4:00

3pNSb3. Noise measurement, litigation, positioning and acoustic remediation of a heat pump installed at INAIL headquarters in the city of Salerno (ITA). augusto papa (UOT CVR Napoli, INAIL, viale di Augusto, 9, Napoli 80125, Italy, a.papa@inail.it) and Raffaele Mariconte (Dipartimento Innovazioni Tecnologiche e Sicurezza Degli Impianti, Prodotti Ed Insediamenti Antropic, Inail, Roma, Italy)

The study concerns a particular case of an air conditioning system subject to a legal dispute with the residents of the building of the INAIL headquarters in the city of Salerno. The system moved from the terrace to the ground, involved a complex forecast assessment of acoustic impact with subsequent adequate acoustic remediation.

Session 3pPA

Physical Acoustics and Engineering Acoustics: Artificial Intelligence for Metamaterials

Feruza Amirkulova, Cochair
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David Powell, Cochair
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Chengzhi Shi, Cochair
Georgia Inst. of Technology, Atlanta, GA 30332

Chair's Introduction—12:55

Invited Paper

1:00

3pPA1. A wide field-of-hearing metalens for aberration-free sound capture. Dongwoo Lee (Mech. Eng., Pohang Univ. of Sci. and Technol. (POSTECH), 77, Cheongam-ro, Nam-gu, Pohang-si 37673, Gyeongsangbuk-do, Republic of Korea, dwlee93@postech.ac.kr), Beomseok Oh, and Junsuk Rho (Mech. Eng., Pohang Univ. of Sci. and Technol. (POSTECH), Pohang, South Korea)

Metalenses, as a designer tool for manipulating waves, has demonstrated remarkable capabilities thus far. However, their performance is limited to normal incidence and they exhibit significant sensitivity to oblique incidence. This sensitivity often leads to a decrease in focusing efficiency due to coma aberration, posing a considerable challenge. Here, we present a novel solution in the form of a wide field-of-hearing (FOH) metalens, specifically designed for capturing and focusing sound without aberration. To overcome the limitations of conventional metalenses, we propose a hybridized metalens that combines Helmholtz and zigzag building blocks. By incorporating both complete phase modulation and unitary transmission, this hybrid metalens achieves superior performance. We conduct experimental validations to demonstrate the proof-of-concept sound reception, thereby introducing the concept of wide FOH monitoring systems. The rational design of this metalens holds great versatility and can be implemented across various platforms. Its applications potentially span diverse fields such as energy harvesting, monitoring, sensing, imaging, and communication in auditory and submerged environments.

Contributed Papers

1:20

3pPA2. Deep reinforcement learning for optimal sound absorbing structures design. Semere B. Gebrekidan (Dept. of Eng. Phys. and Computation, Tech. Univ. of Munich, Boltzmannstraße 15, Garching b. München 85748, Germany, semere.gebrekidan@tum.de) and Steffen Marburg (Dept. of Eng. Phys. and Computation, Tech. Univ. of Munich, Garching, Germany)

Deep learning algorithms have demonstrated a tremendous success in designing structures that surpass human capabilities. Based on the recent achievements of deep reinforcement learning in surpassing human capabilities, this paper focuses on implementing these algorithms to design optimal configurations of solid and porous materials that achieve a broadband absorption within the frequency range of 300 Hz to 3000 Hz. We employ model-free approaches, specifically deep Q-learning, double deep Q-learning, and dueling deep Q-learning algorithms, to predict material configurations that optimize absorption without requiring expertise knowledge. From a $2^{30\times30}$ different material combinations, the deep reinforcement algorithms learn to predict configurations that yield optimal absorption in few hundred steps. We discuss further the superior performance of a dueling deep learning algorithm compared to the other two deep learning approaches and a heuristic approach, such as genetic algorithm. The proposed model-free algorithms enable the prediction of absorption performance for any material configurations without the need for expertise.

1:40

3pPA3. The wave equation as a differentiable-programmable computer for the design of acoustic metamaterials. Tristan A. Shah (Comput. Eng., San Jose State Univ., 98 North First St., San Jose, CA 95113, tristan.shah@sjsu.edu), Stas Tiomkin (Comput. Eng., San Jose State Univ., San Jose, CA), and Feruza Amirkulova (Mech. Eng., San Jose State Univ., San José, CA 95192-0087, CA)

Designing new metamaterials enables fine-grain control of wave dynamics. Applications of these materials range from wave steering devices for seismic wave manipulation, to "super-focusing" devices for high-resolution ultrasound imaging and surgery. Control of systems governed by partial differential equations is an inherently hard problem. Specifically, control of wave dynamics is a challenging problem, because of the additional physical constraints and intrinsic properties of wave phenomena such as attenuation, reflection, and scattering. In this work, we propose a novel method for transient control of wave equations with Model Predictive Control. The proposed model incorporates the essential physical properties observed in the original dynamics directly into its lower dimensional latent space. We show

that our model is capable of using its latent dynamics to predict scalar integral quantities over a variable length timespan. Moreover, it is fully interpretable by the corresponding physical quantities, which allows it to guarantee solution properties. We demonstrate this method by solving important cases of wave equation, which have not been solved before, such as control of transient waves. We make our code publicly available.

2:00

3pPA4. Reconfigurable labyrinth structures as low frequency sound absorbers. Kian-Meng Lim (National Univ. of Singapore, 9 Eng. Dr. 1, Singapore 117575, Singapore, limkm@nus.edu.sg), Yuelin Chen (National Univ. of Singapore, Singapore, Singapore), and Heow Pueh Lee (Mech. Eng., National Univ. of Singapore, Singapore, Singapore)

In this work, we designed a metamaterial labyrinth structure as sound absorber for low frequency. The labyrinth structure is made by stacking disks with holes or openings at its front and back surfaces, and these openings are linked by air channels within the disks. Five disks with different opening locations were designed and fabricated using 3D printing. The disks were stacked and rotated with respect to each other, connecting different inlets and outlets, to form various patterns of labyrinth. Using a two-microphone impedance tube, the absorption coefficient of these stacks, ranging from one to five disks, was measured. The results show that a stacked structure with more disks enhances the absorption at low frequencies. As each disk was designed with twelve possible rotational positions, a stack of three disks will give more than 8000 combinations. It is not practical to conduct experiments to measure the absorption coefficient of all possible combinations. Hence, a neural network was used to fit and predict the absorption. Using the fitted neural network model, we would be able to predict the combinations to give best absorption and target frequency band. [Work supported by Singapore MOE Tier 1 Grant A-0009119-00-00.]

2:20–2:40 Break

2:40

3pPA5. Manipulating the phase and group delay of a unit cell in metasurface via impedance engineering. Dingcheng Yang (School of Eng. & Technol., Univ. of New South Wales Canberra, Northcott Dr., Campbell, Australian Capital Territory 2612, Australia, dingcheng.yang@adfa.edu.au), Yankei Chiang, and David Powell (Univ. of New South Wales Canberra, Canberra, Australian Capital Territory, Australia)

The emergence of metasurfaces that tailor the phase via carefully controlling each unit cell brings promising applications including wavefront manipulation, acoustic lenses and levitations. Although excellent phase engineering has been reported in the literature, most metasurface designs suffer from narrow operating bandwidth. To achieve consistent performance over a broad bandwidth, it is necessary to engineer the group delay, in addition to the phase delay, however, few studies have considered this. The majority of them realized specific group delay by true time delay via tuning the depth of a cavity or peculiar topology-optimized structures. No systematic method of unit cell design has been proposed which is a significant step to broadband design. In this work, we proposed a unit cell design process incorporated with transfer matrix method and equivalent LC-circuit fitting. The transfer matrix method is a semi-analytical model capable of predicting the phase response if parameter lists are given. The LC-fitting interprets targeted impedance profile of unit cell by the equivalent inductance and capacitance, assisting the parametric sweeps for geometry design. A numerical study presented showing the mapping of LC values with two physical parameters, leading to control of phase and group delay simultaneously.

3:00

3pPA6. Slow vibration via acoustoelastic coupling: Quasi-bound states in the continuum. Dongwoo Lee (Mech. Eng., Pohang Univ. of Sci. and Technol. (POSTECH), 77, Cheongam-ro, Nam-gu, Pohang-si, Gyeongsang-buk-do, Republic of Korea, Pohang 37673, Korea (the Republic of), dwlee93@postech.ac.kr), Jeonghoon Park, and Junsuk Rho (Mech. Eng., Pohang Univ. of Sci. and Technol. (POSTECH), Pohang, South Korea)

Elastic bound states in the continuum (BICs) have recently garnered significant attention due to their remarkably high Q-factor, leading to the decoupling of the confined mode from spectrally coexisting radiational channels. In this presentation, we introduce a novel state capable of generating a slow vibration phenomenon, exhibiting multiphysics analogous to the concept of slow light observed in electromagnetically induced transparency (EIT). This state arises from the interaction of acoustoelastic coupling within a composite structure, featuring two acoustic cavities enclosed in an elastic bar. The proposed design facilitates quasi-BICs with exceptional spatial efficiency localized within a specific area, enabling tuning of the Purcell factor by approximately six orders of magnitude. The findings of such quasi-BICs may expand the BIC family and opens new avenues for applications in diverse fields, including lasing, sensing, screening, and energy storage platforms, where ultrahigh-Q-factor modes coexist with radiative channels.

3:20

3pPA7. Design of volumetric sound metadiffusers using gradient-based optimization. Feruza Amirkulova (Mech. Eng., San Jose State Univ., 1 Washington Sq, San José, CA 95192-0087, feruza.amirkulova@sjsu.edu), Samer Gerges, Bharadawaj Jagannath (Mech. Eng., San Jose State Univ., San José, CA 95192-0087), and Matthew Tran (Mech. Eng., San Jose State Univ., Sa Jose, CA)

Broadband volumetric sound diffusers are designed by using gradient-based optimization (GBO) by adjusting the position and radius of each scatterer in nonuniform configurations. The multiple scattering theory is employed to evaluate the diffusion coefficient by computing the scattered pressure field by nonuniform planar configurations of cylindrical scatterers for a monopole excitation. The analytical formulas for the gradients of the diffusion coefficient with respect to positions and radii of a cluster of cylindrical scatterers are derived which enhance the modeling when integrated with GBO and parallel computing. Sound volume metadiffusers with a high diffusion coefficient are designed by perturbatively rearranging the scatterer configurations. Single frequency and broad-octave-band optimizations are performed to maximize the diffusion coefficient while supplying analytical formulas for its gradients with respect to positions and radii. The GBO is implemented by direct optimization employing Multistart and fmincon solvers with sequential quadratic programming algorithms. The GBO approach is demonstrated providing numerical examples for nonuniform configurations of rigid cylinders embedded in the air environment. The polar plots for scattered pressure are evaluated for a wide range of $^1/_3$ octave bands. The GBO approach can facilitate the automated metamaterial process by combining it with global optimization, generative modeling, and reinforcement learning.

3:40

3pPA8. Bistable origami-inspired acoustics metasurface for dynamic beam scanning. Dinh Hai Le (School of Eng. and Information Technol., The Univ. of New South Wales, 10 Oodgeroo Ave. Franklin, Australian Capital Territory 2913, Australia, hai.le@adfa.edu.au), Felix Kronowetter (Chair of Vibroacoustics of Vehicles and Machines, Tech. Univ. of Munich, Munich, Garching, Germany), Yankei Chiang, Jing Rao (School of Eng. and Information Technol., The Univ. of New South Wales, Campbell, Australian Capital Territory, Australia), Marcus Maeder, Steffen Marburg (Chair of Vibroacoustics of Vehicles and Machines, Tech. Univ. of Munich, Garching, Germany), and David Powell (School of Eng. and Information Technol., The Univ. of New South Wales, Canberra, Australian Capital Territory, Australia)

The synergistic integration of acoustic metasurfaces with the adaptive reconfigurability of origami art has demonstrated significant promise in the dynamic modulation of sound waves. This paper proposes an innovative approach to acoustic metasurfaces that incorporates bistable origami-inspired folding structure and digital coding to enable dynamic reconfiguration of beam scanning. The unit cell of the metasurface is optimized to realize two equilibrium states with 180-degree phase difference that can be coded as binary 0 and 1. In particular, by programming each element of the metasurface with different coding sequences, two symmetric reflected acoustic beams can be manipulated at different angles. The proposed design signifies a noteworthy advancement in the field of acoustic metasurfaces, offering a flexible and versatile solution applicable to acoustic imaging, communication, and sensing applications.

Session 3pPPa

Psychological and Physiological Acoustics: Binaural Listening and Scene Analysis (Poster Session)

Anna C. Diedesch, Chair
Communication Sciences & Disorders, Western Washington University, 516 High St., MS 9171, Bellingham, WA 98225

All posters will be on display and all authors will be at their posters from 1:00 p.m. to 2:20 p.m.

Contributed Papers

3pPPa1. An Ideal-observer standard for evaluating relative cue reliance on multi-talker speech segregation. Jungmee Lee (Commun. Sci and Dis, Univ. of South Florida, 4202 East Fowler Ave., PCD1017, Tampa, FL 33620, jungmeelee@usf.edu), Michael Zandona, and Robert Lutfi (Commun. Sci and Dis, Univ. of South Florida, Tampa, FL)

Modern hearing research identifies the ability of listeners to segregate simultaneous speech streams with a reliance on three major voice cues, fundamental frequency, level, and location. Few of these studies, however, present cues simultaneously as in natural listening, and fewer still consider the relative reliance listeners placed on these cues owing to the cues' different units of measure. In the present study, trial-by-trial analyses were used to isolate listener reliance on the three voice cues presented simultaneously, with the behavior of an ideal observer (Green and Swets, 1966, pp. 151–178) serving as standard for determining their relative reliance. Listeners heard on each trial a pair of randomly selected, simultaneously recordings of naturally spoken sentences. One of the recordings was always from the same talker, a distracter, the other, with equal probability, was from one of two target talkers differing in the three voice cues. The listener's task was to identify the target talker. Among 33 clinically normal-hearing adults only one relied predominantly on voice level, the remaining were split between voice fundamental frequency and location. The results are discussed regarding their implications for the common practice of using target-distracter level as a dependent measure of multi-talker speech segregation.

3pPPa2. Binaural renderers accuracy comparison. Lisa LaFountaine (Belmont Univ., 1900 Belmont Blvd, Nashville, TN 37212, Lisa.LaFountaine@bruins.belmont.edu), Raymond Plasse, and Wesley Bulla (Belmont Univ., Nashville, TN)

This study explored the ability of binaural renderers to accurately reproduce the placement of objects within a three-dimensional soundscape. Previous works have only tested localization on the horizontal plane; whereas, this research expanded on prior methodology by adding vertical targets along the medial and sagittal planes. Two commercially available binaural renderers were compared. Subject task was to map from where they heard the localization percept onto a planar response sheet. Results were consistent with previous horizontal plane research. However, renderer performances in the medial and sagittal domains were considerably less accurate than the horizontal plane. Findings here suggest more work is needed to ensure greater accuracy of localization cues provided by the binaural rendering process.

3pPPa3. Investigating the effect of cochleotopic region on temporal processing using binaural fusion. Prajna BK (Speech & Hearing Sci., UIUC, 901 S 6th St., Champaign, IL 61820, prajnab2@illinois.edu), Justin Aronoff, and Simin Soleimanifar (Speech and Hearing Sci., Univ. of Illinois at Urbana Champaign, Champaign, IL)

Processing of temporal information in a signal may vary across apical, mid and basal regions of the cochlea—there is evidence for this in both cochlear-implant and typical-hearing listeners. This effect may also be task dependent and has not been studied with respect to binaural fusion. Fusion depends on tracking the temporal fluctuations across ears which can be manipulated by varying the interaural correlation (IC) of a signal. To investigate the effect of place of stimulation on fusion, 9 bilateral cochlear implant users were presented with 1000 Hz pulse trains with varying envelope IC. An apical, mid, and basal electrode were separately selected as reference, each of which was then paired with five electrodes in the contralateral ear, spread across the array. Listeners indicated the spatial diffuseness and the number of auditory "images" of the perceived sound by manipulating a visual representation of their perception superimposed on a picture of a head. Preliminary results indicate that, for optimally paired electrodes, fusion increases similarly for the three regions of the cochlea with increasing IC, and when IC is 1, the degree of fusion is comparable across regions. Processing of temporal information for binaural fusion is not significantly different across cochleotopic regions.

3pPPa4. Does sensitivity to interaural correlation changes correlate with binaural fusion performance? Prajna B. K. (Speech & Hearing Sci., UIUC, 901 S 6th St., Champaign, IL 61820, prajnab2@illinois.edu), Simin Soleimanifar (Speech and Hearing Sci., Univ. of Illinois at Urbana Champaign, Champaign, IL), Leslie Bernstein (Depts. of Neurosci. and Surgery, Univ. of Connecticut School of Medicine, Farmington, CT), and Justin Aronoff (Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, Champaign, IL)

The purpose of this study was to assess whether and to what degree sensitivity to changes in interaural correlation (IC) in bilateral cochlear implant (CI) listeners is related to the overall degree of binaural fusion those listeners experience. Compared to typical-hearing listeners, CI users often report perceiving a reduced degree of binaural fusion. Because IC is a strong determiner of perceived fusion, degraded perception of fusion might be coupled with a relative insensitivity to changes in IC. To investigate this, two experiments were conducted. First, discrimination of IC changes was measured

using the method of constant stimuli in which CI listeners identified the "target" interval containing an IC differing from the "reference" values of either 0.7 or 1.0. Stimuli were generated by modulating 1-kHz pulse trains by speech-derived envelopes the IC of which were manipulated. The same stimuli were used in a second procedure in which the perceived degree of fusion was quantified for ICs varying between 0.4–1.0. Preliminary results from both studies suggest that relative insensitivity to changes in IC are associated with a lesser degree of perceived fusion, overall.

3pPPa5. Towards a comprehensive model of binaural processing. Mathias Dietz (Dept. für Medizinische Physik und Akustik, Universität Oldenburg, Küpkersweg 74, Oldenburg, Deutschland 26129, Germany, mathias.dietz@uni-oldenburg.de), Bernhard Eurich, Jonas Klug, and Jörg Encke (Dept. für Medizinische Physik und Akustik, Universität Oldenburg, Oldenburg, Germany)

Most classic models of binaural processing are partly built on assumptions that have been challenged or even contradicted by physiologic or psychoacoustic data. For example, that delay lines encode interaural time difference (ITD), that "binaural filters" are wider than auditory filters, or that the binaural system is sluggish and cannot encode rapidly changing ITDs. While these assumptions have been motivated by psychoacoustic data, they have also been contradicted by other psychoacoustic data. The present study revisits and challenges the respective reasonings. One result is that the psychoacoustic proof of delay lines contained an unmet assumption. Instead of "sometimes wider filters" we propose a spectral incoherence interference mechanism that appears to resolve the impasse of inconsistent filter bandwidth requirements. In the temporal domain, we model fast ITD encoding and binaural sluggishness is realized by slow reformation of the interaural statistics of auditory objects. The resulting model still has some unproven assumptions; however, it does not need to change its parameters depending on the type of stimulus or task. As the binaural filter bandwidths and processing speeds are now equivalent to their monaural counterparts and consistent with physiological findings, the model may be instrumental in moving to a more unified understanding of auditory processing.

3pPPa6. Echo threshold measurement based on headphone presentation. Teng Cao, Guangzheng Yu (School of Phys. and Optoelectronics, South China Univeristy of Technol., Guangzhou, Guangdong, China), and Dan Rao (Acoust. Lab., School of Phys., South China Univeristy of Technol., Tianhe District, Guangzhou, Guangdong 510641, China, phdrao@scut.edu.cn)

Echo threshold(ET), as the upper threshold of the precedence effect, refers to the critical delay value when the listener transitions from perceiving a single fused sound image to two clear and separated sound images in the presence of two relatively delayed sounds. This study investigates the ET measurement using two kinds of headphone-based virtual sound source presentation methods, i.e., interaural time difference (ITD) lateralization and virtual auditory synthesis (VAS). The "lead-lag" paradigm was adopted in the experiment and an adaptive up-down procedure was used to measure ETs of four stimuli, including noise burst, speech, music and pink noise. The results demonstrate that ET is related to the type of signal and the presentation method. Long-duration signals (i.e., speech, music and pink noise) have greater ETs than short-duration signal(noise burst), which is due to the fact that the temporal overlap between lead and lag sound of long-duration signal would suppress the identification of the lag. ETs of ITD-based presentation were significantly greater than that of VAS-based method for long-duration signal, which indicate that different spatial cues produced by different presentation methods would influence the ET of long-duration signal.

3pPPa7. Comparing the influence of auditory streaming cues on binaural pitch fusion in normal-hearing and hearing-impaired listeners. Nicole Dean (Otolaryngol., OHSU, 3181 SW Sam Jackson Park Rd., Portland, OR 97239, nikki.lavee.dean@gmail.com), Frederick Gallun, and Lina Reiss (Otolaryngol., OHSU, Portland, OR)

Binaural pitch fusion occurs when dichotic signals eliciting a different pitch are perceived as one sound. Previously, we investigated the effects of auditory streaming cues on binaural fusion using a modification of the ABA streaming paradigm. A dichotic stimulus (fixed tone at frequency f_{ref} in the reference ear and variable frequency in the contralateral ear) was presented at alternating intervals with up to 3 "capture" tones at f_{ref}. Participants reported whether they perceived galloping or two streams. Binaural fusion frequency range decreased with the number of capture tones and longer inter-stimulus intervals (ISIs), suggesting competition of sequential grouping with capture tones versus simultaneous grouping (fusion) with the dichotic tone. In the present study, we compared the effects in NH and hearing-impaired (HI) listeners. Preliminary data from 13 NH and 5 HI listeners suggest greater reductions in fusion range with number of capture tones and with longer ISI in HI listeners, especially in listeners with broad binaural fusion. These findings indicate that auditory grouping cues can reduce binaural fusion in both NH and HI listeners. Funded by NIH/NIDCD R01DC013307.

3pPPa8. Effects of visual stimuli on auditory separation of sound images spatially split by synthesized binaural signal. Daisuke Morikawa (Toyama Prefectural Univ., Kurokawa 5180, Imizu, Toyama 9390398, Japan, dmorikawa@pu-toyama.ac.jp), Tsubasa Sakai, and Parham Mokhtari (Toyama Prefectural Univ., Imizu, Toyama, Japan)

In order to clarify the effect of visual stimuli on the spatially split perception of sound images by interaural differences, we conducted spatially split perception experiments under conditions in which visual stimuli were presented on a head mounted display. The auditory stimuli were synthesized binaural signals consisting of two uncorrelated pink noises convolved with the receiver's own head related impulse responses. As manipulated by the listener using the mouse scroll-wheel, the visual and auditory stimuli spread to the left and right in front of the listener, and the listener was made to answer at what point the sound images were spatially split into two. As a result, the detection limit was smaller in the conditions with split visual stimuli than in the condition without visual stimulus. These results suggest that visual stimuli that give the impression of separation reduce the detection limit of the auditorily perceived spatially split sound images.

3pPPa9. Measuring listeners' distraction to spatially presented masking noise conditions. Akira Takeuchi (Golisano College of Computing and Information Sci., Rochester Inst. of Technol., 1 Lomb Memorial Dr., Rochester, NY 14623, akira-musico@outlook.jp), Hwan Shim (Dept. of Elec. and Comput. Eng. Technol., Rochester Inst. of Technol., Rochester, NY), Inyong Choi (Dept. of Commun. Sci. and Disord., Univ. of Iowa, Iowa, IA), and Sungyoung Kim (Dept. of Elec. and Comput. Eng. Technol., Rochester Inst. of Technol., Rochester, NY)

This study investigates listeners' biological responses to speech under different masking noise conditions during a spatial selective attention task. Ten participants with normal hearing took part in the experiment which measured both behavioral and biological responses during a speech identification task with varying noise streams. The target speech stream was played through a front-left speaker, while masking noise, consisting of music and non-intelligent speech, was presented through either the front-right or back-

center speaker. Behavioral performance was assessed based on the correct answer ratio, while biological responses were analyzed using Event-Related Potentials (ERPs) for four distinct conditions. These conditions included two Signal-to-Noise Ratios (SNRs) and two spatial positions of the masking noise stream. Using an EEG measurement headset with dry electrodes, the study collected neural responses from the participants and calculated the ERPs for each of the four conditions. After excluding one outlier, the ANOVA result revealed a difference ($F(1, 3.5)$, $p = 0.0984$) in this listener group's ERP response between the two masker positions, with a lower mean peak value observed for the rear masker. This finding suggests that they were less influenced by the rear-located masking noise.

3pPPa10. The effect of angular disparity between sound source sequences on the perceptual organization of auditory objects. Shuichi Sakamoto (Res. Inst. of Elec. Commun. and Graduate School of Information Sci., Tohoku Univ., 2-1-1 Katahira, Aoba-ku, Sendai, Miyagi 980-8577, Japan, saka@ais.riec.tohoku.ac.jp) and Maho Tamakawa (Res. Inst. of Elec. Commun. and Graduate School of Information Sci., Tohoku Univ., Sendai, Miyagi, Japan)

Within auditory scene analysis is a well-known streaming phenomenon which is observed when listeners group several sounds into "auditory objects" in complex auditory scenes. When two auditory sequences consist of similar sounds, these sequences tend to be perceived as a single stream, but can form segregated streams when the sounds are presented from disparate angles. This study investigated the tolerance thresholds for the angular disparity at which the sound sources would segregate rather than group together into single auditory objects, when 50-ms one-third octave bands of noise were presented in the horizontal plane with changing azimuth angle. Listeners were asked to judge whether the presented noise sequences were perceived as a single stream or not. The results revealed that listeners tend to perceive the presented sequences as a single stream when the angular disparity between noises was small; however, these tolerance thresholds depended on the spatial region within which the sequences were presented. Tolerance thresholds became large when the noise sequences were presented from the side, in comparison to the situation when sequences were presented from in front of the listeners. Results were consistent with the minimum audible angle on the horizontal plane.

3pPPa11. Acoustical verification of binaural features in hearing aids. Anna C. Diedesch (Commun. Sci. & Disord., Western Washington Univ., Commun. Sci. & Disord., 516 High St., MS 9171, Bellingham, WA 98225, anna.diedesch@wwu.edu)

Hearing aid verification systems are designed to verify sound pressure levels in a patient's ear canal, as well as ensure hearing aid features are performing to company specifications. In addition to verifying sound output, verification systems can analyze frequency specific compression, prescription targets, directional microphones, noise reduction performance, etc. However, current verification systems are unable to verify several advanced digital signal processing (DSP) features that companies have included in their hearing aids over the recent years. Particularly, binaural features which require both hearing aids to be analyzed simultaneously to measure interaural cues. Here, interaural cues were analyzed from recordings made on an acoustic manikin in an anechoic chamber across various noise reduction and directional microphone features using open-fit and occluding coupling compared to an unaided condition. During data collection, settings in the fitting software did not always align to hearing aid coupling (i.e., open versus occluding dome conditions), an error likely to occur clinically. For some noise reduction hearing aid conditions, a large interaural level difference (ILD) was introduced to one hearing aid, distorting the sound source location. ILD cues measured here, across different DSP and coupling conditions, suggest verification protocols should include additional features, such as binaural measurements.

3pPPa12. A comparison of binaural cues transmitted by clinically available cochlear-implant stimulation strategies. Paul G. Mayo (Dept. of Hearing and Speech Sci., Univ. of Maryland-College Park, 0100 LeFrak Hall, 7251 Preinkert Dr., College Park, MD 20742, paulmayo@umd.edu), Andrew D. Brown (Dept. of Speech and Hearing Sci., Univ. of Washington, Seattle, WA), and Matthew J. Goupell (Dept. of Hearing and Speech Sci., Univ. of Maryland-College Park, College Park, MD)

Cochlear implants (Cis) effectively restore speech understanding in quiet for many recipients. However, bilateral CI listeners continue to perform poorer than acoustic-hearing listeners on spatial-hearing tasks (e.g., speech understanding in noise and sound localization). One explanation is that current clinically available stimulation strategies and unsynchronized sound processors degrade the fidelity of binaural cues. However, no conclusive comparison of the binaural cues transmitted by different stimulation strategies has been made. Therefore, this study compared binaural cues transmitted by two clinically available cochlear implant stimulation strategies. Two unsynchronized Cochlear Ltd. Nucleus 6 sound processors were placed on a binaural mannequin centered in a horizontal ring of loudspeakers in an anechoic chamber. Acoustic stimuli were presented from the loudspeakers and the resulting electrical pulse trains were recorded for both a peak-picking stimulation strategy and constantly stimulating strategy. Envelope interaural time differences and interaural level differences were extracted from the recordings and compared between stimulation strategies. Compared to the constantly stimulating strategy, results show that the peak-picking strategy led to more variable and less monotonic binaural cues, which could constrain sound-localization abilities. These findings argue for the bilateral synchronization of CIs in multiple domains, including electrode selection in peak-picking stimulation strategies and pulse timing.

3pPPa13. Crossmodal cueing of masker location in a localization-in-noise task. Brian D. Simpson (Air Force Res. Lab, 19041 NE 165th Pl, Woodinville, WA 98072, briandavidsimpson@gmail.com), Nathaniel J. Spencer, Michelle Wang, Robert H. Gilkey (Wright State Univ., Dayton, OH), and Eric R. Thompson (Air Force Res. Lab, Wright-Patterson AFB, OH)

Simpson *et al*. [*JASA* **129**(4), 2489, (2011)] found that localization of a target presented in a simultaneous masker improved when, on a trial-by-trial basis, a preview of the masking stimulus was presented at the location of the upcoming masker. Subsequent research revealed that this improvement resulted solely from cueing the *location* of the masker (i.e., cueing spectro-temporal properties of the masker provided no additional benefit). In the present study, we examined the degree to which the benefit of cueing masker location depended on the modality in which this cue was provided. In separate blocks of trials, the location of the masker was cued auditorily, visually (LED activated at the masker location) and audio-visually, and compared to performance in a no-cue baseline. The results revealed a substantial ($\sim$10 dB) benefit of cueing the masker auditorily over the no-cue condition, consistent with previous results. Importantly, there was also a substantial (albeit smaller) benefit of a visual cue ($\sim$6 dB). Performance with a bimodal cue was no different than performance with an auditory cue. Thus, although hearing a sound from the upcoming masker location was most effective, the ability to effectively cue space crossmodally suggests that the spatial information itself is most critical.

3pPPa14. Spatial separation between two sounds affects the timing of action potentials elicited by the sounds in the rat's auditory midbrain neurons. Mathiang Chot (Dept. of Biomedical Sci., Univ. of Windsor, Windsor, ON, Canada) and Huiming Zhang (Dept. of Biomedical Sci., Univ. of Windsor, 401 Sunset Ave., Windsor, ON N9B3P4, Canada, hzhang@uwindsor.ca)

Timing of action potentials (i.e., spikes) elicited by sounds is used by auditory neurons to encode and process acoustic information. In the presence of multiple sounds, the timing of sound-driven spikes is dependent on the temporal, spectral, and spatial relationships among the sounds. We used two tone bursts with different frequencies to form a train of stimuli that were presented at a random odor and a constant rate. Such a train was used to mimic two competing sounds that occurred at the same (50%) probability or a novel sound (i.e., a low probability oddball sound) that was interleaved with a frequently occurring background sound (i.e., a high probability standard sound). We used the rat as an animal model to study how the spatial relationship between two sounds affected the timing of spikes elicited by the sounds in individual neurons in the auditory midbrain. Results indicate that a lower probability of sound presentation led to a higher temporal precision of the timing of the first spike elicited by the sound and the timing could be affected by a spatial separation between two sounds. These results are important for understanding neural mechanisms responsible for hearing in a natural acoustic environment.

3pPPa15. Adaptation to sentences and melodies when making judgments along a voice/non-voice continuum. Zi Gao (Dept. of Psych., Univ. of Minnesota, 75 E River Rd., Minneapolis, MN 55455, gao00196@umn.edu) and Andrew J. Oxenham (Dept. of Psych., Univ. of Minnesota, Minneapolis, MN)

Perceptual contrastive adaptation effects can be used to identify important perceptual features or categories. Our previous work revealed contrastive adaptation effects between voice and non-voice categories when adapting with repeated single vowels or musical tones. The current study investigated whether the effect generalizes to adaptors with higher ecological validity—sentences and melodies. Ten-step continua between a "voice" (female /a/, /o/, or /u/ vowels) and an "instrument" (bassoon, horn, or viola) were generated for each possible pair. A sentence spoken by a female voice, or a melody played on bassoon or horn, was presented, followed by a test stimulus from along the continuum. Contrastive adaptation effects were observed, with the test stimulus more likely to be identified as a voice following a musical melody and vice versa. Pilot data showed similar trends when the female voice adaptors were replaced with their male counterparts, suggesting that the effects may generalize across speaker gender and fundamental frequency. The results show that contrastive adaptation to voice and non-voice stimuli is a robust effect that does not rely on the repetition of simple adaptors or on a shared frequency range between the adaptors and test stimuli. [Work supported by NIH grant R01DC005216.]

3pPPa16. Accounting for individual differences in the effects of hearing protectors on sound source localization. David J. Audet, Aoi A. Hunsaker (Speech and Hearing Sci., Univ. of Washington, Seattle, WA), Nathaniel T. Greene (Otolaryngol., Univ. of Colorado Anschutz Medical Campus, Aurora, CO), Mallory A. Butler (Otolaryngol., Univ. of Colorado Anschutz Medical Campus, Aurora, WA), David A. Anderson (Elec. Eng., Univ. of Minnesota Duluth, Duluth, MN), Jennifer Jerding, Ted Argo (Appl. Res. Assoc., Inc., Littleton, CO), and Andrew D. Brown (Speech and Hearing Sci., Univ. of Washington, Seattle, WA, andrewdb@uw.edu)

Modern hearing protection devices (HPDs) mitigate the risk of noise-induced hearing loss when used as intended, but negative auditory perceptual side-effects continue to limit usability in critical settings. Dozens of studies, including previous studies by our group, have shown that HPDs lead to significant errors in sound source localization, including large errors in source elevation perception and disorienting front-back confusions. Degradation of performance relative to open-ear listening arises due to peripheral disruptions of monaural and binaural acoustic cues for sound localization; for higher-attenuation devices, reduced audibility can also limit access to these cues, even if cues are relatively intact. A recent multi-site effort by our group leveraged manikin-based acoustic measurements of HPD-induced signal distortions to predict human localization performance during HPD use. Manikin-based acoustic metrics were positively correlated with behavioral performance across a large population of human listeners ($n \geq 120$), enabling discrimination of localization impacts across HPDs, but significant individual variability was evident. Here we consider individual factors that may contribute to variability in HPD localization impact across wearers, including audiometric factors and auricular/acoustic factors. Correlational analyses of individualized predictors with localization performance provide new insight on factors that may differentially constrain – or facilitate – sound localization during HPD use.

3pPPa17. Perceptual anchoring to auditory textures in human listeners. Divya Mehta (Linguist., Macquarie Univ., 16 University Ave., Australian Hearing Hub, Sydney, New South Wales 2109, Australia, divya.mehta@students.mq.edu.au), Kurt Shulver (Dept. of Psychol. & Brain Sci., Macquarie Univ., Macquarie University, New South Wales, Australia), David McAlpine, and Heivet Hernandez Perez (Linguist, Macquarie Univ., Sydney, New South Wales, Australia)

Learning is crucial for the development of species, enabling them to acquire behaviours, accumulate knowledge, and refine skills. An example of implicit and unsupervised learning is "perceptual anchoring," where the brain creates an internal representation of the statistical properties of a stimulus that it encounters repeatedly. Behaviourally, both recognition and discrimination of the repeated stimulus i.e., the anchor, increases over time. Here, we modified a white noise anchoring paradigm (Agus *et al*., 2014) to incorporate synthetic auditory textures (McDermott *et al*., 2013). Four sound textures recordings were used in this study: wind blowing, crackling fire, insects, and bubbling water. Normal hearers identified if two synthetic auditory textures were identical. Participants encountered three blocks of 80 trials per synthetic texture: 20 Fixed-Repeated (two identical excerpts within and across trials), 20 Repeated (two identical excerpts within the trial but never heard again) and 40 Novel (two different excerpts presented within and across trials). Anchoring to the Fixed-Repeated stimuli was observed in all participants. Sensitivity to the Fixed-Repeated stimulus was always significantly higher than to the Repeated stimuli for all textures individually and when collapsed. Here, we demonstrated that synthetized auditory textures are suitable to assess perceptual anchoring in human listeners.

3pPPa18. Disrupting perceptual anchoring to pure-tone sequences in human listeners. Kurt Shulver (School of Psychol. Sci., Macquarie Univ., Australian Hearing Hub, 16 University Ave., Sydney, New South Wales 2109, Australia, kurt.shulver@hdr.mq.edu.au), David McAlpine (Macquarie University Hearing, Macquarie Univ., Sydney, New South Wales, Australia), Nicholas A. Badcock (School of Psychol. Sci., The Univ. of Western Australia, Perth, Western Australia, Australia), and Heivet Hernandez Perez (Linguist., macquarie Univ., Sydney, New South Wales, Australia)

Perceptual anchoring, a process akin to statistical learning, occurs rapidly and without conscious awareness and is integral to our ability to successfully navigate a noisy world. Here, we investigated anchoring abilities in typical hearing and reading participants by implementing an anchoring paradigm (Agus *et al*., 2014) using rapid pure-tone sequences (Barascud *et al*., 2016). We then attempted to disrupt anchoring by applying rapid transcranial magnetic stimulation (rTMS) to frontal cortical regions—areas implicated in the processing of and integration of tone sequences (Abla and Okanoya, 2008). Pure-tone sequences consisted of 50 ms tone-pips that were arranged according to two segments, random (RAND) and regular (REG). RAND segments were generated as tones of random frequencies, and REG segments were generated as in RAND but were iterated to create a repeating pattern. Sequences were presented across three conditions: $\mathrm{REP}_{\mathrm{fixed}}$ (identical sequences repeated across trials), $\mathrm{REP}_{\mathrm{novel}}$ (identical sequences not repeated across trials), and $\mathrm{NonREP}_{\mathrm{novel}}$ (nonidentical sequences not repeated across trials). We observed a significantly higher sensitivity to $\mathrm{REP}_{\mathrm{fixed}}$ relative to $\mathrm{REP}_{\mathrm{novel}}$ across all participants (i.e., an anchoring effect). The disruption of frontal regions using rTMS did not significantly impact overall performance but did alter how participants completed the task over time.

3pPPa19. Towards a yardstick of spatial impression: Psychophysical experiments using virtual piano stimuli. Fusako Ishimura (Yamaha Corp., 10-1 Nakazawa-cho Naka-ku, Hamamatsu, Shizuoka 4308650, Japan, fusako.ishimura@music.yamaha.com), Jorge Trevino, Masaru Tanaka (Yamaha Corp., Hamamatsu, Shizuoka, Japan), and Yasuo Shiozawa (Yamaha Corp., Hamamatsu, Naka-ku, Shizuoka, Japan)

A rich "spatial impression" is an important factor in the product value of musical instruments. However, it also represents a development challenge since there is no established method to measure and control it. We introduce a novel yardstick for spatial impression, changing tacit knowledge into a measurable property. We focused on the difference in spatial impression between digital pianos and grand pianos, and conducted psychophysical experiments using piano sounds to identify the features that contribute to grand pianos' spatial impression. We used multiple virtual stimuli which are created by varying physical parameters and conducted the experiment in a virtual environment including head-tracking from a VR headset. This allowed us to find a trend in the spatial impression and identify a physical parameter that correlates to the perceptual results of the experiment. This parameter, "phase entropy," denotes the time complexity of the interaural phase difference. The correlation coefficient between the psychophysical experiment results based on virtual stimuli and the phase entropy was 0.8752. In addition to the virtual stimuli, we confirmed the correlation between perceptual score and phase entropy in an experiment using actual piano recordings; the correlation coefficient in this case was 0.6500.

Session 3pPPb

Psychological and Physiological Acoustics: Cochlear Mechanics and Physiology (Poster Session)

Gabriel Alberts, Chair
Otolaryngology–Head & Neck Surgery, Harvard Medical School, Massachusetts Eye and Ear, 243 Charles St, Boston, MA 02114

All posters will be on display and all authors will be at their posters from 2:20 p.m. to 3:40 p.m.

Contributed Papers

3pPPb1. Explaining the deterioration of the relative timing accuracy for across-channel by a simple mathematical model. Satoshi Okazaki (Univ. of the Ryukyus, 1 Sembaru, Nakagami-gun, Nishihara-cho, Okinawa 9030213, Japan, sokazaki@grs.u-ryukyu.ac.jp)

The perceptual simultaneity range, within which two asynchronous pure tones are perceived to start simultaneously, and the gap detection threshold are known to be wide when the frequency separation of the tones is large. It is generally said that the accuracy of the relative timing for different frequency channels deteriorates. However, there is no clear explanation of why such deteriorations are necessary. This study aimed to show that a simple mathematical model leads to the deterioration of relative timing accuracy for two tones with different frequencies. As a result of the calculations, the model simulated the deterioration of the perceptual simultaneity range not only with the increase of the frequency separation but also with the decrease of the frequency region of two tones. The model also simulated the behavior of the gap detection threshold for two different frequencies (across-channel) with its asymmetric deterioration. Further, the model simulated the behavior of the gap detection threshold for two identical frequencies (within-channel). These results suggest that one simple mathematical model may explain the mechanisms underlying perceptual simultaneity, within-channel gap detection, and between-channel gap detection for two tones.

3pPPb2. Effect of cortical activity enhancement on medial olivocochlear reflex. Kandai Uchiyama (Dept. of Medical Eng., Chiba Univ., Inage-ku Yayoi-cho 1-33, Chiba-shi, Chiba-ken 263-8522, Japan, u_kandai0815@chiba-u.jp), Sho Otsuka, and Seiji Nakagawa (Chiba Univ., Chiba-shi, Chiba-ken, Japan)

The medial olivocochlear reflex (MOCR) is reported to protects the inner ear from acoustic overexposure. It has been shown that MOCR by fluctuations in cognitive function, e.g., attention and expectation. However, the mechanism by which the cortical cognitive processing modulates MOCR has not been clarified. To investigate this mechanism, we compared the variability of cortical activity and MOCR after a mental calculation task that demands widespread cortical network activities. Cortical activity was assessed by measuring slow vertex response (SVR). Although MOCR strength and SVR amplitude did not change significantly after the calculation task, two measures had reverse dependence on the difficulty of the calculation task; A moderately difficult calculation task increased SVR amplitude while reduced MOCR strength. The results implicate that MOCR variations can be attributed to cortical activity changes. In addition, by simultaneously measuring MOCR and EEG during the calculation task, the time course of the effect of the cortical modulatory effect on MOCR was also examined.

3pPPb3. Wavenumber-frequency relationships in the cochlea: Measurements and models. Gabriel Alberts (Speech and Hearing Bioscience and Technol. Graduate Program, Harvard Univ., Massachusetts Eye and Ear, 243 Charles St., Boston, MA 02114, gabrielalberts@g.harvard.edu) and Sunil Puria (Eaton-Peabody Labs., Massachusetts Eye and Ear, Boston, MA)

A fundamental characteristic of cochlear mechanics is the traveling wave—a transverse wave which propagates from base to apex along the basilar membrane. Experiments and models have shown that its wavenumber (2π/wavelength) increases with stimulus frequency for a given membrane place, linearly at first then quadratically as the stimulus frequency approaches the best frequency (BF; Puria and Steele, 2008, The Senses). The wavenumber-frequency relationship (WFR) beyond the BF in experiments and models remains to be understood. We used finite-element models of gerbil and mouse cochleae to calculate the WFR to frequencies well above the BF. Our models replicated previous findings of the WFR increasing linearly in the long-wave region then quadratically in the short-wave region, but the relationship subsequently turned linear again near the BF region and continued this way at higher frequencies. We hypothesize that this linear increase is due to viscous damping in the cochlea which continues to dominate the response well beyond BF. There is no sign of a mass dominated region in the WFR suggesting that there is no harmonic oscillator resonance at least up to a few octaves above the BF. [Work supported by the Amelia Peabody Scholars Fund and NIDCD R01DC07910, F31DC021079, and T32 DC000038.]

3pPPb4. An expanding mechanical role for the tiny mammalian endolymphatic duct: 1. Modification of Békésy model is specifically needed to cope with ambient pressure variation—with weather, altitude or ocean depth. Eric L. LePage (Director, innerearmechanisms.org, PO Box 2564, Mount Claremont, Western Australia 6010, Australia, ericlepage@innerearmechanisms.org)

von Békésy's pioneering work and two-chamber model with basilar membrane (BM) separating the perilymph chambers was a first-pass assumption which worked to explain basic frequency-band analysis. Yet it did not cover psychoacoustical phenomena. Nor did it cover the repeated demonstrations by this author of baseline displacements of the BM, nor mysterious changes in the volume of the tiny endolymphatic duct—even in human ears with normal thresholds. Since 2009 the duct has been shown to possess all the necessary characteristics of a pressure vessel, viz., it is lined with aquaporins and tight junctions while the cells of the stria vascularis maintain a strict osmotic gradient. We submit that such evolutionary change was necessary because ambient pressure changes are too large to be

corrected by passive equalization and need to be actively eliminated to preserve hearing sensitivity. Any theory must cover marine mammals which have adapted to ocean pressures measured in megapascals. In this case, infrasound is modelled as a varying pressure pedestal which must be cancelled. This video presentation covers one possible mechanical configuration for how such a cancelation might be achieved—a hypothesized feedback control system utilizing outer hair cell somatic motility plus our previously-modelled otolith-based pressure transducer.

3pPPb5. Modeling of auditory nerve fiber input/output functions near threshold. Ian C. Bruce (Dept. of Elec. & Comput. Eng., McMaster Univ., Rm. ITB-A213, 1280 Main St. W, Hamilton, ON L8S 4K1, Canada, brucei@mcmaster.ca), Abigail Buller (Dept. of Elec. & Comput. Eng., McMaster Univ., Hamilton, ON, Canada), and Muhammad S. Zilany (Dept. of Elec. and Comput. Eng., Texas A&M Univ. at Qatar, Doha, Qatar)

The instantaneous discharge rate of auditory nerve fibers (ANFs) near their threshold exhibits an approximately exponential relationship to the instantaneous pressure at the eardrum after compensating for the discharge latency. Some simplified models of the pressure-to-discharge transduction process in the cochlea have attributed this exponential relationship to an inner hair cell (IHC) transduction current that is either an exponential function with a level-dependent slope parameter or a first-order Boltzmann function (which is approximately exponential around the zero-input point) followed by a low-pass filter. However, such IHC transduction functions do not produce level-dependent changes in the AC and DC components of the IHC potential that match the behavior observed in physiological recordings. In this study, we show that retaining the physiologically-accurate IHC transduction model of Bruce *et al.* (Hear. Res., 2018) and following it by an exponential or exponential-like function that maps the IHC potential to the input of the synaptic power-law adaptation in that model produces the desired exponential input/output behavior near threshold while preserving the appropriate level-dependent changes in ANF discharge rate and phase-locking. [Work supported by NSERC Discovery Grant #RGPIN-2018-05778.]

3pPPb6. Extending existing models of cochlear mechanics to emphasize physiological mechanisms that support innate perception of harmonic relations among pure tone frequencies. Amitava Biswas (Speech and Hearing Sci., Univ. of Southern MS, 118 College Dr. #5092, USM-CHS-SHS, Hattiesburg, MS 39406-0001, Amitava.Biswas@usm.edu)

Tonotopic mapping along the basilar membrane of the cochlear organ of Corti naturally supports a consistent relationship between the physical frequency of a tone and its perception of pitch. For example, higher frequencies tend to be mapped to higher pitch perceptions. Nevertheless, a precise perception of harmonic relations among pure tone frequencies apparently lacks a strong foundation on the tonotopic mapping mechanism alone. We are presenting a supplementary hypothesis based on temporal delays and cross-correlations between delayed signals in the cochlear fluids and the surrounding structures that provide a robust mechanism to establish innate perception of applicable harmonic relations among pure tone frequencies.

3pPPb7. Influence of cortical inhibition on auditory efferents in humans. Lilly A. Leaver (Linguist, Macquarie Univ., 6/2 finlay Rd., Sydney, New South Wales 2074, Australia, lilly.a.leaver@gmail.com), Heivet Hernandez Perez (Linguist., Macquarie Univ., Sydney, New South Wales, Australia), David McAlpine (Hearing, Macquarie Univ., Sydney, New South Wales, Australia), and Sriram Boothalingam (Linguist., Macquarie Univ., Madison, WI)

Auditory activity in humans is influenced by feedback networks in the brain. The final leg of this pathway originates in the brainstem and plays an important role in protecting and aiding communication in competing noise. Previous studies have used auditory and/or visual attention tasks to manipulate cortical activity when exploring its influence on the periphery via the auditory efferent system. Using subjective attention does not control for variability across individuals in attentional state, alertness, and arousal. Hence, a consensus on the relationship between cortical activity and how it influences more caudal efferent reflexes is still needed. Here we describe the first assay to non-invasively influence the human auditory cortex activity using repetitive transcranial magnetic stimulation and measure its consequence on peripheral structures using otoacoustic emissions. We hypothesized that depressing brain activity temporarily affects auditory functioning. By instigating a virtual cortical lesion in normal hearing participants, we were able to characterize the influence that the auditory cortex has on the sensory cells in the cochlea, mediated via the medial olivocochlear reflex. Our results will open avenues for further research into new diagnostic and therapeutic approaches for hearing disorders.

3pPPb8. Voltage-dependent bundle motion in mammalian auditory hair cells. Jamis McGrath (Otolaryngol., Stanford, 1291 Welch Rd., Biomed, Stanford, CA 94305, mcgrathj@stanford.edu)

Auditory hair cells have bundles of protrusions on their surface held together by proteinaceous links. Sound-induced vibrations deflect these bundles, transferring force through the links to open nearby mechanically sensitive ion channels and providing our sensation of sound. An increase in bundle compliance caused by channel gating can also be detected as motion. The channels open during depolarization and the bundle moves rapidly toward shorter stereocilia, followed by a larger but slower movement toward the taller stereocilia. To determine how voltage changes affect mammalian inner hair cell (IHC) bundles, we used whole cell patch clamp and high-speed imaging to apply electrical stimulation and measure bundle motion. We found two components in the motions of freestanding IHC bundles in response to depolarization, a fast offset away from the tallest row and a slower movement back toward the tallest row. Like in turtles, the slower motion was sensitive to channel block and suggested ion flux, particularly calcium, played a role. Our findings show IHCs can rapidly respond to changes in the bundle setpoint without ion influx. This work improves our understanding of the mechano-electrical transduction process and helps address the controversial role of calcium in regulating the channel current response.

3pPPb9. Contralateral suppression of input/output function of distortion-product otoacoustic emissions in young adults. Sinyoung Lee (Univ. of Yamanashi, 4-3-11 Takeda, Kofu-shi, Yamanashi 400-8511, Japan, leesinyoung@yamanashi.ac.jp), Yuta Hara, and Takuji Koike (The Univ. of Electro-Communications, Tokyo, Japan)

The level of distortion-product otoacoustic emissions (DPOAEs) could be suppressed by contralateral stimulus caused by medial olivocochlear reflex. Several studies reported potential clinical application of contralateral suppression of DPOAEs as evaluation in both cochlear afferents and efferents. In this study, changes of input-output (I/O) function of DPOAEs induced by contralateral stimulation were measured for investigating the characteristic features of the suppression in normal-hearing adults. Subjects were set to be in their 20s or 30s for excluding the influence of aging. Pink noise was used for contralateral stimulation and magnitude of the noise was set to from 60 dBA to 65 dBA. As the results, there were two types of changes in I/O function of the cubic DP component ($2f_1$-f_2). First type shows slight reduction of the DP level overall without change of the saturation curve. The other type shows significant reduction when low sound pressure level of probe tone was applied, and the probe tone level where start to saturate was shifted to higher probe tone level. Suppression magnitude of the cubic component was higher in subjects who had higher level of not only cubic component but also quintic component ($3f_1$-$2f_2$).

3pPPb10. Measurement of distortion-product otoacoustic emissions at low frequency and effect of contralateral acoustic stimulation on its level. Takuji Koike (Mech. and Intelligent Systems Eng., The Univ. of Electro-Communications, Tokyo, Japan, t.koike@uec.ac.jp), Yuta Hara (Mech. and Intelligent Systems Eng., The Univ. of Electro-Communications, Tokyo, Japan), and Sinyoung Lee (Univ. of Yamanashi, Yamanashi, Japan)

Measurement of distortion-product otoacoustic emissions (DPOAEs) is one of the functional tests for the outer hear cells in the cochlea. Since the sound levels of the DPOAEs are quite low, it is easy to be buried in noise especially in the low frequency range, and frequency analysis and time

averaging are required to detect them. In this study, focusing on the frequency stability of two stimulus sounds and a most prominent DPOAE-component, we tried to reduce noise floor and shorten the measurement time by choosing an adequate time window length for averaging the sound pressure waveform in the ear canal and the number of the averaging. As a result, it has become possible to measure DPOAEs at the low frequencies around 300 Hz in a relatively short time. Using this method, the changes in the DPOAE level induced by contralateral acoustic stimulation, which are caused by medial olivocochlear reflex, were measured. Although some exceptions were observed, the DPOAE was suppressed with increasing the sound pressure of the contralateral stimulus, and the degree of suppression was greater for the lower-frequency DPOAEs.

3pPPb11. Efferent involvement in neural adaptation to acoustical stimuli: Insights from otoacoustic emissions. Amanda Bridges (Dept. of Linguist., Macquarie Univ., The Australian Hearing Hub, 16 University Ave., Macquarie Park, New South Wales 2109, Australia, amanda.bridges@students.mq.edu.au), Jason Mikiel-Hunter (Dept. of Linguist., Macquarie Univ., Sydney, New South Wales, Australia), David McAlpine (Hearing, Macquarie Univ., Sydney, New South Wales, Australia), and Sriram Boothalingam (Dept. of Linguist., Macquarie Univ., Madison, WI)

Neural adaptation to sound level statistics has been demonstrated at various levels of the auditory pathway, including the auditory periphery. Adaptation is thought to improve the efficiency of encoding acoustic stimuli using limited neural resources without compromising accuracy. However, the precise mechanisms underlying the statistical learning of an acoustic environment are not fully understood. This includes the potential contribution to the stimulus-specific modulation of afferent auditory nerve activity by the medial olivocochlear reflex (MOCR), an efferent feedback loop linking the brainstem to the cochlear amplifier. We used otoacoustic emissions (OAEs) to investigate evidence of neural adaptation in the MOCR in response to broadband elicitors of fluctuating intensity whose mean levels were either predictable (alternating low [48 dB SPL] and high [66 dB SPL]) or unpredictable (randomly occurring low and high mean levels). Magnitudes and time-courses of the MOCR obtained in normal-hearing ears were compared for differences based on the presence or absence of an acoustically predictable context. Our findings will provide valuable insights into the role of the efferent auditory system in neural adaptation to acoustical stimuli at the auditory periphery.

3pPPb12. Comparison of different auto-detection methods for wave V with a wireless automated auditory brainstem response (AABR) measurement system. Chao-Min Wu (Elec. Eng., National Central Univ., Chung-Li 32001, Taiwan, wucm@ee.ncu.edu.tw), Kang-Cheng Peng, and Ming-Xiu Yang (Elec. Eng., National Central Univ., Chung-Li, Taiwan, Taiwan)

The wireless automatic auditory brainstem response measurement system developed in the previous research uses Kalman filter with exponential weight averaging method (Kalman filter with EWA) to filter signal and used the differential method to detect the Wave V of ABR. However, the signals are too noisy to be accurately determined. Therefore, this study compares the Wavelet Kalman filter and Moving average to the Kalman filter with EWA, and the fitted parametric peak (FPP) to the differential method, respectively. The simulation results showed that the signal-to-noise ratio of ABR is the highest after Wavelet Kalman filtering, and the audiologist can mark the waves III and V of ABR the most. The latency of the wave V detected by FPP was used to calculate the Pearson product-moment correlation coefficient with the audiologist's manual mark. The two latencies are highly correlated, and the correlation coefficient is higher than the differential method. Therefore, the Wavelet-Kalman filter and FPP algorithm were implemented in the AABR measurement system. To evaluate the accuracy of the algorithm in this study, subjective experiments were conducted with four normal hearing subjects. The Mann-Whitney U test was used to test the difference between the average value of automatic detection and reproducibility.

3pPPb13. Towards the origin of devotion and happiness: An acoustical and neuro-cognitive exploration of Indian spiritual music. Archi Banerjee (Rekhi Ctr. of Excellence for the Sci. of Happiness, Indian Inst. of Technol. Kharagpur, Kharagpur 721302, India, archibanerjee7@gmail.com), Medha Basu (Sir C. V. Raman Ctr. for Phys. & Music, Jadavpur Univ., Kolkata, India), Shankha Sanyal (School of Lang. and Linguist., Jadavpur Univ., Kolkata, West Bengal, India), and Priyadarshi Patnaik (Rekhi Ctr. of Excellence for the Sci. of Happiness, Indian Inst. of Technol. Kharagpur, Kharagpur, India)

Since the age of the Vedas, devotion has remained a key component of Indian music through centuries of changes and foreign influences. The brightest example of this is the *Bhakti* tradition, a pan-Indian movement (7th-15th Century CE), which integrated poetry and music in the transmission of spiritual and social goals. Devotees often report perception of emotions like devotion, happiness, awe while listening to spiritual music of their own religion and culture. This paper aims to study the acoustical and neuro-cognitive correlates of these emotions for two Indian spiritual music traditions – (a) *Sikh Gurbani*/ *Shabad Kirtan*, (b) *Bangla Vaishnav Kirtan*, both of which emerged from the *Bhakti* tradition. 5 Punjabi and 5 Bengali speaking participants listened to 3-minute long four songs from these two spiritual music traditions while EEG signals were recorded from each participant along with their emotion responses. Acoustical time series of these music clips and their corresponding EEG signals were analysed using nonlinear DFA technique. DFA Scaling Exponent values were calculated for multiple 30-second EEG segments across the total duration of each music clip to understand the perception and induction process of devotion and happiness in human brain, which is a novel step in the domain of music cognition and signal processing.

3pPPb14. What does bilingual experience do to music processing in the brain? Ronald Deng (Hunter College High School, 228-09 56th Ave., Oakland Gardens, NY 11364, thenumber8398@gmail.com) and Yan H. Yu (Commun. Sci. & Disord., St. John's Univ., Bayside, NY)

The relationship between language and music in the brain has constantly been a topic of discussion. Understanding how and to what extent music and language experiences influence cortical dynamics for brain functions (e.g., music and speech processing) will open a crucial door to treat a myriad of neurological conditions with an auditory or language base. Bilingual experience affects nonverbal cognition, and the neural coding of pitch is shaped by language experience in the adult population. However, it is not clear whether and how different bilingual development affects music processing differently. The purpose of this study was to determine whether different early bilingual language experience modulates cortical sensitivity to music features. Participants were exposed to varying musical phrases. The electroencephalograms (EEGs)/event-related potentials (ERPs) where recorded during changes in musical features. The ERPs were compared among bilingual English-Mandarin and bilingual English-Spanish pre-adolescents and adolescents. Preliminary results suggested that bilingual teenagers with English-Mandarin backgrounds showed larger mismatch negativity responses to some, but not all six types of music changes than the bilingual teenagers with English-Spanish backgrounds. Bilingual experience modulates music process and different bilingual experiences influence music processing differently.

3pPPb15. Does harmony function emerge from sequential notes in a diatonic mode? An event-related potential study. Shicheng Zhang (Stanford Univ., 726 Serra St., 473, Stanford, CA 94305, sczhangb@stanford.edu) and Takako Fujioka (Stanford Univ., Stanford, CA)

Harmony can be implied by scales in Western tonal music. This phenomenon has been named chord-scale theory in modern jazz practice. However, neurological evidence of such interaction remains unclear. Previous research utilized Event-Related Potentials (ERP), such as Early Right Anterior Negativity (ERAN) and N5, to study sound expectancy violations and the harmony integration process. The current study investigates whether

sequential musical notes arranged in an ascending diatonic mode (1) violate the sound expectancy and (2) alter the harmonic function of the target chord. We designed a musical sequence, namely Prime A (chords) + Prime B (ascending scale notes) + Target (a chord). Prime A and B set up global and local tonality accordingly. We hypothesized that the processing of the target chord would be influenced by Prime B in the harmonic dimension. For instance, a Neapolitan sixth target chord (F, Ab, and Db) is incongruent under C Ionian but could be congruent with C Locrian or C Phrygian. We found ERAN and N5 amplitudes at the target chord are altered by (1) the implied harmonic degree of the local tonality and (2) the relative distance between the local and global tonality in the circle of the fifth space. Our results suggest that Western listeners' brains can interpret musical notes from diatonic mode as part of harmony processing and integrate them into complex tonal music contexts.

3pPPb16. What, if anything, is coincidence detection? Philip X. Joris (Neurosciences, KU Leuven, Herestraat 49, Bus 1021, Leuven B-3000, Belgium, Philip.Joris@kuleuven.be), Hsin-Wei Lu, Tom Franken (Neurosciences, KU Leuven, Leuven, Belgium), and Philip H. Smith (Neurosci., UW-Madison, Madison, WI)

Neural delays and coincidence detection are neural operations posited as key elements in temporal processing, in the auditory system and beyond. Although longstanding physiological evidence for coincidence detectors exists at a phenomenological level, mechanistic examinations have only recently been performed. We review intracellular recordings from two cell types which are traditionally regarded as prototypes of coincidence detectors. Neurons in the medial superior olive (MSO) are sensitive to "coincidences" in action potentials from the ipsi- and contralateral ear, making the neurons sensitive to interaural temporal differences. Octopus cells in the cochlear nucleus are sensitive to "coincidences" of action potentials across inputs tuned to a wide range of frequencies. We found that in both cases, the spike output of neurons does not simply reflect coincidences of input spikes. When inputs vary in strength, their sequence of activation is an important determinant of response strength. Maximal output is not just determined by the maximal number of coincidences, but reflects temporal input structure in interaction with intrinsic (membrane) properties. These observations suggest a simpler mechanism of temporal sensitivity than traditional lag and coincidence proposals.

3pPPb17. The effect of medial olivocochlear reflex enhancement associated with temporal expectation on listening performance in noise. Yuki Ishizaka (#721, Sci. & Technol. Bldg.II, Chiba Univ., 1-33 Yayoi-cho, Inage-ku, Chiba-shi, Chiba 263-8522, Japan, ishizaka.y@chiba-u.jp), Sho Otsuka, and Seiji Nakagawa (Chiba Univ., Chiba-shi, Chiba-ken, Japan)

The medial olivocochlear reflex (MOCR) is efferent feedback activated by acoustic stimulation and plays a role to improve signal detection in noise. We have reported that the MOCR and phase locking value (PLV) of delta oscillations showed a similar decreasing tendency with increasing jitter added to the preceding sound sequence. This suggests that the processing at the cortical regions is involved in the regularity -based enhancement of MOCR. Further, the dependency disappeared as the load of the interfered task increased, suggesting that the cortical predictive control of MOCR requires top-down attention to sounds. Considering the anti-masking effect of MOCR, MOCR enhancement with anticipation may improve listening performance in noise. We extended our previous experiments by measuring listening performance in noise and compared its dependence on the jitter size with MOCR strength and PLV of delta oscillation. In the low load condition, the three indices increased with decreasing the jitter. In the high load condition, by contrast, the dependency on the jitter size disappeared. The similarities of the indices in the dependency on jitter size and attention implicate that the cortical predictive control can facilitate the antimasking of MOCR, therefore improving the listening performance in noise.

Session 3pSC

Speech Communication: Speech Production II (Poster Session)

Isabel S. Schiller, Chair
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All posters will be on display from 2:20 p.m. to 4:20 p.m. Authors of odd-numbered abstracts will be at their posters from 2:20 p.m. to 3:20 p.m. and authors of even-numbered abstracts will be at their posters from 3:20 p.m. to 4:20 p.m.

Contributed Papers

3pSC1. Acoustic and articulatory modifications of Mandarin vowels in clear and Lombard speech. Yung-hsiang Shawn Chang (Dept. of English, National Taipei Univ. of Technol., No. 1, Sec. 3, Zhongxiao Rd., Taipei 10608, Taiwan, shawnchang@mail.ntut.edu.tw)

This study investigated the acoustic modifications of Mandarin vowels /a, i, u/ in clear speech and Lombard speech. Their underlying articulatory movements in the tongue and lips were also examined using ultrasound imaging and lip videos. Plain (i.e., conversational), clear and Lombard speech productions were elicited through a series of map tasks, where participants were prompted by pre-recorded videos to interact with the listener on the screen. Results showed that clear and Lombard speech in Mandarin resulted in longer syllable duration as well as greater vowel RMS intensity than plain speech. Spectrally, not all three vowels were modified to enhance their phonological features in response to clear or Lombard speech conditions. However, the differential F1/F2 modifications did result in an expanded vowel space. Some associations and mismatches between F1/F2 measurements and articulatory movements were observed. Particularly, greater mouth opening and lowering of tongue body in /a/ in clear and Lombard speech could account for the raising of F1. Few modifications in /i/ tongue position and lip configurations also corresponded to no significant F1/F2 change. The acoustic-articulatory relationship for /u/, in contrast, was less straightforward. These observations are discussed in relation to existing clear speech and Lombard speech literature.

3pSC2. Right, but not left, posterior superior temporal gyrus is causally involved in vocal feedback control. Hanjun Liu (Rehabilitation Medicine, The First Affiliated Hospital of Sun Yat-sen Univ., 58 Zhongshan 2nd Rd., Guangzhou, Guangdong 510080, China, lhanjun@mail.sysu.edu.cn), Dongxu Liu (Rehabilitation Medicine, The First Affiliated Hospital of Sun Yat-sen Univ., Guanghzou, China), and Jeffery A. Jones (Psych., Wilfrid Laurier Univ., Waterloo, ON, Canada)

The posterior superior temporal gyrus (pSTG) has been implicated in the integration auditory feedback and motor system for controlling vocal production. However, the question as to whether and how the pSTG is causally involved in vocal feedback control is currently unclear. To this end, the present study selectively stimulated the left or right pSTG with continuous theta burst stimulation (c-TBS) in healthy participants, then used event-related potentials to investigate neurobehavioral changes in response to altered auditory feedback during vocal pitch regulation. The results showed that, compared to control (vertex) stimulation, c-TBS over the right pSTG led to smaller vocal compensations for pitch perturbations accompanied by smaller cortical N1 and larger P2 responses. Enhanced P2 responses received contributions from the right-lateralized temporal and parietal regions as well as the insula, and were significantly correlated with suppressed vocal compensations. Surprisingly, these effects were not found when comparing c-TBS over the left pSTG with control stimulation. Our findings provide evidence that supports a causal relationship between right, but not left, pSTG and auditory-motor integration for vocal pitch production. This lends support to a right-lateralized contribution of the pSTG in not only detecting vocal feedback errors but also driving motor commands for error correction.

3pSC3. Articulatory working space in first and second languages. Yunjung Kim (School of Commun. Sci. and Disord., Florida State Univ., 201 W. Bloxham St., Warren Bldg., Tallahassee, FL 32301, ykim19@fsu.edu) and Austin Thompson (Dept. of Commun. Sci. and Disord., Univ. of Houston, Houston, TX)

Articulatory working space (acoustic and kinematic) is often studied to understand the overall size (limits) of a speaker's articulatory behaviors. For example, prior research has shown that the magnitude of articulators' movement (e.g., maximum tongue advancement, lip aperture) changes as a function of speech effort (loud, clear, and slow speech). To better understand second language acquisition in adults (i.e., articulatory working space determined by the language or anatomical differences), we compare both acoustic and kinematic working space of adult learners of English between their first language (L1) and second language (L2), which is also compared with that of native speakers of English. Specifically, the articulatory convex hull is measured during passage reading for both acoustic (F1 and F2 trajectories) and kinematic (tongue trajectories on x- and y-dimensions) data. Participants include 11 adult learners of English (four men and seven women) with a Korean language background and 10 adult L1 speakers of English (six men and four women). In the presentation, the findings will be discussed to address whether articulatory space is (1) different between native and nonnative languages (determined by linguistic needs) or (2) rather a constant articulatory characteristic of speakers between the languages, regardless of the speaker's English proficiency.

3pSC4. Vocal fold contact pattern during phonation and comparison to Hertzian contact theory. Zhaoyan Zhang (Univ. of California, Los Angeles, 1000 Veteran Ave., Ste. 31-11, Los Angeles, CA 90095, zyzhang@ucla.edu)

The vocal folds are subject to repeated collision during phonation. The resulting contact pressure is often considered to play an important role in vocal fold injury, and has been measured in many experimental studies. In this study, vocal fold contact pattern and contact pressure during phonation were numerically investigated and compared to Hertzian contact theory. The results show that vocal fold contact in general occurs within a horizontal strip on medial surface, first appearing at the inferior medial surface and propagating upward. Because of the localized and traveling nature of vocal fold contact, sensors of a finite size may significantly underestimate the peak vocal fold contact pressure (by about as large as 80% for a sensor diameter of 2 mm), particularly for vocal folds of low transverse stiffness or thick medial surface. In general, the contact pressure increases with the size and curvature of the contact area, as predicted from Hertzian contact theory, indicating the possibility of estimating vocal fold contact pressure from

medial surface kinematics. However, deviations from Hertzian theory were also observed, which need to be taken into consideration in estimating vocal fold contact pressure.

3pSC5. Speaker embedding space cosine similarity comparisons of singing voice conversion models and voice morphing. Elaine M. Liu (Elec. Eng., National Tsing Hua Univ., No. 101, Section 2, Guangfu Rd., Hsinchu 300, Taiwan, eliu@gapp.nthu.edu.tw), Jih-Wei Yeh, Jen-Hao Lu, and Yi-Wen Liu (Elec. Eng., National Tsing Hua Univ., Hsinchu, Taiwan)

We explore the use of cosine similarity between x-vector speaker embeddings as an objective metric to evaluate the effectiveness of singing voice conversion. Our system preprocesses a source singer's audio to obtain melody features via the F0 contour, loudness curve, and phonetic posteriorgram. These are input to a denoising diffusion probabilistic acoustic model conditioned with another target voice's speaker embedding to generate a mel spectrogram, which is passed through a HiFi-GAN vocoder to synthesize audio of the source song in the target timbre. We use cosine similarity between the converted audio's speaker embedding and that of the target voice as an objective metric in two tasks. First, we show that we can morph between two voices: a smooth transition between two speaker embeddings in latent space results in a smooth transition of timbre in generated audio. This shows potential for creativity in the speaker embedding latent space to represent new voices. Second, we use cosine similarity to compare our diffusion acoustic model with a model based on DurIAN. We find that the latter has better conversion results, fewer parameters, and less training time. Overall, we conclude that cosine similarity is a helpful objective metric for voice morphing and conversion.

3pSC6. An electroglottographic study on the effect of following context on glottal constriction in Australian English coda /t/. Louise Ratko (Linguist., Macquarie Univ., Sydney, New South Wales, Australia), Joshua Penney (Linguist., Macquarie Univ., Macquarie Univetrsity, New South Wales, Australia), and Felicity Cox (Linguist., Macquarie Univ.16 University Ave., Macquarie University, New South Wales, Australia, felicity.cox@mq.edu.au)

To achieve inhibition of voicing for voiceless speech sounds, the glottis may be either spread or constricted. Glottal spreading may lead to breathiness, and constriction to glottalization. Previous research has shown that glottalization associated with English coda /t/ differs as a function of surrounding environment, with glottalization more likely when the following word begins with a sonorant consonant (such as /l, n/) compared to other environments. However, it is unclear the extent to which following sonorant consonants induce glottalization compared to following vowels. We used an electroglottograph to record 12 Australian English speakers producing phrases eliciting coda /t/ in target words preceding words with onset sonorants /n, l/ (*now, long*) and onset vowels /əʉ, ə/ (*only, again*). Open quotient (OQ) was established through the second half of the target word vowel. Increasing OQ indicates breathiness whereas lowering OQ indicates constriction. Using generalized additive mixed modelling we found that the OQ of vowels preceding coda /t/ in the sonorant-following context was significantly lower than in vowel-following contexts. This indicates greater glottal constriction when a sonorant consonant follows the coda /t/, showing that coda /t/ voicelessness is implemented differently before sonorant consonants compared to vowels

3pSC7. The babble of articulatory learning networks. Aarya N. Menon (Dept. of Linguist., Univ. of AB, 4-32 Assiniboia Hall, Edmonton, AB T6G 2E7, Canada, anmenon@ualberta.ca) and Clark Arenberg (Ann Arbor, MI)

Articulatory learning models utilize traditional neural network approaches to model speech. However,they differ in their inclusion of an intermediary step in which the model is trained to produce sound utilizing a physical model of the human vocal tract. This style of training is theoretically able to represent the impact of the physical nature of the human vocal tract on language in a way that is more directly translatable to the physical world. This project seeks to create an articulatory learning model which is trained to recognize and repeat input sounds utilizing the University of Wisconsin's X-Ray Microbeam Database. We attempt this utilizing an inverse model which seeks to predict the vocal tract configuration of a given speech signal, and a forward model which predicts the acoustic form produced by vocal tract configurations. Further, we analyze the performance of the model across various training lengths to investigate the comparison to child language learning that is commonly made when implementing models of this type. Should this analysis of articulatory learners be valid, we look to find commonalities in the phonetic patterning of articulatory learning neural networks and those observed in the phonetic acquisition stages of child language development.

3pSC8. Kinematic redundancies in /u/ production with and without dysarthria. Jeffrey J. Berry (Speech Pathol. & Audiol., Marquette Univ., Harriet Barker Cramer Hall, 230L, Milwaukee, WI 53233, jeffrey.berry@marquette.edu)

The speech sensorimotor control system exploits kinematic redundancies: various articulatory degrees of freedom can exhibit trading relationships in service of a speech goal. These redundancies affect movements within and across articulators, resulting in multiple "motor equivalent" patterns that may lead to perceptually equivalent phonemes. One classic example of this redundancy is the purported trading relationship between the tongue and lips in controlling the frequency of the second formant (F2). For the acoustic goal of achieving a relatively low F2 frequency in the vowel /u/, neurotypical talkers may exhibit trade-offs in tongue dorsum and lip position to allow motor adaptability while achieving the acoustic goal. The current work examines speech kinematic and acoustic data from many talkers with and without dysarthria during connected speech to further evaluate the proposition that tongue-lip movement trade-offs allow relatively adaptable articulatory movement in service of a relatively stable acoustic goal during /u/. It is hypothesized that idiosyncrasies in the kinematic form of tongue-lip trade-offs may reflect speaker-specific articulatory-acoustic mappings and that dysarthria may compromise the use of kinematic redundancies, particularly when articulation is severely impaired. Discussion will focus on how methodological issues, particularly kinematic and acoustic normalization, influence the findings.

3pSC9. Exploring the relationship between real-time midsagittal images of the vocal tract and volumetric data. Craig Jin (School of Elec. and Information Eng., Univ. of Sydney, Maze Crescentm Bldg, J03, Sydney, New South Wales 2006, Australia, craig.jin@sydney.edu.au), Amelia Gully (Dept. of Lang. and Linguistic Sci., Univ. of York, Heslington, York, United Kingdom), Michael I. Proctor (Linguist., Macquarie Univ., North Ryde, New South Wales, Australia), Kirrie Ballard, Sheryl Foster (Sydney School of Health Sci., Univ. of Sydney, Sydney, New South Wales, Australia), Tharinda Piyadasa, and Yaoyao Yue (School of Elec. and Information Eng., Univ. of Sydney, Sydney, New South Wales, Australia)

3D analysis of the vocal tract using dynamic MRI remains a technically difficult challenge. Various approaches have been explored such as using parametic models of the vocal tract (Yehia *et al*., 1997); integrating data across parallel slices of 2D dynamic data (Zhu *et al*., 2012); applying stack-of-spiral MRI sampling with 3D constrained reconstruction (Zhao *et al*., 2020); and combining static 3D and dynamic 2D data (Douros *et al*., 2019). In this work, we follow a similar approach to Douros *et al*. and explore the relationship between 2D real-time midsagittal images and 3D volumetric scans of the vocal tract. The real-time MRI midsagittal images are recorded during vowel-consonant-vowel vocal tasks, while the 3D volumetric scans are recorded during sustained vowels. We use large deformation diffeomorphic metric mapping as the foundation for this modeling work. We explore techniques to use constraints provided by the real-time MRI midsagittal images to enable smooth deformations of the 3D volumetric data. We focus on the feasibility of the methods and report on the types of constraints explored and the resulting deformations of the 3D volumetric data.

3pSC10. Exploring velar function in speech: An integrated approach using nasometry and high-speed nasopharyngoscopy. Liran Oren (Dept. of Otolaryngol., Univ. of Cincinnati, 231 Albert Sabin Way, Cincinnati, OH 45229, orenl@ucmail.uc.edu), Brittany Fletcher (Dept. of Commun. Sci. & Disord., Univ. of Cincinnati, Cincinnati, OH), Huy Le (Dept. of Otolaryngol., Univ. of Cincinnati, Cincinnati, OH), Yashish Maduwantha, Carol Espy-Wilson (Dept. of Elec. and Comput. Eng., Univ. of Maryland, College Park, MD), Mark Tiede (Haskins Labs., New Haven, CT), and Suzanne Boyce (Dept. of Commun. Sci. & Disord., Univ. of Cincinnati, Cincinnati, OH)

It is well established that coupling between the oral and nasal cavities is controlled by velopharyngeal opening and closing movements and that coarticulatory patterns in these movements can affect both the intelligibility and the naturalness of speech. In addition, poor control can cause individuals to develop undesirable compensatory patterns. Most studies of these patterns have relied on indirect methods to assess both extent and timing of movement due to the difficulty of direct observation of the velopharyngeal port. This study focused on the correlation between indirect and direct methods of assessing both timing and magnitude of velopharyngeal opening for coarticulation using complementary data from nasometry and a novel technique – high-speed nasopharyngoscopy. Nasometry measures the ratio of distinct nasal and oral cavity outflow to determine the degree of nasality in speech. High-speed nasopharyngoscopy captures images of the nasopharyngeal region and can resolve velar motion during speech. By aligning these data, we can infer the functional changes of the velopharyngeal port and show the correlation between its status and resulting acoustic effects. Preliminary findings replicate previous work on coarticulatory patterns associated with prosodic structure and low versus high vowels. This work will illuminate how variability in velar function affects phonetic outputs.

3pSC11. Real-time imaging of vocal tract configuration in Australian english vowel production. Michael I. Proctor (Macquarie Univ., 16 University Ave., North Ryde, New South Wales 2109, Australia, michael.proctor@mq.edu.au), Kirrie Ballard, Craig Jin (Univ. of Sydney, Sydney, New South Wales, Australia), Amelia Gully (Univ. of York, Heslington, York, United Kingdom), Sheryl Foster, Tharinda Piyadasa, and Yaoyao Yue (Univ. of Sydney, Sydney, New South Wales, Australia)

Dynamic phonetic properties of Australian English vowels have been well described in acoustic (Cox, 2006; Elvin *et al*., 2016) and articulographic studies (Ratko *et al*., 2023), but the global configuration of the vocal tract during Australian English vowel production has not previously been examined. Midsagittal configuration of the upper airway during production of Australian English vowels was tracked using real-time magnetic resonance imaging (Kennerley *et al*., 2022) on a 3T scanner (Siemens Magnetom Prisma) using a 64-channel head/neck receiver array coil. Speech audio was simultaneously recorded in-scanner at 16 kHz using a ceramic noise-canceling microphone (Opto-acoustics FOMRI-III). 3D configuration of the vocal tract during sustained productions of the same monophthongs was captured using volumetric imaging of the upper airway at a resolution of 2 mm $\times$ 2 mm $\times$ 2 mm. These data offer new details on configuration of the entire vocal tract and the relationship between articulatory and acoustic targets during production of Australian English vowels.

3pSC12. Robustness of lateral tongue bracing in second language speech. Grace Bengtson, Amanda Moniz, Maria Samarskaya, Yadong Liu (Dept. of Linguist., Univ. of BC, Vancouver, BC, Canada), and Bryan Gick (Dept. of Linguist., Univ. of BC, Vancouver, BC, Canada, gick@mail.ubc.ca)

Lateral tongue bracing, a universal speech posture, is an actively maintained robust position, akin to standing and locomotion [Gick *et al*., 2017, JSLHR 60; Liu *et al*., 2022, JIPA, Phonetica 79]. Studying the robustness of bracing in second-language (L2) speakers can provide insight into learning of speech posture. In this study, we investigate the extent to which speaking in a second language affects the robustness of lateral tongue bracing compared to first language (L1) speech. Participants read two short texts, one in their native language and one in their second language, under a 10 mm bite-block condition. Intra-oral videos are analyzed for the percentage of time bracing occurred in continuous speech. Our preliminary results showed a decrease in the overall percentage of lateral contact during L2 continuous speech compared to L1. These insights invite a reassessment of our existing parallel posture model. We suggest adapting this model to account for the cognitive demands of L2, and the possible role of learning language-specific articulatory settings in postural robustness for L2 speakers.

3pSC13. The frequency modulation transfer function provides supplemental measures for the objective evaluation of pitch extractors. Hideki Kawahara (Ctr. for Innovative Res. and Liaison, Wakayama Univ., 930 Sakaedani, Wakayama 640-8510, Japan, kawahara@wakayama-u.ac.jp), Ken-Ichi Sakakibara (Health Sci. Univ. of Hokkaido, Ishikari, Hokkaido, Japan), and Kohei Yatabe (Tokyo Univ. of Agriculture and Technol., Koganei, Tokyo, Japan)

We propose to apply the frequency modulation transfer function as a supplemental measure for evaluating pitch extractors' performance. We introduced a new family of time-stretched-pulse for acoustic system analysis (Kawahara and Yatabe, ICASSP2021, called CAPRICEP). Composing a test signal using the CAPRICEP units by Walsh-Hadamard sequence enables simultaneous measurement of the linear-time-invariant, non-linear-time-invariant, and random-time-varying attributes. We applied the proposed analysis method to eighteen pitch extractors and visualized their behavior (represented more than five thousand plots) as a movie. The proposed procedure will be available as open-source tools in our GitHub repository.

3pSC14. Effects of horizontal-to-vertical stiffness ratio on the vibration characteristics of a two-mass vocal fold model. Daiki Kurosawa (Toyohashi Univ. of Technol., 1-1 Hibarigaoka, Toyohashi, Aich 441-8580, Japan, kurosawa.daiki.or@tut.jp), Tsukasa Yoshinaga (Osaka Univ., Toyohashi, Japan), and Akiyoshi Iida (Toyohashi Univ. of Technol., Toyohashi, Japan)

In order to evaluate the effects of stiffness ratio of vocal fold on three-dimensional behavior of vocal fold vibration, numerical simulations were performed by using a two-mass vocal fold model with different stiffness ratio of the horizontal and vertical direction. A ratio of the spring constants for the same direction were fixed, while it for the orthogonal directions was changed. To simulate the airflow through the glottis, the one-dimensional Bernoulli's equation was solved with an equivalent circuit for the compressible flow propagation. The vocal fold shape was represented by a smooth shape with cylindrical masses and tangential faces. The flow separation point was estimated using the boundary layer theory. The results showed that overall fundamental frequency of the vocal folds increased with increasing horizontal stiffness, whereas the vertical stiffness changes had few effects on it. We also found that there were topical conditions in which fundamental frequency rapidly changed with a few changes in stiffness ratio. The stiffness ratio at that time was from 0.80 to 1.08. Before and after the change, there was difference whether the motion trajectory of mass was circular or linear. These results indicate that the vertical stiffness largely affects the oscillation modes of vocal folds.

Session 3pSP

Signal Processing in Acoustics: Signal Processing Potpourri II

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Contributed Papers

1:00

3pSP1. Frequency shifting of transmitted ultrasound in thick composites containing fiber wrinkles and its application in non-destructive evaluation. Andong Cao (Tongji Univ., Yangpu District, Shanghai, 200082, China, 1747356286@qq.com), Zhen Zhang, and Qian Li (Tongji Univ., Shanghai, China)

Thick composite structures are prone to defects such as fiber wrinkles and voids during manufacturing. It is challenging to accurately evaluate the forming quality of the structures using conventional ultrasonic inspection, due to the coupling effects between time-frequency features of ultrasound and mixed-type defects. Propagation behaviors of through-thickness transmitted ultrasound in thick composites with fiber wrinkles, voids and no defects were comparatively investigated by both simulation and experiment in this study. Influence of out-of-plane fiber wrinkles on the ultrasonic frequency shifting was studied by comparing the time-frequency characteristics of simulated ultrasonic signals after propagating different distances in the intact and wavy composites. Changes of transmission coefficients of converted transverse waves caused by fiber wrinkles were concluded be the reason for the appearance of high-frequency components above the excitation frequency in the transmitted ultrasound signal. Thereby ultrasonic defect inspection indices based on the phenomenon of ultrasonic frequency shifting were defined. Effects of interaction angles between ultrasonic propagation direction and fibers on the defined inspection indices was further investigated quantitatively. The thick composites with diverse defects were successfully diagnosed using the ultrasound frequency shifting and energy dissipation behaviors.

1:20

3pSP2. Matching pursuit algorithm for cement evaluation behind the casing. Said Assous (DSSP, Weatherford, Gotham Rd., Loughborough LE126JX, United Kingdom, said.assous@weatherford.com), Mark Bacciarelli (DSSP, Weatherford, Loughborough, United Kingdom), and Arezki M. Belloul (Wireline Res. & Eng., Weatherford, Houston, TX)

The oil and gas sector makes extensive use of ultrasonic tool technology for casing and cement examination for well integrity. This requires the determination of the thickness of the casing, its internal structure, and the mechanical properties of the cement behind the casing. In essence an ultrasonic instrument analyses the ultrasonic pulse echoes returned off the internal surface of the casing. To determine the acoustic impedance of the material behind the casing, it can be difficult to distinguish between the primary reflection from the interior surface of the casing and the ones due to the ringing inside the casing wall itself. The primary reflection offers details regarding the internal geometry of the casing, and the decay of the energy within the casing wall reveals more about how well the cement is bonded to the casing. The ultrasonic signals that the transducer receives when it functions as a receiver are easily contaminated by noise, or part of the reflection may be blocked, or the main waveform and the many ringing waveforms are difficult to separate. As a result, the echo signals will have frequency and phase drift or be deformed. To address the various energy packets and extract the pertinent component independently, a matching pursuit algorithm was developed with a specific dictionary to deal with the different energy packets and extract the relevant component separately. The algorithm was tested on simulated and field data.

1:40

3pSP3. Monitoring of bolt loosening using vibro-acoustic modulation (VAM). Jianbin LI (School of Aerosp. Eng. and Appl. Mech., Tongji Univ., 100 Zhangwu Rd., Shanghai 20000, China, 1145805494@qq.com) and Zhen Zhang (School of Aerosp. Eng. and Appl. Mech., Tongji Univ., Shanghai, China)

Detection of bolt loosening using vibro-acoustic modulation (VAM) has been increasingly investigated in the past decade. However, conventional nonlinear coefficients, derived from theoretical analysis, are usually based on the assumption of ideal wave-surface interactions at the joint interfaces. Such coefficients show a poor correlation with the tightening torque when the joint is under the combined influences from structural and material nonlinearities. A reliable inspection method of residual bolt torque was proposed in this study using Support Vector Regression (SVR) with acoustic features from VAM. By considering material intrinsic nonlinearity (MIN) and dissipative nonlinearity (DN), responses of aluminum-aluminum and composite-composite bolted joints during the VAM test were accurately simulated. SVR were subsequently established based on the database built by a mixing of simulated and experimental nonlinear spectral features when the joints were inspected at different scenarios. The results show that the evaluation of residual torque using the SVR models driven by the acoustic nonlinear responses show a higher accuracy compared with the conventional nonlinear coefficients. Requiring limited experiment data, the proposed method can achieve reliable inspection of bolt torque by including the simulated data into machine training.

2:00

3pSP4. Silent speech recognition using data augmentation based on a three-dimensional lip model. Kenko Ota (Nippon Inst. of Technol., 4-1 Gakuendai, Miyashiro, Minamisaitama, Saitama 345-8501, Japan, otakenko@nit.ac.jp)

In this study, we proposed a data augmentation method using a three-dimensional model of a speaker's face for machine lip reading, which estimates the content of speech without using speech data. The proposed method converted a speaker's face into a three-dimensional model using DECA (Detailed Expression Capture and Animation), and generated a large amount of learning data by rotating the three-dimensional model in different directions. Then we obtained the facial features around the lips as time

series data using dlib, which is an OpenCV module. The time series data is fed into a recognition model. We introduced end-to-end recognition model which is used in continuous speech recognition. The recognition model is based on DeepSpeech2, and we modified the filter size of CNN (convolutional neural network) and the number of layers of BiGRU (bidirectional gated recurrent unit) to adapt to our goal. We evaluated the proposed method on ten ordinary Japanese words. We used the phoneme error rate as the evaluation index. As a result of the evaluation, the error rate of about 0.32 was achieved even for data where the speaker of the evaluation data was not included in the speaker of the training data.

2:20–2:40 Break

2:40

3pSP5. Abstract withdrawn.

3:00

3pSP6. Multipath time delay estimation based on the improved Wigner-ville distribution. Yifei Zou (National Key Lab. of Underwater Acoust. Technol., Harbin Eng. Univ., Harbin 150001, China, zyf0422@hrbeu.edu.cn), Xiukun Li, and Ge YU (National Key Lab. of Underwater Acoust. Technol., Harbin Eng. Univ., Harbin, Heilongjiang, China)

The accuracy of underwater multipath channel time delay is crucial in underwater acoustic signal processing. The temporal resolution of the traditional matched filter is restricted by the transmitted signal bandwidth. Wigner-Ville distribution (WVD) which has excellent energy concentration characteristics and effectively avoids the interaction between temporal and frequency resolutions, and is widely used in non-stationary signal processing. However, considering the multipath signal processing, the cross-term in echo WVD would cause inaccurate time delay estimation. Therefore, in this paper, a novel multipath signal time delay estimation method based on the improved WVD is proposed. By utilizing the difference in the continuity of energy changes between auto-term and cross-term in the time-frequency domain, the Seeded Region Growing algorithm is applied and improved to extract auto-term in the WVD. The simulation results show that the method removes cross-term and maintains high concentration characteristics of auto-term, compared with pseudo-WVD and smooth pseudo-WVD, etc. In addition, it improves the accuracy of multipath signal delay estimation and has some anti-noise ability.

3:20

3pSP7. Real-time, adaptive microphone subset selection for a 128-element array. William G. Frazier (NCPA, Univ. of MS, 145 Hill Dr., P.O. Box 1848, University, MS 38677, frazier@olemiss.edu) and Noah Knutson (NCPA, Univ. of MS, University, MS)

Some advantages of high-sensor count microphone arrays include gain over random noise, increased multiple-source resolution, and potential increases in effective frequency bandwidth. But in real-time applications, computational capabilities of embedded computing hardware might limit the number data channels that can be processed without data loss. For example, given a specific processing pipeline and sampling rate, perhaps only 24 of 128 channels can be supported in real-time. Therefore, the question of which 24-element subset of the 128 possible elements should be being used to detect and track multiple moving acoustic sources. In this presentation, principles of genetic search algorithms are used to develop a method to adaptively select optimal subsets of sensors as data is being processed. Discussion concerning optimality criteria, types of genetic algorithms, and algorithm performance will be presented as well as results corresponding to a 3-D 128-element array of digital MEMS microphones when multiple types of moving acoustic sources were nearby.

3:40

3pSP8. Estimating forward-looking sonar (FLS) image quality using water column data features. Greg Vetaw (Arizona State Univ., 650 E Tyler Mall, Tempe, AZ 85281, gvetaw@asu.edu), Suren Jayasuriya (Arizona State Univ., Tempe, AZ), Brian ODonnell, and Wendy Newcomb (Georgia Tech Res. Inst., Smyrna, GA)

Forward-looking sonar (FLS) underwater imaging systems are primarily used for measuring seafloor backscatter. These returns, however, may be occluded due to scattering from other objects in the water such as fish, bubbles, or other suspended objects, and in-band acoustic interference from sources like engines or other acoustic transmissions. The presence of non-seafloor scatterers or interference is most easily observed and estimated in the water column data, the portion of the time-series after the sonar has transmitted a ping, but before that ping has scattered off the seafloor and returned to the FLS. Analysis of the water column time-series returns can be leveraged to estimate the quality of seafloor imagery. In this study, we characterize bright scatterers that appear in the water column data using statistical tools and generate corresponding quality metrics to evaluate imagery.

Session 3pUW

Underwater Acoustics and Signal Processing in Acoustics: Mobile Underwater Acoustic Sensor Networks: Communication, Localization, and Networking Challenges I

Aijun Song, Cochair
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Chair's Introduction—12:55

Invited Papers

1:00

3pUW1. Acoustic modems and models in constrained marine spaces. Michael B. Porter (Heat, Light, and Sound Res., Inc., 1130 Wall St., Ste. 518, La Jolla, CA 92037, Porter@HLSResearch.com), Timothy F. Duda, Peter A. Traykovski, Jennifer J. Johnson, Sandipa Singh, and Kevin Manganini (Woods Hole Oceanographic Inst., Woods Hole, MA)

Constrained underwater spaces such as harbors and ports present many interesting acoustic features and challenges. Quays, wharves, seawalls, ships, floats, bridge piers, jetties, are often part of the "scenery." Passive and active acoustic security systems are often used in these settings. Our focus is on how acoustic communications between system modems are affected by these reflectors. The evidence is mainly anecdotal, but there is a concern that the complexity of the environment can be bad for communications, just as a convention center room can have lousy acoustics. One might think that standard ocean acoustic models would allow one to study this. However, ocean acoustic modeling has been shaped by the needs of open ocean applications. There the standard process has been to assume that the sound launched along a specific azimuth, stays in that vertical plane. This is the so-called Nx2D approximation and not appropriate to model the above mentioned features that knock the sound out of its launch plane. We have been developing a 3D beam tracing model for feature-laden applications and testing it in various sites. Here we discuss the modeling and the measurements in terms of both the fundamental echo structure and the effect on acoustic modems.

1:20

3pUW2. Coordinated target localization and tracking with node scheduling for a network of autonomous underwater vehicles. Wen Xu (Zhejiang Univ., 38 Zheda Rd., ISEE, Hangzhou 310027, China, wxu@zju.edu.cn), Cheng Jiang, and Jianlong Li (Zhejiang Univ., Hangzhou, China)

Underwater mobile sensor networks (UMSNs) have the advantages of large area coverage, flexible network structure, and good local resolution. Particularly, spatial diversity can be exploited through node mobility to get more target information. To take those advantages in the context of regional cooperative target localization and tracking, a joint node scheduling and acoustic model fusion method is proposed for an UMSN consisting of multiple autonomous underwater vehicles (AUVs). First, a target tracking algorithm framework based on the random finite set theory is established. To take the acoustic environment into account, the *a priori* acoustic field and target distribution information are combined to evaluate the detection probability. A path-planning scheme with node scheduling is then developed based on the information gain to optimize the detection performance. To achieve effective node scheduling, three reward functions with different information gains respectively derived from the Rényi divergence and Fisher information matrix are proposed and compared. The rationale and the performance of the developed method are verified via simulations. A simplified version of the approach is also tested in field experiments with three AUVs, verifying the effectiveness of node scheduling in a realistic scenario.

1:40

3pUW3. Lessons learned from time reversal communications. Heechun Song (Scripps Inst. of Oceanogr., Univ. of California, San Diego, San Diego, CA, hcsong@ucsd.edu)

Time reversal (TR) is a technique that utilizes spatial diversity to achieve spatial and temporal focusing in complex environments. This approach has been successfully applied to phase-coherent acoustic communications in time-varying multipath environments, providing an alternative to conventional adaptive multichannel equalizers (M-DFE). Essentially, the TR method involves combining multichannels as a spatial precombiner, effectively reducing channel complexities and achieving optimal performance. This presentation will discuss the insights gained from using the TR approach for underwater communications under diverse underwater conditions.

2:00

3pUW4. Orthogonal time-frequency space (OTFS) modulation for underwater mobile acoustic communications. Yukang Xue (ECE, Lehigh Univ., Bethlehem, PA) and Yahong R. Zheng (ECE, Lehigh Univ., 404B Packard Lab, 19 Memorial Dr. West, Bethlehem, PA 18015, yrz218@lehigh.edu)

This paper investigates the Orthogonal Time-Frequency Space (OTFS) Modulation for Underwater Mobile Acoustic Communications where the communication channel suffers from severe multipath and Doppler effects simultaneously. Practical OTFS modulation schemes with different parameters are designed for acoustic transmission at a center frequency of 115 kHz and a bandwidth of 11.5 kHz. The schemes are tested in lake experiments where the transmitter was anchored and the receiver was towed by a boat at a speed of approximately 1 m/s or 3.6 km/h. The receiver utilizes low-complexity channel estimation and equalization algorithms, such as NLMS (Normalized Least Mean Squares) and IPNLMS (improved proportionate NLMS) algorithms. The results show some insights of the OTFS scheme for acoustic communications. First, the mobile acoustic channel characteristics are different in the 2D delay-Doppler domain than those in the 1D time and frequency domains for the Single-Carrier Coherent Modulation (SCCM) and the Orthogonal frequency division modulation (OFDM), respectively. Second, in mobile scenarios, the OTFS scheme receiver effectively and significantly reduces the accuracy requirements of the Doppler compensation algorithm and provides better frequency-selective fading suppression and Doppler effect robustness compared to SCCM and OFDM schemes. Third, the OTFS scheme has better anti-multipath performance and reduces multipath interference by effectively differentiating signals on different paths in the 2D delay-Doppler domain.

2:20–2:40 Break

2:40

3pUW5. R&D of a low-complexity acoustic communication payload for small-size AUV. feng Tong (Xiamen Univ., Xiping Bldg., Xiamen University, Xiangan Campus, Xiamen, Fujian 361102, China, ftong@xmu.edu.cn), yifang Qiu (Xiamen Univ., Xiamen, China), shuaifeng hao (Xiamen Univ., Xiamen, Fujian, China), and Fumin Zhang (Hong Kong Univ. of Sci. and Technol., Hong Kong, Hong Kong)

Due to its features of low-cost, deployment-convenience and low detection probability, small size AUV is drawing more and more attention from diverse marine fields. While the technique of acoustic communication is capable of enabling wireless link of small size AUV, difficulties caused by adverse underwater acoustic mobile channel such as multipath, Doppler and noise pose significant challenges. Moreover, extremely strict constaints in terms of size, power supply and real-time caculation capability means that R&D of acoustic communication payload need to seek a tradeoff between the algorithm performance and system overhead. To deal with the abovementioned hurdles, a lightweight acoustic communication payload for small-size AUV is designed. Specifically, the proposed payload adopts symbol-wise low-complexity Doppler estimation and compenstaion process, which is incorporated with a compact channel equalizer to achieve reliable mobile communication under Doppler and multipath conditions. The hardware and software design associated with the efficient system implementation of the proposed scheme is given. Finally, lake trial results performed based on small size AUV equiped with the proposed acoustic communication payload is presented to verify its effectiveness.

3:00

3pUW6. Doubly-spread channel equalization based on channel shortening in mobile underwater acoustic communications. Xingbin Tu (Ocean College, Zhejiang Univ., Zhoushan, China, xbtu@zju.edu.cn), Shaojian Yang, Jie Xi, Yulin Jiang, Wei Yan, and Fengzhong Qu (Ocean College, Zhejiang Univ., Zhoushan, China)

The shallow water environments and the movement of autonomous underwater vehicles contribute to the distortion of acoustic signals in both the delay and Doppler domains. Numerous studies have focused on eliminating the distortion in underwater acoustic communications but have struggled to find an optimal trade-off between performance and computational complexity, rendering them impractical for marine engineering applications. In our research, we propose a channel-shortening equalization scheme based on frequency-domain decision feedback equalization (FD-DFE) to address this issue. This approach leverages FD-DFE to reduce the number of multipaths and decreases the Doppler spread of the channel. A time-domain decision feedback equalization based on recursive least squares (RLS-DFE) is then employed to eliminate residual delay and Doppler spreads. The mismatch between the block- and symbol-wise equalization is minimized by an overlapping subblock structure. Numerical simulations have demonstrated the effectiveness of this scheme. Error-free transmissions with a data rate of up to 6 kbps were achieved in a reservoir experiment between an autonomous underwater helicopter (AUH) and a shore-base receiving node.

3:20

3pUW7. Extrinsic information based enhanced Doppler compensation for high-speed underwater acoustic mobile communication. Yaokun Liang (School of Electron. and Information Eng., South China Univ. of Technol., Shaw Sci. Bldg., 381 Wushan Rd., Tianhe District, Guangzhou 510640, China, msyaokun@mail.scut.edu.cn), Hua Yu, Jiahong Lin, Fei Ji, and Fangjiong Chen (School of Electron. and Information Eng., South China Univ. of Technol., Guangzhou, China)

In high-speed underwater acoustic mobile communication, achieving robust Doppler compensation is crucial due to the unpredictable changes in the Doppler factor. Due to the difficulty in optimizing Doppler estimation in turbo equalization through closed-form methods, a joint design of Doppler compensation, channel equalization, and decoding, is proposed to improve the accuracy of the Doppler compensation through integrated methods. Specifically, the proposed scheme utilizes extrinsic information provided by the decoder to evaluate the effectiveness of Doppler compensation because the extrinsic information indirectly reflects the accuracy of Doppler estimation. The proposed algorithm performs parallel Doppler resampling through multiple joint Doppler compensation, equalization, and decoding to optimize Doppler estimation by searching for the best extrinsic information. With iterative Doppler compensation and channel equalization, the accuracy of Doppler estimation is enhanced, leading to improved signal detection performance. The proposed

algorithm was verified by underwater test data collected in Qiandao Lake, Hangzhou, China in May 2016. The results of this evaluation provide practical validation and demonstrate the effectiveness of the proposed algorithm in real-world underwater acoustic communication scenarios.

Contributed Papers

3:40

3pUW8. Downlink Non-Orthogonal multiple access underwater acoustic communication receiver design Deep neural network. Habib H. Zuberi (College of Underwater Acoust. Eng., Harbin Eng. Univ., No.145, Nantong St., Nangang District Harbin City, Heilongjiang Province, China, Harbin, Heilongjiang 150001, China, habib_zuberi@hrbeu.edu.cn), Songzuo Liu, and Mohammad Bilal (College of Underwater Acoust. Eng., Harbin Eng. Univ., Harbin, Heilongjiang, China)

Downlink non-orthogonal multiple access (NOMA) is necessary to handle the underwater acoustic channel's limitations on bandwidth. NOMA downlink underwater acoustic (UWA) communication transmits data symbols from a source station to many users. Superimposed coding with variable power levels allows successive interference cancelation (SIC) receivers to decode data. However, SIC receivers require knowledge of channel conditions and channel state information (CSI). This is difficult to acquire, particularly in UWA communication. To address this problem, this paper proposes downlink underwater acoustic using 1D Convolution neural network (CNN). Two users with different power levels and distances from the transmitter employ BPSK modulation to support multiuser communication. Users far from the base station receive the most power. The base station uses superimposed coding. BELLHOP algorithm generates the training dataset with user depth modifications. For training the model, a composite signal passes through the samples of the UWA channel and is fed to the model along with labels. DNN receiver learns the characteristic of the UWA channel and does not depend on CSI. The testing CIR is used to evaluate the trained model. The results are compared to the traditional SIC receiver. The DNN-based DL NOMA underwater acoustic receiver outperformed the SIC receiver in simulations.

4:00

3pUW9. Robust Doppler compensation for low frequency underwater communications. Chang Sung (Underwater Systems, Thales Australia, 274 Victoria Rd., Rydalmere, New South Wales 2116, Australia, chang.sung@thalesgroup.com.au), Patrick Cooper, Daniel La Mela, and Tim Cain (Underwater Systems, Thales Australia, Rydalmere, New South Wales, Australia)

Impairment caused by Doppler frequency shift and harsh channel condition distinguish underwater acoustic communications from radio communications. Slow speed of acoustic waves produces relatively high Doppler shift and result in expansion or compression in time samples, which is known as Doppler scaling. In particular, the Doppler scaling causes significant carrier frequency offset (CFO) in orthogonal frequency division modulation (OFDM). To compensate the CFO, it is essential to resample signal samples according to the Doppler scale. In theory, the Doppler scale can be estimated by measuring the distance between peaks of the matched filter output of consecutive Doppler-invariant preambles. In practice, multipath reflections and non-orthogonality between the HFM and OFDM payload produce large numbers of local false peaks. In this paper, we propose a robust Doppler estimation technique by employing constant false alarm rate (CFAR) filter to remove false peaks. We validate the performance of the proposed scheme through two sea-trials carried out in shallow water near Jervis Bay, Australia. In the sea-trials, a slow-moving towed hydrophone array was deployed as the receiver and the source was a stationary, low source level, vertical transmit array. Sea-trial results demonstrated up to 112 bps information rate using bit-interleaved coded OFDM over 400 Hz bandwidth.

4:20

3pUW10. Characterization and modeling of high-speed underwater acoustic communication channels. Honglu Yan (College of Underwater Acoust. Eng., Harbin Eng. Univ., No. 145, Nantong St., Nangang District, Harbin City, Heilongjiang Province, Harbin 150001, China, 2015053219@hrbeu.edu.cn), Songzuo Liu, Lu Ma, Gang Qiao, and Xue Han (College of Underwater Acoust. Eng., Harbin Eng. Univ., Harbin, China)

High-speed mobile underwater acoustic communication (HSM-UAC) is a relatively unexplored terrain, but its importance is expected to increase with the widespread application of flexible underwater mobile platforms such as AUVs and UUVs. Knowledge of HSM-UAC channels is a prerequisite to achieving reliable communication. This work conducted two HSM-UAC channel measurement experiments with a maximum speed of 20 knots. The analysis of channel time-varying characteristics demonstrates the shortcomings of traditional consistent Doppler frequency offset channel model. The experimental findings reveal the presence of two non-stationary modes in the HSM-UAC channel. For short moving distances, the changes in channel geometry result in each path experiencing a different Doppler scale, thus observing the multiscale-multilag (MSML) channels. For long moving distances, the alteration in sound ray propagation tracks entirely reshapes the channel impulse response (CIR) structure, leading to a rapid decline in channel time correlation. In further analysis, the correlation coefficient of the power delay profile (PDP) is used to measure channel stationarity. Subsequently, the stationary time and corresponding stationary distance are calculated as characterization parameters for the time-varying channel model. The quantitative evaluation results are helpful for the communication signal design and determining the update frequency of channel state information.

ACOUSTICAL SOCIETY OF AMERICA

Silver Medal in

Speech Communication

Ann Cutler

2020

The Silver Medal is presented to individuals, without age limitation, for contributions to the advancement of science, engineering, or human welfare through the application of acoustic principles, or through research accomplishment in acoustics.

PREVIOUS RECIPIENTS

Franklin S. Cooper	1975	Katherine S. Harris	2005
Gunnar Fant	1980	Ingo R. Titze	2007
Kenneth N. Stevens	1983	Winifred Strange	2008
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Arthur S. House	1991	Sheila E. Blumstein	2014
Peter Ladefoged	1994	John J. Ohala	2015
Patricia K. Kuhl	1997	Anne Cutler	2020
		Joanne L. Miller	2021

CITATION FOR ANNE CUTLER

… for contributions to understanding speech recognition by native and non-native listeners, and leadership in speech science.

SYDNEY, AUSTRALIA • 1 DECEMBER 2023

Anne Cutler was born in Melbourne and grew up in Tasmania. She studied German and psychology at the University of Melbourne and received her PhD in psycholinguistics from the University of Texas at Austin in 1975, under the supervision of Don Foss. She held postdoctoral appointments at the Massachusetts Institute of Technology and then the University of Sussex. In 1982, she took up a research position in Cambridge at the Medical Research Council Applied Psychology Unit. In 1993, she was appointed director of the Comprehension Group at the Max Planck Institute for Psycholinguistics in Nijmegen. She became professor of comparative psycholinguistics at Radboud University Nijmegen in 1995. She took up a part-time research professorship at the MARCS Auditory Laboratories at the University of Western Sydney in 2006. This position became full-time in 2013, when she retired from her positions in Nijmegen. She was appointed Distinguished Professor at Western Sydney University in 2018. She played a key role in establishing the ARC Centre of Excellence in the Dynamics of Language and was a program leader of the Centre. Anne died suddenly on her first post-pandemic visit to Europe, in June 2022.

One of Anne's major contributions was to show that speech science at its best is interdisciplinary. Consider her work on the speech segmentation problem: the problem that listeners must recognize individual words to understand talkers' messages while needing to segment those words out of a continuous acoustic signal with no word boundaries reliably marked. Anne worked with computational linguists, phoneticians, and engineers to specify the nature of the segmentation problem and to propose possible solutions; with cognitive psychologists to collect experimental evidence for underlying algorithms and hence develop computational models of speech recognition, with developmental psychologists to explore how segmentation ability arises in infancy, and with neuroscientists to examine the neural implementation of segmentation. Anne thus taught us that although there are different disciplines in speech science, with different questions, traditions, and techniques, our greatest advances in understanding are likely to come when we integrate across those disciplines.

Another of Anne's major contributions was to convince the field that cross-linguistic comparison is essential. We know that some aspects of speech processing are universal (e.g., listeners of all languages have a segmentation problem to solve) and that others are languagespecific (the language the listener hears undoubtedly shapes their perception). This is indeed the thesis of Anne's wonderful 2012 book "*Native Listening*". The critical question, therefore, is where the boundary lies between those aspects of speech processing that are true for listeners and speakers the world over and those that are learned responses to features of individual languages. It is only through an analysis of where speech processing is the same, similar or different across languages and of how processing changes developmentally, that we can answer this question. Anne has given us many examples of the benefits of this comparative approach, not only in her work on speech segmentation but also in her work on lexical stress and on the processing differences between vowels and consonants. Through her rigorous inter-disciplinary approach, she characterized the many ways in which the process of tuning in to the mother tongue benefits the listener in their recognition of native speech but also has costs for their recognition of non-native speech.

Anne was an excellent role model (especially but not only for female scientists), a strict but supportive mentor, and a brilliant motivator. Her boundless enthusiasm for speech science was perhaps her greatest strength. Caroline Junge, one of Anne's 50 PhD students, in an address given after Anne's farewell lecture as professor at the Radboud University, singled out Anne's enthusiasm as the reason she was cherished by her students and mentees. Anne's engagement with and passion for her field, however, extended far beyond the mentorship she gave her students and other members of her own group; countless researchers from around the world benefitted from her advice, her support, and her constructive criticism. She thus shaped current and future generations of speech scientists. She led her field.

Anne's papers have had a major impact and continue to be widely cited. Her h-index on Google scholar currently stands at 97, based on 38607 citations. She made pioneering discoveries on speech segmentation, perceptual learning, the role of prosody in speech processing, and so much more. Her contributions to special issues of the journal Cognition capture Anne's wit and her love of word play. It is entirely unsurprising, given Anne's preeminence, that her work was recognized in many ways. For example, she was the first woman to be awarded a Spinoza prize, the highest award in Dutch science. She was a fellow of the Royal Society, the British Academy, the Netherlands Royal Academy of Sciences, the Academy of the Social Sciences in Australia, and the International Speech Communication Association, and she was a foreign associate of the US National Academy of Sciences. Anne was very proud of her academic pedigree; she traced her ancestry, through a series of "*Doktor-Vaters*", back to Willem Wundt. She gave all her PhD students a graduation present showing how they too were part of that academic family, through her as "*Doktor-Mutter*".

Anne Cutler was awarded the Silver Medal in Speech Communication in 2020, at a time when a physical award ceremony was not possible. She was honored and moved to be recognized by the ASA in this way. It was her wish that the ceremony take place at an ASA meeting in Australia when pandemic restrictions were lifted. The ceremony at the 2023 Sydney meeting respects that wish even though Anne, whose loss has been mourned deeply by the international speech science community, will now not be present. This award recognizes Anne's outstanding contributions to our understanding of speech recognition and, through her role as mentor and leader, her major and continuing influence on the field of speech communication.

JAMES MCQUEEN
CATHERINE BEST
ANN BRADLOW

Session 4aAA

Architectural Acoustics, Psychological and Physiological Acoustics, and Signal Processing in Acoustics: Three-Dimensional Sound Display and Analysis for Virtual Auditory Immersive Environments I

Ning Xiang, Cochair
Graduate Program in Architectural Acoustics, Rensselaer Polytechnic Institute, 110 8th Street, Troy, NY 12180

Lamberto Tronchin, Cochair
Architecture, University of Bologna, Via G.F. Barbieri 76, Bologna, I 40129, Italy

Angelo Farina, Cochair
Univ. of Parma, Parma, Italy

Chair's Introduction—7:35

Invited Papers

7:40

4aAA1. Experimental reconstruction of sound fields over large spatial domains. Efren Fernandez-Grande (Tech. Univ. of Denmark, Kgs. Lyngby, Denmark), Xenofon Karakonstantis (Tech. Univ. of Denmark, Ørsteds Plads, Bldg. 352, Kgs. Lyngby 2800, Denmark, xenoka@dtu.dk), Antonio Figueroa-Duran, and Samuel A. Verburg (Tech. Univ. of Denmark, Kgs. Lyngby, Denmark)

This talk discusses sensing methods and reconstruction techniques for characterizing sound fields over large spatial domains. We focus specifically on reconstructing reverberant soundfields in rooms and enclosures. Two complementary approaches are presented in the talk: on the one hand, we present a method that leverages on modeling the spatio-temporal and statistical properties of enclosed sound fields via wave expansions, enabling to estimate the sound field over a large volume of space. On the other hand, we present a Physics Informed Neural Network, which learns the governing wave equation using a small set of measured data and enables to meaningfully interpolate and extrapolate the sound field to any position where no observations are available. Experiments in real rooms are presented, including an opera hall and an auditoria for oral communication. The experiments successfully demonstrate the volumetric acquisition and three-dimensional characterization of the sound fields. The presented techniques can be of relevance in applications pertaining to immersive and navigable audio, architectural acoustics and heritage preservation.

8:00

4aAA2. Three-dimensional sound recording using directional beams. Mark Poletti (Callaghan Innovation, PO Box 31-310, Lower Hutt, Wellington 5040, New Zealand, mark.poletti@callaghaninnovation.govt.nz)

The recording of three-dimensional spatial audio is typically carried out using microphone arrays that produce higher-order B-format responses, which are defined by spherical harmonics. Spherical harmonics provide a compact, orthonormal representation of the sound field, but can be intuitively challenging for general users because they do not have a direction, as opposed to typical microphones used in the audio industry. Furthermore, spherical microphone arrays typically use microphone angles based on Platonic solids which do not have intuitive directions such as front, rear, left, right, up, and down. This paper considers the use of sets of directional beams that have orthonormal properties in a similar manner to spherical harmonics. Sets of angles that have intuitive directions are used, and a method of approximating orthonormal beams is developed. The approach can also be applied to angles obtained from Platonic solids. Finally, a prototype microphone array that implements the directional recording format is described.

8:20

4aAA3. On the acoustics of "Argentina" and "Costanzi" Opera Houses in Rome by means of 3-D maps representation. Lamberto Tronchin (Architecture, Univ. of Bologna, Via G.F. Barbieri 76, Bologna, BO I 40129, Italy, lamberto.tronchin@unibo.it), Angelo Farina (Univ. of Parma, Parma, Italy), Cobi van Tonder (Architecture, Univ. of Bologna, Cesena, Italy), Antonella Bevilacqua (Univ. of Parma, Parma, Italy), and Ruoran Yan (Architecture, Univ. of Bologna, Cesena, FC, Italy)

The acoustics of Italian-style Opera Houses has been widely studied in the last 30 years, especially after the burning of the Teatro La Fenice in Venice in 1996. Starting from that disaster, the acoustics of these important buildings became a priority, especially for storing as much information about their acoustics as possible. The recent war events in Ukraine have underlined the importance of these acoustics for a proper reconstruction of theaters, and especially Oper Houses. This paper analyzes the acoustics of the two most important opera houses located in Rome, i.e., the "Teatro Argentina" and the "Teatro Costanzi" (also called "Teatro dell'Opera). Both the theaters

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are located in the very city center of Rome and have a strong artistic activities. In both the cases, no info about their acoustics are available among the scientific community. This paper describes the most important results obtained measuring their acoustics using monoaural, binaural b-format microphones. Moreover, in order to determine 3-D acoustic maps, an EM32 Microphone from MHAcoustics was used for capturing sparial RIRs and mapping early reflection in a 360 pictures.

8:40

4aAA4. A real-time auralization framework for scientific experiments. Philipp Schaefer, Lukas Aspoeck, Pascal Palenda, Janina Fels (IHTA, RWTH Aachen Univ., Aachen, Germany), and Michael Vorlaender (IHTA, RWTH Aachen Univ., Kopernikusstr 5, Aachen 52074, Germany, mvo@akustik.rwth-aachen.de)

The software *Virtual Acoustics* (VA) is an open-source auralization framework developed at the Institute for Hearing Technology and Acoustics, RWTH Aachen University, Germany. Using physics-based models for source emission and sound propagation, it allows plausible rendering of virtual scenes based on purely synthetic data. Its modular design provides a variety of rendering modules using different assumptions for the sound propagation. This allows the rendering of various indoor and outdoor scenarios with adjustable degree of complexity. Based on a similarly modular concept, VA allows a flexible way of reproducing the sound via headphones or loudspeaker arrays. For this purpose, the signals can be spatialized using binaural synthesis, higher-order ambisonics or VBAP. The respective scenarios, including source and receiver movement, can be controlled using the C++, C#, Matlab, Python programming languages. Additionally, there are interfaces for the Unreal and Unity game engines allowing the combination with sophisticated visual cues, e.g., using HMDs. The software's flexible structure and configuration possibilities enable the controlled application in listening experiments while providing an adequate level of plausibility. This work discusses the options as well as the limitations of the software and presents application examples in scientific experiments.

9:00

4aAA5. Reconstruction of individualized near-field head-related transfer functions from a small set of far-field data based on tensor decomposition. Tong Zhao (School of Phys. and Optoelectronics, South China Univ. of Technol., Guangzhou, China), Bosun Xie (Acoust. Lab., School of Phys. and Optoelectronics, South China Univ. of Technol., Guangzhou 510641, China, phbsxie@scut.edu.cn), and Jun Zhu (School of Phys. and Optoelectronics, South China Univ. of Technol., Guangzhou, China)

Head-related transfer functions (HRTFs) are essential for virtual auditory display. Generally, near-field HRTFs vary with source direction, distance, frequency, and individual. The huge dimensionality of data makes the measurement or calculation of individualized near-field HRTFs difficult. In the present work, a method to estimate individualized near-field HRTFs with high directional and distance resolution from a small set of directional measured or calculated data at a far-field distance is proposed. Based on tensor decomposition, the near-field HRTFs are decomposed into a weighted combination of direction-, distance-, frequency-, and a few of individual-related modes. The universal direction-, distance-, and frequency-modes as well as the weights are evaluated from a baseline near-field HRTFs dataset. For an arbitrary new individual outside the baseline dataset, the individual modes are estimated from a small set of directional measured or calculated of far-field HRTFs, and then the individualized near-field HRTFs with high directional resolution are estimated. An example of analysis indicates that 15 individual modes account for more than 98 % individual-related energy variation of HRTFs; and near-field HRTFs with high directional resolution can be estimated from far-field data of 30 directional measurements or calculations. Two psychoacoustic experiments validate the proposed method.

9:20–9:40 Break

9:40

4aAA6. Enhancing acoustic contrast while tracking acoustic transfer functions: A simultaneous control approach. Jing Lu (Nanjing Univ., 504 Acoust. Bldg., 22th Hankou Rd., Nanjing, Jiangsu 210093, China, lujing@nju.edu.cn), Meiling Hu, and Haishan Zou (Nanjing Univ., Nanjing, Jiangsu, China)

In the context of sound zone control (SZC) systems, the ability to effectively track variations in acoustic transfer functions is essential due to the inevitable variation of acoustic environments. This paper addresses the non-uniqueness problem encountered in time domain SZC, arising from the coupling observed in multichannel transfer function modeling. To address this issue, the utilization of time-varying control filters is proposed, which offers a viable solution. In order to ensure a stable acoustic contrast performance regardless of environmental changes, as well as to mitigate extreme distortion resulting from maximal acoustic contrast (AC) in the time domain, the paper focuses on the objective of maintaining a desired level of AC while tracking the acoustic transfer functions. Simulations are carried out to evaluate the proposed method, demonstrating its capability to effectively track environmental changes and continually update the control filters, consequently providing a stable acoustic contrast performance.

10:00

4aAA7. Divided spectro-temporal transformer for sound event localization and detection in real scenes. Yusun Shul (Elec. Eng., Korea Adv. Inst. of Sci. & Technol., 291 Daehak-ro, Yuseong-gu, Daejeon 34141, Korea (the Republic of), shulys@kaist.ac.kr) and Jung-Woo Choi (Elec. Eng., Korea Adv. Inst. of Sci. & Technol., Daejeon, Korea (the Republic of))

Sound event localization and detection (SELD) involves the detection of sound events (SED) and the estimation of their direction of arrival (DoA) by utilizing multichannel sound signals. Recent research in SELD has predominantly focused on deep neural network (DNN) based models, which specifically emphasize learning temporal context. Examples of these models include the convolutional recurrent neural network (CRNN) and the ResNet-conformer architecture, which handle spectral and channel information only as the embeddings of temporal features. To fully exploit spectral information providing a crucial cue for both SED and DoA, it is imperative to devise a network architecture that effectively learns both spectral and temporal contexts. In this regard, we propose a divided transformer architecture that separately identifies the spectral and temporal context to encourage the model to learn more spectral characteristics of signals while retaining the temporal context. The efficacy of the divided spectro-temporal transformer approach is validated using the

DCASE 2022 and 2023 challenge task 3 datasets. Furthermore, a series of parameter studies carried out to optimize the performance of SELD demonstrates that the number of frequency bins for attention and the pooling location impact the performance, and the divided spectro-temporal transformer is beneficial for both SED and DoA.

10:20

4aAA8. Effects of sounds on the visitors' experience in museums. Milena J. Bem (School of Architecture, Rensselaer Polytechnic Inst., 1605 Hutton St. Apt 10, Troy, NY 12180, jonasm@rpi.edu), Samuel R. V. Chabot, and Jonas Braasch (School of Architecture, Rensselaer Polytechnic Inst., Troy, NY)

Museums increasingly recognize the significance of acoustics for a high-quality visitor experience. Objective/physical parameters have been extensively studied, but there is a need for research focusing on user perception. This study addressed this gap by investigating visitors' soundscape preferences and the effects of different soundscapes on the museum experience. We hypothesized that exhibition-congruent sounds are more effective than conventional masking sounds in reducing distractions, enhancing engagement with artwork, and creating immersive experiences. The research utilized the Cognitive Immersive Room (CIR) at Rensselaer Polytechnic Institute, featuring a 360° visual display with a multi-channel loudspeaker system for spatial audio rendering to project panoramic photographs and ambisonic audio recordings recorded in 16 exhibitions in five relevant museums—for example, MASSMoca and the New York State Museum. Four scenarios were presented to the participants: originally recorded soundscape, recorded soundscape combined with a conventional sound masker, recorded soundscape combined with a congruent sound masker, and silence. After experiencing the immersive environment, 23 subjects responded to a questionnaire. The results showed congruent sounds increased focus and engagement and contributed to a more comprehensive and immersive experience. In 58% of cases, the participants preferred the congruent-sound scenario over the others.

Session 4aBA

Biomedical Acoustics and Physical Acoustics: Cavitation Therapies for Cancer Treatment I

Mike Averkiou, Cochair
Bioengineering, University of Washington, 616 NE Northlake Pl, Box 355013, Seattle, WA 98195

Tatiana Khokhlova, Cochair
Department of Medicine, University of Washington, 1013 NE 40th St., Seattle, WA 98105

Chair's Introduction—8:55

Invited Papers

9:00

4aBA1. Cavitation in an antivascular cancer therapy setting. David Goertz (Univ. of Toronto, Rm. S665a, 2075 Bayview Ave., Toronto, ON M4K 3M5, Canada, goertz@sri.utoronto.ca)

Focused ultrasound in combination with circulating microbubbles is being widely investigated as a means to promote spatially targeted drug delivery. This approach has considerable potential in oncology for a range of therapeutic agents. At sufficiently high pressures, above those typically employed in drug delivery, tumor microvessel damage can be induced to an extent that leads to perfusion shutdown and subsequent ischemic tissue necrosis. This approach is often referred to as antivascular ultrasound (AVUS), which has been shown in preclinical work to be capable of enhancing the effects of radiation therapy, antiangiogenic therapy, chemotherapy, and immunotherapy. At present, the mechanisms of AVUS are not well established. It is important to gain a more detailed understanding of the bubble–microvessel interactions that lead to perfusion shutdown, along with the cavitation signatures associated with these behaviors to enable the rational development of effective cavitation based control methods. This talk will provide a brief overview of AVUS therapy in oncology and highlight recent efforts employing two-photon microscopy, high speed optical imaging, and acoustic emission monitoring to gain insights into bubble behavior within small channels and *in vivo* microvessels under AVUS exposure conditions.

9:20

4aBA2. Cancer ablation with histotripsy. Timothy L. Hall (Univ. of Michigan, 2200 Bonisteel Blvd, 1107 Gerstacker Bldg., Ann Arbor, MI 48109, hallt@umich.edu) and Zhen Xu (Univ. of Michigan, Ann Arbor, MI)

Histotripsy is an ablation method using mechanical action of cavitation bubbles to fracture cells and tissue structures as a non-invasive cancer treatment for tumor masses occurring at locations in the body with a suitable acoustic window. Non-thermal ablation modalities including histotripsy have been shown to enhance immune system stimulation with potential for anti-tumor benefit beyond the targeted volumes. This talk will give an overview of histotripsy cancer research to date treating cancer models and early clinical trials. Key areas of research include instrumentation for treating rodent models, best strategies for spatial-temporal dosing of histotripsy, and working to understand the mechanisms of immune stimulation to best take advantage of abscopal effects and adjuvants for combination therapy. We have constructed several precision histotripsy systems for targeting subcutaneous and orthotopic tumors in mice and rats using a combination of ultrasound, MRI, and stereotactic guidance. Evidence to date has shown the strongest immune stimulation occurrences with moderate cavitation doses. Histotripsy ablation studies show stimulation of tumor-specific immune responses capable of magnifying the impact of checkpoint inhibition immunotherapy.

9:40

4aBA3. Cavitation-enhanced high-intensity focused ultrasound treatment for cancer therapy. Shin-ichiro Umemura (Biomedical Eng., Tohoku Univ., Aoba 6-6-12, Aramaki, Aoba-ku, Sendai, Miyagi 980-8579, Japan, sumemura@ecei.tohoku.ac.jp), Shin Yoshizawa (Elec. and Commun. Eng., Tohoku Univ., Sendai, Miyagi, Japan), Jun Okamoto, and Kazuhito Nemoto (Sonire Therapeutics, Tokyo, Japan)

Cavitation has been known to have a potential to enhance HIFU treatment in a few ways. It can multiply the particle displacement by orders of magnitude and thereby enhance the mechanical effect (1) of ultrasound hugely and the *in vivo* thermal effect (2) significantly. It can even induce chemical effect (3) when its bubble collapses. We are developing a cavitation-enhanced HIFU therapy system by which the effects (2) and (3) are aimed to obtain. The higher priority was set on the effect (2), because combination with a certain drug may be needed for the effect (3), which can make the process for the approval more difficult. *In vivo* cavitation threshold can hugely vary on the conditions. It can be reduced by stabilized microbubbles and nanobubbles, nanodroplets, and certain chemicals. However, these are not assumed in combination with our system for the same reason above. Instead, an ultrasonic pulse at an extremely high-intensity in the order of 10 MPa with a duration in the order of 10 ms is used to generate cavitation, which is immediately followed by a typical HIFU burst to obtain the effect (2). The results from swine tests of our HIFU system will be shown in the presentation.

10:00

4aBA4. Delivery of ultrasound cavitation therapy with a modified clinical scanner. Lance H. De Koninck (Bioengineering, Univ. of Washington, Benjamin Hall, 616 NE Northlake PL, Rm. 361, Seattle, WA 98105, lancedek@uw.edu), Connor Krolak, Kaleb Vuong (Bioengineering, Univ. of Washington, Seattle, WA), Jeffry Powers (Philips Ultrasound, Bothell, WA), and Mike Averkiou (Bioengineering, Univ. of Washington, Seattle, WA)

Ultrasound-induced bioeffects from stable and inertial cavitation of microbubbles in tumors have been shown to enhance systemic drug delivery. However, the high pressures and long cycle lengths needed for cavitation treatment are not available on current clinical ultrasound scanners. Here, we implement acoustic conditions (1.6 MHz, 12-500 cycles, $<$3 MPa, 10-200 Hz PRF) suitable for cavitation treatments (USCTx) on the S5-1 probe of a Philips EPIQ scanner. Hydrophone measurements confirmed the acoustic conditions. Passive cavitation detection (PCD) in solutions and tissue mimicking phantoms containing microbubbles demonstrated the spatial and temporal extent of the produced cavitation with 20 consecutive pulses. Inertial cavitation was observed at peak negative pressures of $\sim$0.5 MPa and $\sim$0.9 MPa in the solutions and phantoms, respectively. At low PRFs ($<$100 Hz), inertial cavitation was only observed within the first 5-10 pulses for both media, suggesting USCTx may generate most bioeffects in the first few pulses. The higher PRFs sustained cavitation activity for a greater duration, consistent with gas diffusion of micron size bubbles after shell disruption with the first pulse. Having implemented USCTx with a clinical scanner and confirmed the produced cavitation *in vitro*, we plan to evaluate this technique in an upcoming mouse study.

10:20

4aBA5. Impact of ultrasonic inertial cavitation on the pancreatic tumor stiffness in mice. Adrien Rohfritsch, R. A. Drainville (LabTAU, INSERM, Université de Lyon, Lyon, France), Gilles Renault, Giovanna Bibaki, Sarah Urro (Institut Cochin, INSERM, Paris, France), Bruno Giammarinaro, Litan Wang, Jacqueline Ngo (LabTAU, INSERM, Université de Lyon, Lyon, France), Frédéric Prat (Institut Cochin, INSERM, Paris, France), Maxime Lafond (LabTAU, INSERM, Université de Lyon, 151, cours Albert Thomas, Lyon 69424, France, maxime.lafond@inserm.fr), and Cyril Lafon (LabTAU, INSERM, Université de Lyon, Lyon, France)

Pancreatic ductal adenocarcinoma (PDAC) is among the fastest-growing cancers and is predicted to become the second leading cause of cancer-related deaths worldwide by 2030. Ultrasound-induced inertial cavitation (IC) is an emerging therapeutic strategy with promise for treating patients with unresectable tumors. However, the physical impact of IC on tumors remains unclear. In this study, we used orthotopic murine models of pancreatic cancer and employed passive elastography to measure elasticity before and after applying IC to the whole tumor volume. The IC was generated using two focused confocal transducers (center frequency, 1.1 MHz). The output level was adjusted with a real-time feedback loop based on broadband signal acquired with a passive cavitation detector. Cavitation clouds were visualized during treatment using an imaging probe. The results from eight treated mice showed a 24% decrease ($p < 0.05$) in tumor stiffness after the IC treatment, indicating that the chosen IC sequence can soften the tumor environment. After sacrifice, we utilized second-harmonic imaging microscopy to visualize the collagen network within the tumor, a known contributor to tissue stiffness, and correlated various features of this network (length, width, curvature) with the measured elasticity.

Session 4aID

Interdisciplinary: Keynote: The Listening Brain's Response to Adversity

Benjamin J. Halkon, Chair
University of Technology Sydney, 15 Broadway, Ultimo 2007, Australia

Chair's Introduction—11:00

Invited Paper

11:05

4aID1. The listening brain's response to adversity. Catherine McMahon (Linguist, Macquarie Univ., cath.mcmahon@mq.edu.au)

How the brain hears, responds to adversity, and adapts with hearing interventions, can inform clinical decisions, device design and, ultimately, broader policy and practice. While audiological practice today remains focused on the use of a limited test battery, improving assessment techniques to identify how well adults comprehend speech in across a range of listening environments (and beyond the pure tone audiogram) can lead to greater personalization of treatments, devices, and rehabilitation. Moreover, understanding how the brain responds to interventions, such as cochlear implants or rehabilitation programs for tinnitus, could improve our understanding of which work, and for whom. In this presentation, I discuss a number of experimental techniques that we have used to explore how the brain responds to sound under challenging listening conditions using speech-in-noise and vocoded stimuli. In addition, I will discuss how the brain adapts after cochlear implantation for those with significant hearing loss, and following a 30-week rehabilitation program for those with significant tinnitus. Finally, potential limitations of electrophysiological measures within current clinical assessment batteries will be considered, as well as opportunities to better understand how we can improve hearing healthcare for individuals and populations.

Session 4aPA

Physical Acoustics and Structural Acoustics and Vibration: Nonlinear Acoustics in Solids

Ching Tai Ng, Cochair
The University of Adelaide, Adelaide 5005, Australia

Anubhav Roy, Cochair
Engineering Science and Mechanics, The Pennsylvania State University, 409 B Earth and Engineering Sciences Building, University Park, PA 16802

Christopher Kube, Cochair
Penn State, 212 Earth and Engineering Sciences Bldg., University Park, PA 16802

Invited Papers

8:00

4aPA1. Investigating the behavior of the slow dynamic recovery at early times. John Yoritomo (Acoust., Naval Res. Lab, 4555 Overlook Ave. SW, Washington, DC 20375, john.yoritomo@nrl.navy.mil)

Slow dynamics (SD), a type of non-classical nonlinear elastic behavior, is characterized by (i) a material softening due a minor mechanical conditioning and (ii) a subsequent slow recovery that approaches the original macroscopic elastic state. Early work in sandstones and other consolidated granular materials—and more recent work in unconsolidated granular materials—have found the recovery to be logarithmic-in-time, at least for seconds to hours after conditioning. However, other recent studies have observed recoveries in consolidated granular materials that deviate from log(time), particularly for early recovery times. This talk will present experimental investigations in to the nature of the SD recovery, especially at early times. The experimental venue will be the single bead system (a single bead confined between two large plates of a similar material) introduced previously. We will measure SD recoveries using multiple methods, such as Dynamic Acoustic Emission Testing (DAET) and Larsen Monitoring, to better compare our experiments with previous work. The effect of different conditioning mechanisms on the early time recovery will also be investigated. The presentation will conclude with some discussion of proposed mechanisms for slow dynamics.

8:20

4aPA2. Effect of high temperature on nonlinear guided wave in reinforced concrete structures. Ching Tai Ng (School of Architecture and Civil Eng., The Univ. of Adelaide, Adelaide, South Australia 5005, Australia, alex.ng@adelaide.edu.au) and Aseem Ahmed (School of Architecture and Civil Eng., The Univ. of Adelaide, Adelaide, South Australia, Australia)

There are some ations that reinforced concrete (RC) structures may be subjected high temperatures, e.g., building fires, rocket and jet aircraft engine blasts, and nuclear power reactors. It is crucial to ensure the integrity and safety of these RC structures. This study investigates the effects of high temperature on the behavior of nonlinear guided wave, which can potentially be used for non-destructive evaluation, in reinforced concrete structures. In this study, the reinforced concrete specimens are heated from 100 to 800°C. Two sets of experimental studies are considered. At temperatures 100–300°C, piezoceramic transducers are embedded at the rebars in the reinforced concrete specimens to generate and measure the longitudinal wave. At temperatures 300–800°C, the longitudinal transducers are installed at the rebars exposed outside the reinforced concrete specimens. The results show that the high temperature induces the debonding between the rebar and the concrete, which causes the generation of the second harmonic when the longitudinal wave interacts with the debonding. A nonlinear parameter is defined to monitor the effects of the heat damage caused by the increasing temperature on the reinforced concrete structures.

8:40

4aPA3. The complexity of harmonically scattered nonlinear waves from triangular, circular, and rectangular corners of the 2-D domain. Pravinkumar Ghodake (Mech. Eng., Indian Inst. of Technol. Bombay, B423, Hostel 14, IIT Bombay, Powai, Mumbai, Maharashtra 400076, India, mech7pkumar@gmail.com)

Considering recent advancements in the nonlinear pulse-echo technique, understanding reflected nonlinear waves from inaccessible edges and surfaces becomes important. A unique geometrical model solved numerically using the finite element method is proposed and studied via extensive numerical experiments to gain insight into harmonically scattered waves from different shapes of the 2-D spaced corners considering the challenges of theoretical solutions that can capture the interplay between multiple phenomena. Tang *et al*. (2012), Kube (2017-18), and Achenbach and Wang (2017-18) studied the harmonic scattering of waves from nonlinear inclusions using analytical techniques. Linear longitudinal waves scattered from the triangular, circular, and rectangular-shaped free and fixed edges of the 2-D spaced corner show mode conversion and energy transfer between bulk wave modes at fundamental frequencies. The interaction of nonlinear ultrasonic waves with the edges makes things complex due to an interplay between harmonic generation, linear scattering, harmonic scattering, bulk wave mode conversion, and harmonic energy redistribution between all harmonics of the scattered longitudinal and transverse waves. This results in non-intuitive interesting responses. These studies are extended to explore one-way and two-way two-wave mixing of longitudinal waves and their interesting nonlinear effects. Phase difference introduced during harmonic scattering distinguishes the sensitivity of fundamental harmonics.

9:00

4aPA4. Evaluation of the Fresnel approximation for the longitudinal particle displacement in shear wave beams. Philip G. Kaufinger (Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX 78758, pkaufinger@utexas.edu), Branch T. Archer (Chandra Dept. of Elec. and Comput. Eng., Univ. of Texas at Austin, Austin, TX), John M. Cormack (Div. of Cardiology, Dept. of Medicine, Univ. of Pittsburgh, Pittsburgh, PA), Kyle S. Spratt, and Mark F. Hamilton (Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX)

The longitudinal particle displacement in shear wave beams generated with disks that vibrate longitudinally at several hundred hertz is used to assess stiffness of soft tissue *in vivo*. Nonlinear elasticity of tissue, although less commonly used than linear stiffness, may be a more sensitive or specific biomarker of disease. A mathematical model for linear propagation of both transverse and longitudinal motion in a shear wave beam in an isotropic elastic half-space, and comparisons with measurements in tissue-mimicking phantoms, was published earlier this year [Archer *et al*., JASA **153**, 1591 (2023)]. When including elastic nonlinearity in a propagation model, the complexity is reduced substantially by deriving an evolution equation based on the Fresnel approximation [Zabolotskaya, Sov. Phys. Acoust. **32**, 296 (1986)]. For application to elastography, the parameter space in which the Fresnel approximation provides an accurate description of the longitudinal particle displacement must be established. In this presentation, the accuracy of the Fresnel approximation is assessed for linear shear wave beams, both unfocused and focused, by comparison with the existing mathematical model. Attention is devoted primarily to the longitudinal particle displacement in nearly incompressible media. [PGK is supported by the ARL:UT Chester M. McKinney Graduate Fellowship in Acoustics.]

9:20

4aPA5. Discrete element models for elasto-plastic wave propagation in nonlinear architected materials. Samuel P. Wallen (Appl. Res. Labs. and Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX, sam.wallen@utexas.edu), Michael Haberman (Appl. Res. Labs. and Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX), and Washington DeLima (Honeywell, Federal Systems, Kansas City, MO)

Nonlinear architected materials are known to exhibit a plethora of dynamic phenomena that enhance our capacity to manipulate elastic waves. Since these properties stem from complex, subwavelength geometry, dynamic simulations at high resolutions are often intractable at scales of interest. Therefore, prior studies have turned to effective medium models that capture essential properties in the long-wavelength limit. One class of such models is a periodic structure composed of discrete mass, spring, and damper elements, whose constitutive relationships are computed to match behaviors observed in experiments or fine-scale simulations of representative volume elements. While models of this type have been implemented successfully in many cases, the majority of literature has been focused on recoverable deformation, possibly including linear damping. However, history-dependent effects, such as friction and plasticity, have been much less explored. In this work, we develop a discrete element modeling framework for nonlinear architected materials undergoing plastic deformation. Due to the history- and rate-dependent characteristics of plasticity, the framework generally yields a system of differential-algebraic equations whose computational cost is significantly greater than an elastic system of comparable size. We demonstrate the method using several examples from the literature and explore means to obtain phenomenological plasticity models for general geometries.

9:40

4aPA6. Theory for highly stressed thick-walled elastic spheres and their vibration without small strain approximations. Darryl McMahon (Curtin Univ., Bldg. 301, Rm. 113, Kent St., Belmont, Western Australia 6102, Australia, d.mcmahon@curtin.edu.au)

This paper considers how the radii and vibration of an elastic thick-walled sphere is affected by high stresses imposed by high external pressures or by high internal pressures. For simplicity, the sphere consists of idealized Hookean material allowing unlimited compression or expansion. The sphere is filled by material with an isotropic pressure that generally differs from the pressure on the outer surface of the sphere. The analysis includes large strain nonlinear curvature effects where, unlike linear elasticity theory, the difference in radial and tangential strains is not small compared to those strains. Despite the sphere's robustness, destructive physical consequences are predicted for nonzero Poisson coefficients where, for instance, a high-pressure tank inner stresses make the inner radius approach the less stressed outer radius. This would be avoided if the Poisson coefficient changed with large strains and approached zero. To model the effect of high stresses and strains on breathing mode vibration, an effective spring constant is derived from quadratic radius deviations of the system potential energy from equilibrium. Insights from analysis avoiding linear elasticity approximations may be applicable to improved understanding of deep-sea marine creature survival, improved underwater vessel design for large depths, and safer containers of fluids at high pressures.

10:00

4aPA7. Single focused-beam acoustical tweezers: Trapping cells in 3-D. Zhixiong Gong (Key Lab. of Marine Intelligent Equipment and System, State Key Lab. of Ocean Eng., School of Naval Architecture, Ocean and Civil Eng., Shanghai Jiao Tong Univ., Dongchuan Rd. #800, Minhang District, Shanghai 200240, China, zhixiong.gong@sjtu.edu.cn), Shiyu Li (Key Lab. of Marine Intelligent Equipment and System, State Key Lab. of Ocean Eng., School of Naval Architecture, Ocean and Civil Eng., Shanghai Jiao Tong Univ., Shanghai, China), and Zhichao Ma (Inst. of Medical Robotics, School of Biomedical Eng., Shanghai Jiao Tong Univ., Shanghai, China)

Single focused beams have shown great potential in acoustofluidics and medical ultrasound because of their large pressure gradient and acoustic-thermal effect, such as acoustical tweezers and high-intensity focused ultrasound (HIFU) technique. However, the three-dimensional trapping of typical cells is still challenging since cells in water have a positive acoustic contrast factor and will be repulsive from the focus [Gong and Baudoin, Phys. Rev. Applied **18**, 044033 (2022)]. In this work, to reverse this positive acoustic contrast factor into negative, we propose to use a cell-friendly medium (made of iodixanol with water) [Augustsson *et al*., Nat. Commun. **7**, 11556 (2016)]. This enables a single focused beam to selectively trap and control cells in 3-D, and more importantly, keeps the manipulated cells with good viability. This work will extend the applications of acoustical tweezers, which may be beneficial to single cell analysis, cellular phenotyping, precise assembly of different cells in tissue engineering, and controlled drug delivery.

10:20

4aPA8. Mutual interactions between bulk waves and guided waves in a quadratic nonlinear elastic medium. Christoph Bös (Dept. of Civil Eng., Univ. of Siegen, Paul-Bonatz-Str. 9-11, Siegen 57076, Germany, christoph.boes@uni-siegen.de) and Chuanzeng Zhang (Dept. of Civil Eng., Univ. of Siegen, Siegen, Germany)

The high sensitivity of nonlinear ultrasonic waves to the early stages of local material deteriorations makes them ideal candidates for nondestructive material characterization. Guided elastic waves, which can be emitted and received on the same surface, expand the possibilities to inspect inaccessible domains to test and monitor existing structures. The pure measurement of the amplitude of the second harmonic has only limited significance, since it is difficult to distinguish between the rather weak material nonlinearity and the nonlinearities from the technical equipment. This can be avoided by mixing two ultrasonic waves of different frequencies, which results in superposed harmonics at these frequencies. In this study, the mutual interactions between bulk waves and guided waves in a quadratic nonlinear elastic medium are investigated by highly efficient computations. In addition, numerical examples for the wave interaction in a finite area with nonlinearity in an otherwise linearly elastic host medium are presented and discussed to explore the effects of individual parameters as well as possible experimental implementation.

4a THU. AM

Session 4aPP

Psychological and Physiological Acoustics: Physiology and Psychophysics of Predictive Auditory Scene Analysis and Object Formation

Joerg Encke, Cochair
Macquarie University, 16 University Avenue, Macquarie 2109, Australia

Michael Pecka, Cochair
Biocenter, Neurobiology, Ludwig-Maximilians University Munich, Grosshaderner Str. 2, Martinsried 82152, Germany

Chair's Introduction—7:55

Invited Papers

8:00

4aPP1. Interactions between attention and auditory scene analysis. Barbara Shinn-Cunningham (Neurosci. Inst., Carnegie Mellon Univ., 5000 Forbes Ave., Pittsburgh, PA 15213, bgsc@andrew.cmu.edu)

Do auditory objects form automatically, or only when a listener focuses attention on a sound? This review argues that the question is ill posed—that both are simultaneously correct. Intuition suggests that the processes of source segregation and attention happen sequentially: first segregation parses a complex scene into constituent objects, and then selection pulls out an important sound to be analyzed in detail. But, rather than a processing hierarchy in which object formation occurs first, followed by selection, auditory scene analysis and attention interact in a heterarchy: formation and selection influence one another, feed back upon each other, and are not easily separable in terms of either how they are implemented in the brain or how their effects can be measured. There is not one particular site in the auditory pathway where objects "first appear;" instead, an object-based representation emerges gradually and imperfectly from the auditory nerve through the brainstem and midbrain to the various divisions of the cortex. Similarly, attentional selection accrues at every stage as one moves from the ear to the cortex. This talk will review examples from perceptual and physiological experiments supporting the idea that attention and object formation interact heterarchical.

8:20

4aPP2. Corticofugal regulation of predictive coding. Alexandria Lesicko (Otorhinolaryngology, Univ. of Pennsylvania, 904 S Farragut St., 1, Philadelphia, PA 19143, lesicko1@pennmedicine.upenn.edu), Christopher Angeloni (Northwestern, Chicago, IL), Jennifer Blackwell (Stony Brook, Stony Brook, NY), Mariella De Biasi (Univ. of Pennsylvania, Philadelphia, PA), and Maria Geffen (Otorhinolaryngology, Univ. of Pennsylvania, Philadelphia, PA)

Sensory systems must account for both contextual factors and prior experience to adaptively engage with the dynamic external environment. In the central auditory system, neurons modulate their responses to sounds based on statistical context. These response modulations can be understood through a hierarchical predictive coding lens: responses to repeated stimuli are progressively decreased, in a process known as repetition suppression, whereas unexpected stimuli produce a prediction error signal. A potential substrate for top-down predictive cues is the massive set of descending projections from the cortex to subcortical structures, although the role of these corticofugal neurons in predictive processing has never been directly assessed. We tested the effect of optogenetic inactivation of the auditory cortico-collicular feedback in awake mice on responses of auditory midbrain neurons to stimuli designed to test prediction error and repetition suppression. Inactivation of the cortico-collicular pathway led to a decrease in prediction error and repetition enhancement in these neurons. Overall, our results demonstrate that the auditory cortex provides cues about the statistical context of sound to subcortical brain regions via direct feedback, regulating processing of both prediction and repetition.

8:40

4aPP3. Naturalistic behaviors for studying object identification in auditory cortex. Dardo N. Ferreiro (Div. of Neurobiology, Ludwig-Maximilians-Univerisity Munich, Grosshaderner Str. 2, Martinsried, Bayern 82152, Germany, ferreiro@bio.lmu.de), Diana Amaro, Benedikt Grothe, and Michael Pecka (Neurobiology, LMU Munich, Munich, Germany)

Localizing and identifying sensory objects while exploring the environment are fundamental brain functions. Yet despite decades of study, we still do not fully understand how auditory cortex neurons represent sound locations, or how those representations are affected by context and behavior. In particular, little is known about brain function when sensory inputs are processed during active sensing. Here, we present results of gerbils, freely moving, performing the Sensory Island Task (SIT). In SIT, animals are free to forage for a specific spot (the island) in the experimental arena which, upon being found, triggers a change in the auditory stimulation location; which

animals report by staying within the island for an extended time period, in order to receive a reward. By coupling one of the sound sources to reward delivery, we demonstrate that sound source identity shapes single cell spatial preference in the primary auditory cortex during active navigation. We further showcase other implementations of SIT which enable the study of the neural processes and behaviors underlying active perception in ethologically relevant scenes.

9:00

4aPP4. Listening to the room: Disrupting activity of dorsolateral prefrontal cortex impairs learning of room acoustics. Heivet Hernandez Perez (Linguist, Macquarie Univ., 16 University Ave., Australian Hearing Hub, Sydney, New South Wales 2109, Australia, heivet.hernandez-perez@mq.edu.au), Jessica J. Monaghan (National Acoust. Labs., Sydney, New South Wales, Australia), Jason Mikiel-Hunter (Linguist, Macquarie Univ., Sydney, New South Wales, Australia), James Traer (Dept. of Psychol. & Brain Sci., The Univ. of Iowa, Iowa City, IA), Paul Sowman (School of Psychol. Sci., Macquarie Univ., Sydney, New South Wales, Australia), and David McAlpine (Linguist, Macquarie Univ., Sydney, New South Wales, Australia)

Navigating complex sensory environments is critical to survival, and brain mechanisms have evolved to cope with the wide range of surroundings. In noisy spaces listeners place more emphasis on early-arriving sound energy, nevertheless, reverberant energy is highly informative about those spaces *per se*, and human listeners show improved speech understanding when re-encountering known reverberant environments. We assessed the ability of listeners to perceive speech (Coordinate Response Measure corpus) in noisy and reverberant rooms. We mimicked the acoustic characteristics of real rooms using loudspeakers positioned within an anechoic chamber. Listeners were also exposed to repetitive transcranial stimulation (rTMS) to disrupt the dorsolateral prefrontal cortex activity (DLFPC), a region believed to play a role in statistical learning. Our data suggest listeners rapidly adapt to statistical characteristics of an acoustic environment to improve speech understanding. This ability is impaired when rTMS is applied bilaterally to the DLPFC. The data demonstrate that speech understanding in noise is best when exposed to a room with reverberant characteristics common to human-built environments. Our findings provide evidence for a reverberation "sweet spot" and the presence of brain mechanisms that might have evolved to cope with the acoustic characteristics of listening environments encountered every day.

9:20

4aPP5. Towards audiovisual scene analysis and object formation. Adrian KC Lee (Speech & Hearing Sci., Univ. of Washington, 1417 NE 42nd St., Seattle, WA 98105, akclee@uw.edu), Ross Maddox (Biomedical Eng. and Neurosci., Univ. of Rochester, Rochester, NY), and Jennifer Bizley (UCL Ear Inst., Univ. College London, London, United Kingdom)

The neural mechanisms underpinning auditory scene analysis and object formation have been of intense research interest in the past two decades. Fundamentally, however, we live in a multisensory environment. Even Cherry in his original paper posited that "lip reading" as a way for us to solve the cocktail party problem. Yet, how different aspects of visual cues (e.g., timing, linguistic information) help listeners follow conversation in a complex acoustic scene is still not well understood. In this talk, we present a theoretical framework to study audiovisual scene analysis that has been extrapolated from the unisensory object-based attention literature and posit the following questions: How do we define a multi-modal object? What are the predictions from unisensory object-based attention theory when we apply to the audiovisual domain? What are the conceptual models to test the different neural mechanisms that underpin audiovisual scene analysis? Answering these questions would move us closer to addressing the cocktail party problem in the real-world setting as well as help us create, *de novo*, audiovisual scenes that are more engaging in the augmented/virtual reality world.

9:40

4aPP6. RESTARTing the stabilized auditory image: How transient sampling of binaural information supports the emergence of stable auditory scenes. G. Christopher Stecker (Ctr. for Hearing Res., Boys Town National Res. Hospital, 555 N 30th St., Omaha, NE 68131, cstecker@spatialhearing.org)

The perception of a spatial auditory scene involves the extraction and integration of multiple dynamic and unreliable sensory features ("cues"). Variation in each cue reflects the competing effects of multiple features of the auditory scene—e.g., relevant and irrelevant sound sources. Understanding which cue changes belong together—and to which objects—represents a fundamental challenge to the neural mechanisms of auditory scene perception. RESTART theory suggests a solution: transient, envelope-triggered sampling creates a temporally sparse representation of spatial features. Sparsity minimizes overlap between auditory objects and reduces ambiguity in the face of environmental distortions such as noise and reverberation. Envelope-triggered ("strobed") sampling stabilizes the auditory image by emphasizing the most reliable cues. Here, we review the key evidence supporting RESTART theory, its neurocomputational mechanisms, and prior efforts to model them. [Work supported by US NIH R01-DC016643.]

Contributed Paper

10:00

4aPP7. When and where to listen? Joerg Encke (Macquarie Univ., 16 University Ave., Macquarie, New South Wales 2109, Australia, joerg.encke@mq.edu.au) and David McAlpine (Macquarie Univ., Sydney, New South Wales, Australia)

In natural listening environments, humans hear multiple concurrent sounds arriving from different locations. In order to attend to a single source, concurrent sounds must be separated into individual sound objects, a process that relies on binaural hearing. However, the fluctuating binaural cues that arise when sounds from multiple sources merge at each ear generate fluctuating binaural cues that may be unreliable as to source location. This reliability can be quantified in terms of the interaural coherence (IAC). To overcome unreliable binaural cues, it is suggested that the brain monitors IAC and extracts spatial information from sound energy during epochs where IAC is high. To test this hypothesis, we designed a stimulus allowing us to modulate IAC over time and frequency. Our data indicate that IAC weighting plays a relatively minor role in source separation. Rather, binaural cues only contribute to source localization during rising sound energy. Binaural cues in later epochs of a modulated sound waveform are completely ignored even when IAC is high and spans a wide frequency range. The data do not support weighting based on IAC but, rather, suggest a fast, "non-sluggish" processing of binaural cues to extract information only during the rising energy envelope.

Session 4aSA

Structural Acoustics and Vibration: Acoustic Treatments and Vibration Isolation

Trevor W. Jerome, Cochair
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Pei-Tai Chen, Cochair
Center of Oceanic Engineering, National Taiwan Ocean University, 2 Pei-Ning Rd., Keelung 886, Taiwan

Contributed Papers

8:00

4aSA1. Experimental validation of a multi-material acoustic black hole. Beth Austin (Inst. of Sound and Vib. Res., Univ. of Southampton, Southampton, Hampshire SO17 1BJ, United Kingdom, b.austin@soton.ac.uk), Jordan Cheer (Inst. of Sound and Vib. Res., Univ. of Southampton, Southampton, United Kingdom), and Anil Bastola (Ctr. for Additive Manufacturing, Univ. of Nottingham, Nottingham, United Kingdom)

Geometric acoustic black holes (ABHs) have already been proven as an effective passive vibration control measure in plates and beams. However, the thin geometries inherent to the design raise concerns about the structure's resistance to damage through fatigue. Multi-material ABHs (MM-ABHs) have been proposed as an alternative solution where by the material properties vary along the direction of wave propagation instead of the geometry. This, in theory, produces a change in acoustic impedance without the issues of fatigue seen in the geometric ABHs. Previous work has been performed on the design and modeling of polymer multi-material ABHs. This work discusses practices for experimental measurement and validation of multi-material ABHs produced through additive manufacturing.

8:20

4aSA2. Experimental measurements of stress in an acoustic black hole using a laser Doppler vibrometer. Archie Keys (ISVR, Univ. of Southampton, University Rd., Southampton, Hampshire SO17 1BJ, United Kingdom, a.keys@soton.ac.uk) and Jordan Cheer (ISVR, Univ. of Southampton, Southampton, United Kingdom)

Acoustic Black Holes (ABHs) make use of modifications to a structure to effectively decrease the structural wavespeed, thus increasing the effect of damping material applied in the ABH taper region, resulting in greater vibration attenuation. The most common way in which this is implemented is by gradually reducing the thickness of the structure over a finite interval, to a very thin tip. The focusing effect of the ABH results in high amplitude vibrations occurring in the thin part of the structure, resulting in high stresses and raising significant concerns about fatigue life. This paper presents an experimental assessment of stress in the taper section of an ABH used to terminate a uniform beam, using laser Doppler vibrometer measurements to avoid the mass loading associated with accelerometers or strain gauges. A calculation of stress using Euler–Bernoulli beam theory is then presented, and the validity of this approach is assessed for a thick damping layer applied to a thin structure. A comparison is then made to predictions from a numerical model, in order to validate the results from the experimental measurements.

8:40

4aSA3. On the acoustic wedge absorber design in underwater and its interior pressure computation in a chamber. Pei-Tai Chen (Ctr. of Oceanic Eng., National Taiwan Ocean Univ., 2 Pei-Ning Rd., Keelung, Taiwan 886, Taiwan, ptchenline@gmail.com)

This study presents a design of wedged shape absorber used in underwater in range of high frequency. The analysis of a set of periodical wedges, where each wedge is oriented 90 deg with other wedges. The set of periodical wedges is analyzed using finite element methods as a 3-D virtual impedance tube. The incident wave is on the normal direction of the tube. This 3-D model is able to simplify to a simple 2-D model and a two-point transfer function method is employed to compute reflection coefficient. The ation of frequencies higher than the cut-off plane wave frequency is addressed of that higher duct acoustic modes are excited by the wedge. These higher modes are recognized as stationary non-propagating waves, not contributing reflected waves away from the wedge. An impedance value associated with the reflection coefficient is identified based on plane wave condition. The impedance value is applied to surfaces of a closed space to computed interior pressures.

9:00

4aSA4. Measurement of the transmission loss of a rubber hose. Henry O'Callaghan-Reay (School of Elec. and Mech. Eng., The Univ. of Adelaide, Adelaide, South Australia, Australia), Stephen Moore, James Forrest, Orddom Leav (DST Group, Fishermans Bend, Victoria, Australia), Carl Howard (School of Elec. and Mech. Eng., The Univ. of Adelaide, Adelaide, South Australia 5005, Australia, carl.howard@adelaide.edu.au), Richard Craig, and Morgan Hunter (School of Elec. and Mech. Eng., The Univ. of Adelaide, Adelaide, South Australia, Australia)

Flexible rubber hoses can provide significant attenuation of vibro-acoustic energy in fluid-filled piping systems. The vibro-acoustic performance of rubber hoses can be measured by using a variation of a water-filled impedance tube, using the two-source method, to determine the impedance and transmission loss. This paper presents the test method and results from measurements performed to characterize the vibro-acoustic behavior of a water-filled, 1.5 m length of DN100 fiber reinforced rubber hose, with beaded and flanged terminations, at two pressures. The test results indicated that the rubber hose test specimen achieved a fluid-borne transmission loss of at least 5-10 dB at low frequencies increasing to more than 45 dB at 1 kHz.

9:20

4aSA5. Comparative theoretical and experimental analysis of vibration damping performance in multilayer constrained layer damping structures: Aluminium versus glass fiber reinforced polymer. Gaurav Sharma (Mech. Eng., Defence Inst. of Adv. Technol., C-116 Points Hostel Defence Inst. of Adv. Technol., Pune, Maharashtra 411025, India, sharmag603@gmail.com), Adepu Kumaraswamy (Mech. Eng., Defence Inst. of Adv. Technol., Pune, India), and Sangram Kesari Rath (Polymer Sci. and Technol. Div., Naval Material Res. Lab., DRDO, Mumbai, India)

This study investigates the dynamic behavior of multilayer Constrained Layer Damping (CLD) structures, specifically focusing on constraining layers made of aluminum and Glass Fiber Reinforced Polymer (GRP). Numerical simulations are compared with experimental results using the finite element method (FEM) and the widely used Ross–Kerwin–Ungar (RKU) model to achieve this. Experimental modal analysis is conducted on a vibration shaker to validate the FEA findings. The research reveals valuable insights into the damping performance of aluminum and GRP as constraining layers. Notably, aluminum demonstrates superior damping properties, particularly in the higher frequency range (ultimate loss factor 0.2), while GRP exhibits better-damping characteristics in the lower frequency range (ultimate loss factor 0.13). These results underscore the significance of considering the frequency-dependent behavior of materials when selecting appropriate constraining layers. By shedding light on the dynamic behavior and damping performance of multilayer CLD structures with different constraining layers, this study contributes to the effective mitigation of vibrations in various engineering applications. This endeavor will enhance the understanding and application of constraining layer materials for effective structural vibration control, paving the way for improved engineering solutions.

9:40

4aSA6. Abstract withdrawn.

Session 4aUW

Underwater Acoustics, Acoustical Oceanography, and Physical Acoustics: Acoustics of Extreme Weather Events

Natalia Sidorovskaia, Cochair
Physics, University of Louisiana at Lafayette, PO Box 43680, Lafayette, LA 70504

Roger Waxler, Cochair
Univ. of Mississippi, University, MS 38677

Chair's Introduction—8:55

Invited Papers

9:00

4aUW1. Acoustical effects of extreme weather events. Jie Yang (Appl. Phys. Lab, Univ. of Washington, 1015 NE 40th St., Seattle, WA 98105, jieyang@uw.edu), Eric I. Thorsos, and Mohamed A. Ghanem (Appl. Phys. Lab, Univ. of Washington, Seattle, WA)

The transfer of water from evaporating regions to precipitating regions drives the global water cycle, and understanding its dynamics is essential in determining how weather patterns will respond to changes in global climate. An intensification of the global water cycle has been reported using 50-year sea surface salinity data (Durack and Wijffels, 2010, J. Climate) with the consequence of more frequent extreme weather events. However, data under extreme weather conditions over the ocean are rare and a lot of physics remains to be investigated. As part of the Ocean Climate Stations that NOAA is maintaining, the 1-year data presented here are from Ocean Station Papa (50°N, 145°W), including measurements of wind speed, rain rate, currents, wave height and direction, bubble plume spatial structure and dynamics, and ambient noise for wind speeds up to 25 m/s. This full suite of data enables the study of the statistical properties of bubble plumes, their relation to meteorological parameters, and how to take them into account in ambient noise prediction, especially under severe weather conditions. Knowledge gained through this unique dataset can be critical for understanding air–sea interaction, validating satellite rain products, and predicting the ambient noise field under extreme weather conditions.

9:20

4aUW2. Tropical cyclones' acoustic reconnaissance. Natalia Sidorovskaia (Phys., Univ. of Louisiana at Lafayette, PO BOX 43680, Lafayette, LA 70504, natalia.sidorovskaia@louisiana.edu), Roger Waxler (National Ctr. for Physical Acoust., University, MS), Naomi Mathew (Phys., Univ. of Louisiana at Lafayette, Lafayette, LA), and Claus H. Hetzer (National Ctr. for Physical Acoust., University, MS)

Acoustic reconnaissance of tropical cyclones with underwater gliders is an emerging approach for a comprehensive understanding of the oceanic conditions leading to the formation and intensification of cyclones and for aiding in wind speed forecasting. Shaw *et al.* (1978) showed that the underwater sound pressure level is in the linear relationship to the logarithm of the moderate local wind speed. Wilson and Makris (2006) proposed an acoustic surface source model for hurricane wind speeds. The model requires a parameterization of the winds' contribution. The inversion of the acoustic data leads to estimation of the local wind speeds. The Gulf of Mexico acoustic dataset, collected by the bottom-anchored monitoring system during the passage of storm Barry (2019), is analyzed and correlated with the high-resolution wind speed data generated by the Weather Research and Forecast model. The efficacy of gliders outfitted with acoustic system to provide the real-time reporting of the local wind speeds is also discussed based on the flights' recorded soundscapes. Acoustic glider reconnaissance offers a safe and cost-efficient way of contributing new type data to NOAA and US NAVY weather forecasting models. [Work supported by the Office of the Under Secretary of Defense for Research and Engineering, award# FA9550-21-1-0215.]

Contributed Paper

9:40

4aUW3. Model for underwater noise due to bubble clouds and comparison with observations from Monterey accelerated research system observatory. Ranga P. Raju (Naval Physical and Oceanographic Lab., Thrikkakara, Kochi, Kerala 682021, India, rpraju@outlook.com) and Tarun K. Chandrayadula (Dept. of Ocean Eng., IIT Madras, Chennai, Tamil Nadu, India)

Empirical models for underwater noise spectrum from bubbles suggest a general increase by 5 dB for every doubling of wind speed. However, at wind speeds greater than 10 m/s and frequencies higher than 10 kHz, observations report a decrease in noise with increase in wind. This is due to the formation of bubble clouds, which radiate sound, but also attenuate the intensities because of the air contained within them. The bubble clouds occur in certain shapes called α, β, and γ plumes, which evolve during the various stages of wave-breaking. There are yet no quantitative underwater noise models that combine the nature of these clouds, with their acoustics. This paper builds the model in two steps. First, inputs such as wind speed and wave height are used to approximate the clouds as spheres, with respective number of bubbles of each radius and mean spacing between the clouds. Second, this approximate model is then used to predict the radiated sound. To test the model, predictions are setup for comparison with measured noise during high winds around Monterey, California. To input relevant environment parameters to the model, the work uses measurements of wind-velocities, and wave-heights, measured on-site. For acoustic recordings, the work uses hydrophone from the Monterey Accelerated Research System (MARS) cabled observatory.

Invited Paper

10:00

4aUW4. Decoding the hidden sounds from Tornadoes. Brian Elbing (Oklahoma State Univ., OSU-MAE, 201 General Academic Bldg., Stillwater, OK 74078, elbing@okstate.edu), Aaron S. Alexander, and Real KC (Oklahoma State Univ., Stillwater, OK)

Tornado Alley in the United States produces the largest tornadoes in the world, but the majority of tornado related deaths in the United States occur to the east of Tornado Alley. This is because that region has hilly terrain that limits the use of weather radar, which is responsible for the significant improvements in tornado warnings over the past 40 years. This is a growing problem as Tornado Alley appears to be shifting eastward. An alternative method is to use infrasound (sound at frequencies below human hearing) emitted during the formation and life of a tornado to improve warnings. The fluid mechanism(s) responsible for the sound needs to be identified to properly interpret the received signals. This presentation will report on recent laboratory experiments aimed at investigating potential fluid mechanisms as well as field observations with measurements from tornadoes. [This work was funded, in part, by the Gordon and Betty Moore Foundation under grant GBMF11559 (doi.org/10.37807/GBMF11559).]

Contributed Paper

10:20

4aUW5. Physics-informed learning for modeling infrasound propagation in extreme weather conditions. Christophe Millet (CEA, CEA, DAM, DIF, Arpajon 91297, France, cmillet@lmd.ens.fr), Thi Nguyen Khoa Nguyen, and Mathilde Mougeot (ENS Paris-Saclay, Paris-Saclay, France)

Extreme events in fluid flows are characterized by the coexistence of complex nonlinear dynamics, high intrinsic dimensionality and intermittency, which often results in spatially localized disturbances (turbulence spots, gravity wave breaking). Although many studies have shown that atmospheric ducting of infrasound is sensitive to these disturbances, yet the link between their statistical properties and that of the infrasound wavefield remains an open question, mainly because very little data are available for extreme events. The present work focuses on catastrophic events in climate systems where the amount of data available (typically a few decades) is not sufficient to extrapolate the PDFs. This class of problem involves geophysical fluid flows over climate scales where reanalysis data are a reliable source of information. In contrast to methods that rely on standard models to compute the PDFs from available data, the focus here is on data-driven methods that encode some information about the wave dynamics. The idea behind this approach is to combine two sources of information (reanalysis data and wave theory) using physics-informed neural networks to extrapolate the PDFs. The performance of this approach is illustrated around two types of events that affect infrasound propagation: sudden stratospheric warmings and mountain-induced extreme weathers.

Session 4pAAa

Architectural Acoustics, Psychological and Physiological Acoustics, and Signal Processing in Acoustics: Three-Dimensional Sound Display and Analysis for Virtual Auditory Immersive Environments II

Ning Xiang, Cochair
Graduate Program in Architectural Acoustics, Rensselaer Polytechnic Institute, 110 8th Street, Troy, NY 12180

Lamberto Tronchin, Cochair
Architecture, University of Bologna, Via G.F. Barbieri 76, Bologna, I 40129, Italy

Angelo Farina, Cochair
Univ. of Parma, Parma, Italy

Invited Papers

1:00

4pAAa1. Transitioning concert hall soundscapes to vehicle interiors based on psychoacoustic evaluations. Jin Yong Jeon (Architectural Eng., Hanyang Univ., Seoul, Korea (the Republic of), jyjeon@hanyang.ac.kr), Haram Lee, Hyowon Yoon, Beta Bayu Santika (Architectural Eng., Hanyang Univ., Seoul, Korea (the Republic of)), Dongchul Park, and Juin Kim (Automotive, Res. & Development Div., Hyundai Motor Group, Hwaseong, Korea (the Republic of))

In this study, in line with the paradigm shift in the design of interior spaces for future autonomous vehicles, the vehicle interior spaces were categorized according to their suitability for music appreciation and a design for an optimal soundscape design to that of a concert hall was proposed. Based on the sound field investigation conducted with 360-deg camera video and binaural audio recorded inside the vehicle, audibilized computer models were created in a virtual reality environment for comparison with conventional rectangular halls. These models' auralization audio files were subject to a psychoacoustic evaluation experiment of the soundscape. Evaluation indicators were selected based on the collected data, and through a comparative analysis of these indicators, parameters for transitioning the soundscape of a concert hall into a vehicle interior were proposed.

1:20

4pAAa2. Improving the acoustic contrast control performance in car cabins. Shuping Wang (Key Lab. of Modern Acoust. and Inst. of Acoust., Nanjing Univ., No. 22 Hankou Rd., Nanjing, Jiangsu 210093, China, shuping.wang@nju.edu.cn), Qing Xu (Key Lab. of Modern Acoust. and Inst. of Acoust., Nanjing Univ., Nanjing, Jiangsu, China), Qiaoxi Zhu (Univ. of Technol. Sydney, Botany, New South Wales, Australia), Jiancheng Tao (Key Lab. of Modern Acoust. and Inst. of Acoust., Nanjing Univ., Nanjing, Jiangsu, China), and Xiaojun Qiu (Key Lab. of Modern Acoust. and Inst. of Acoust., Nanjing Univ., Sydney, New South Wales, Australia)

The car cabin is an enclosure where personal audio is very important because different occupants may have individual listening requirements. Potential approaches to improve the acoustic contrast control performance in enclosures are discussed in this paper, including the configuration of loudspeakers and the sound absorption of sidewalls. The effect of the number of loudspeakers on the acoustic contrast control performance is first investigated, and the locations of a fixed number of loudspeakers are then optimized to maximize the acoustic contrast achieved with the system. The sound absorption of sidewalls also has an effect on the acoustic contrast control performance and it is discussed in detail. The simulations are based on the acoustic transfer functions obtained with a finite element model of a car cabin. The findings in this paper may serve as a guide to future designs of personal audio systems in car cabins.

Contributed Papers

1:40

4pAAa3. Target sound extraction on reverberant mixture. Dayun Choi (Elec. Eng., Korea Adv. Inst. of Sci. and Technol. (KAIST), 291 Daehak-ro, Yuseong-gu, Daejeon, Korea (the Republic of), cdy3773@kaist.ac.kr) and Jung-Woo Choi (Elec. Eng., Korea Adv. Inst. of Sci. and Technol. (KAIST), Daejeon, Korea (the Republic of))

Target sound extraction is a task to extract only a desired sound signal from a mixture of different sounds, using a clue given by a target class label or a target signal similar to the desired sound. Currently, available network architectures for this task are designed to handle only dry sounds. In this work, we introduce a transformer-based target sound extraction model that can extract reverberant sounds. To separate reverberant sound mixtures, we begin with the Dense Frequency-Time Attentive Network (DeFT-AN) architecture developed for speech enhancement tasks, which generates the complex short-time Fourier transform (STFT) mask of clean speech from a noisy reverberant mixture to suppress noises. To make DeFT-AN compatible with the target sound extraction task, we modify its architecture such that the embedding vector for the target class label can be fused in the middle of sequentially connected DeFT-A blocks constituting DeFT-AN. We

demonstrate that the transformer-based speech enhancement model can be successfully converted into a target sound extraction model and outperforms state-of-the-art extraction models in the test carried out with reverberant mixtures.

2:00

4pAAa4. Single-value frequency-average measures of early decay time and clarity index to predict reverberance and clarity. Fernando M. del Solar Dorrego (Graduate Program in Acoust., The Penn State Univ., University Park, PA 16802, fsolar@gmail.com) and Michelle C. Vigeant (Acoust., Penn State Univ., University Park, PA)

The most accepted acoustical parameters for the assessment of concert hall acoustics are summarized in the Annex A of the ISO 3382-1 standard, along with recommendations that are important for the practical application of these parameters. These recommendations have their origin in the practical experience of acoustical consultants and a few scientific studies, and further research is needed to properly validate them. In particular, more information is needed on how to average the octave-band values of acoustical parameters to obtain single-value frequency-average measures (SVFA) that are better correlated with the associated perceptual aspects. In this study, SVFA measures of early decay time (EDT) and clarity index (C80) were obtained by means of a listening test in which 24 subjects rated a set of stimuli in terms of reverberance and clarity. The participants listened to auralizations of orchestral music convolved with measured spatial room impulse responses in an anechoic chamber using third-order Ambisonic reproduction. The results of the listening test indicate that the mid-frequency, 500 and 1000 Hz, octave-band average of EDT and the average of the 4000and the 8000 Hz octave bands of C80, have the highest sensitivity in the prediction of reverberance and musical clarity, respectively.

2:20

4pAAa5. Auralization as a tool for design and subjective analysis. Ana M. Jaramillo (AFMG Services North America, 8717 Humboldt Ave. N, Brooklyn Park, MN 55444, ana.jaramillo@afmg.eu), Gustavo Basso (IPEAL, Facultad de Artes, Universidad Nacional de La Plata, La Plata, Argentina), and Bruce C. Olson (AFMG Services North America, Brooklyn Park, MN)

During the remodel of Teatro Colon in Argentina in the period 2006/2010, acousticians performed acoustic measurements at ten different stages of deconstruction and then repeated them during construction. Each stage allowed them to verify that the internationally renowned acoustic quality of the venue was not going to be lost with the use of new materials, especially the new upholstery of the seating. This study re-creates some of those stages using EASE acoustic modeling software, explores how auralizations of a model compare to auralizations of measured IRs, and how this type of prediction can be used as a design tool.

2:40

4pAAa6. Use of Virtual Reality and auralization as a training tool for acoustic practitioners. Matthew Ottley (Marshall Day Acoust., C14 372 Wattle St., Ultimo, New South Wales 2007, Australia, mottley@marshallday.com) and Yuxiao Chen (Marshall Day Acoust., Ultimo, New South Wales, Australia)

Room acoustic qualities for performance spaces are described, including in ISO 3382-1:2009, in objective metrics such as reverberance, clarity (C80), early lateral energy fraction (JLF) or late lateral sound level (LJ). For acoustic practitioners, an understanding of these parameters and their subjective perception in venues is necessary to describe existing acoustic conditions and carry out acoustic designs of venues. Gaining practical experience of venues to understand the changes in acoustic qualities can be difficult for various reasons. Virtual Reality (VR) technology, combined with auralization from 3-D room impulse responses, generated by computer acoustic modeling of virtual spaces, makes it possible to generate virtual listening experiences to demonstrate room acoustic qualities. This presentation describes the development of a VR training video demonstrating various room acoustic qualities in a virtual concert hall. The video was intended for training of acoustic consultants to recognize and differentiate changes in room acoustic parameters for performance spaces. The VR video presented a number of scenarios where each showed a primary change in one of the qualities, through targeted manipulation of the virtual space. Following deployment of the training video, feedback was taken from users to assess the usefulness of the format for training purposes.

3:00–3:20 Break

3:20

4pAAa7. Effect of reference value and ensemble size on perception of reverberation in simulated listening environments. Christopher M. Jasinski (Mech., Aerosp., and Acoust. Eng., Univ. of Hartford, 200 Bloomfield Ave. West Hartford, CT 06117, jasinski@hartford.edu) and Robert D. Celmer (Mech., Aerosp., and Acoust. Eng., Univ. of Hartford, West Hartford, CT)

Reverberation time is an important metric characterizing one of the key differences between performance venues caused by various furnishing, materials, and geometry within such spaces. To justify alterations to a venue, architects and acousticians refer to the just-noticeable-difference (JND) of reverberation time, cited in the ISO 3382-1 room acoustics standard as 5% of the reference value. This JND was determined experimentally using impulsive noise with varying reverberation. In real performance venues, overlapping and simultaneous sounds are more common than isolated impulses. As such, the JND noted in literature may not provide a complete picture of perceived reverberation for persons listening to live performed music. For this study, multiple test methods were evaluated by using different reference reverberation times as well as using a variety of instrumentation and genres for musical signals. Results were generated using test participants who were asked to listen to multiple virtual soundfields and provide subjective evaluation. Reverberant soundfields were recreated in an anechoic listening environment and auralizations were produced using ODEON room acoustics software. Results indicated statistically significant mean differences in perceived reverberation across instrumentation subsets, genre subsets, as well as through different testing configurations. [Work supported by The Paul S. Veneklasen Research Foundation.]

3:40

4pAAa8. The localization of virtual source reproduced by the crosstalk cancellation system under different direct-to-reverberation energy ratios conditions. Wei Tan (School of Phys., South China Univeristy of Technol., Wushan Rd., Tianhe District, Guangzhou, Guangzhou, Guangdong 510630, China, 1044458749@qq.com), Dan Rao, Yewei Wang, and Guangzheng Yu (School of Phys., South China Univeristy of Technol., Guangzhou, Guangdong, China)

This study explores the localization of virtual sources reproduced by the crosstalk cancellation (CTC) system under reverberation environments with different direct-to-reverberation energy ratios (DRRs). Binaural room impulse responses in rectangle rooms are simulated by high-order image source method. Two independent key variables, the reverberation time (RT) and the distance of the sound source to the listener, are employed to manipulate DRR through simulation. Subsequently, a subjective localization experiment based on the CTC system is virtually conducted using earphone reproduction. The results show that the azimuth localization in horizontal plane is almost impervious by RT but is subjected to the change of the source distance. Specifically, as the distance of sound source increases, the perceived azimuth of lateral virtual sources exhibits a bias towards the front, accompanied by larger variances. The results indicate that reflections with small delays have a more detrimental effect on sound localization compared to the late reverberation with large intensity, which may be associated with the inhibition of late reflections based on the precedence effect. Additionally, the localization data are analyzed and explained using binaural cues, including the interaural cross-correlation coefficient, the interaural time difference (ITD), and interaural level difference (ILD).

4:00

4pAAa9. A method of timbral equalization for Ambisonics reproduction with irregular loudspeaker configuration. Shanwen Du (School of Phys. and Optoelectronics, South China Univ. of Technol., Rm. 301, Bldg. 18, No. 381, Wushan, Guangzhou 510641, China, shanwendu1995@163.com) and Bosun Xie (School of Phys. and Optoelectronics, South China Univ. of Technol., Guangzhou, China)

Ambisonics is a spatial sound reproduction technique based on spatial harmonics decomposition and each order approximation of target sound field. Given a spatial region, Ambisonics is able to reconstruct target sound field accurately up to a frequency limit imposed by Shannon–Nyquist spatial sampling theorem. Above that limit, spatial aliasing in reconstructed sound field occurs, and therefore, timbre equalization in loudspeaker signals is needed to reduce perceived timbre coloration in reproduction. Power equalization is a common and effective method in which the overall power of loudspeaker signals are equalized to be independent of target source direction. The power equalization can be easily implemented in the Ambisonics reproduction with regular rather than loudspeaker configuration. In this study, a method of power equalization for Ambisonics reproduction with irregular loudspeaker configuration is proposed. The fluctuation of overall power of loudspeaker signals with target source direction is controlled within 3 dB by appropriate choice of the decoding coefficients of higher-order spatial harmonics in Ambisonics reproduction. This method is applicable to Ambisonics signals obtained by simulation or microphone array recording. Psychoacoustic experiments indicate that the current method reduces the timbral coloration to some extent while keeps the localization performance in reproduction.

4:20

4pAAa10. A novel method for optimizing in-car rolling noise levels within a railway tunnel environment. Luke Zoontjens (SLR Consulting Australia, Level 1, 500 Hay St., Subiaco, Western Australia 6008, Australia, lzoontjens@slrconsulting.com) and Jelena Sostaric (Public Transport Authority Western Australia, Perth, Western Australia, Australia)

In-car noise levels within trains rolling on surface track are usually controlled by noise and vibration which is generated at the wheel–rail interface and then transmitted through the chassis and floor of the train cabin. In a railway tunnel environment, higher noises levels result from the build up of reverberant noise, and if left untreated this increased noise can affect passenger comfort and make it harder to listen or converse. In extreme cases, it can lead to adverse health effects: travellers may complain and avoid using the rail service as a result. Accordingly, the acoustic design of railway tunnel environments generally includes consideration of various treatments to reduce noise ingress into vehicle interiors, particularly given treatments to the vehicles themselves may not be cost effective. Modeling of in-car noise levels within tunnels has traditionally used the Sabine equation as part of the design process. Because the Sabine equation was never intended for this application given the geometry and frequencies involved, an improved method which recognized the near field effects was developed in order to provide confidence in the acoustic design. In this presentation, this novel method for modeling in-car rolling noise levels and optimizing sound absorptive treatments within a tunnel is discussed. Modeling results and measurements from a recent major rail project are used to demonstrate the effectiveness and improvements in outcomes from traditional approaches.

4:40

4pAAa11. Room acoustic computer modeling—A case study to compare industry standard software with measured results. Guy Hopkins (Acoust., WSP Pty Ltd, 4 Currawong Rd., Sydney, New South Wales 2082, Australia, guy.hopkins@wsp.com) and Chris Field (Acoust., WSP Pty Ltd, Sydney, New South Wales, Australia)

This paper extends a previous publication presented at the Inter-Noise 2005 conference [1] and investigates the predictive performance of some industry standard room acoustic software packages compared with measurements. The case study specifically examines the Twyford Theatre, comparing predictions of room acoustic parameters obtained from ODEON, EASE, and Treble software with measurements. To conduct the study, impulse response measurements were performed in the theater using the swept sine method. Architectural drawings and on-site inspection measurements were utilized to construct ODEON, EASE, and Treble models for prediction purposes. The findings of this study provide insights into the predictive performance of these industry standard room acoustic software packages with comparison to measured results. C. Field and S. Shimada, *Acoustic Computer Modelling: A Case Study to Compare Predictions by CATT and Odeon with Measured Results*, Inter-Noise 2005.

5:00

4pAAa12. Design and integration of acoustic treatment to the Former Stock Exchange—Cathedral Room. Frank Butera (Acoust., Arup Australia Pty Ltd, Level 2, 699 Collins St., Docklands, Victoria 3008, Australia, frank.butera@arup.com), Helen Searle, and Jim White (Acoust., Arup Australia Pty Ltd, Docklands, Victoria, Australia)

The Gothic Bank Cathedral Room within the Former Stock Exchange located in Melbourne is architecturally significant and largely original displaying ornate features for a historical public business arena. The Cathedral Room was the main trading floor for the Stock Exchange and was designed to represent the interior of a church, highlighted by the vaulted roof, columns, and the stained-glass windows. Through the guidance of Heritage Council Victoria, the Cathedral Room was granted permission to be converted into a commercial dining room. This paper details the methodology adopted to reality capture the interiors of the Cathedral Room by deploying LiDAR laser scanning, to develop a room acoustics model, and to investigate mitigation options. Furthermore, it explores the link between applying appropriate acoustic targets, guiding stakeholders, and delivering the project, which is essential to operate a successful commercial business to meet the expectations of their cliental. Arup developed an Odeon model, which allowed for the placement of speech sources and receivers and with integrated mitigation options. The modeling was presented to stakeholders in Arup's SoundLab, an ambisonic sound studio, allowing the stakeholders to listen to the untreated space and compare it to various levels of mitigation prior to progressing to detail design phase.

5:20

4pAAa13. Efficient deep learning-based prediction of acoustic parameters in atrium spaces using BIM files. Prachee Priyadarshinee (Sci., Mathematics and Technol., Singapore Univ. of Technol. and Design, #02-07, 53 Changi South Ave. 1, Singapore 485996, prachee@sutd.edu.sg), Jer-Ming Chen, and Balamurali BT (Sci., Mathematics and Technol., Singapore Univ. of Technol. and Design, Singapore)

Acoustic simulation tools, although common in the acoustic evaluation of buildings with large atrium space, are costly both in terms of time and resources. This study aims to develop an efficient prediction tool for Reverberation Time (RT) and Sound Pressure Level (SPL) within the atriums using deep learning, with Building Information Modelling (BIM) files as the only input. Initially, a 3-D acoustic simulation model was benchmarked against experimental measurements of RT for an existing building's atrium, demonstrating an agreement within 15%. Subsequently, BIM files of 60 buildings were used to simulate RT and SPL at various listener locations within their atrium spaces. The BIM files provided essential geometry information, including atrium shape, dimensions, door placements, floor plans, material properties, and sound sources. This dataset was then utilized to train deep learning algorithms, enabling rapid and convenient predictions of RT and SPL for any new building's atrium space. The proposed prediction model serves as a valuable tool for architectural planning and noise regulation during the early design stages. By relying solely on basic building information from the BIM file, this tool obviates the need for time-consuming and computationally expensive simulation software typically used for acoustic evaluations in large atrium spaces.

Session 4pAAb

Architectural Acoustics and Noise: Soundwalking in Sydney

David S. Woolworth, Cochair
Roland, Woolworth & Associates, 356 CR 102, Oxford, MS 38655

Jin Yong Jeon, Cochair
Architectural Engineering, Hanyang University, 222, Wangsimni-ro, Seongdong-gu, Seoul, 04763, Korea (the Republic of)

Invited Paper

1:40

4pAAb1. Introduction to Soundwalking—An important part of the soundscape method. David S. Woolworth (Roland, Woolworth & Assoc., 356 CR 102, Oxford, MS 38655, dave@oxfordacoustics.com), Bennett Brooks (Brooks Acosutics Corp, Pompano Beach, FL), and Brigitte Schulte-Fortkamp (HEAD Genuit Foundation, Herzogenrath, Germany)

The soundscape method provides a holistic understanding of the perception by local stakeholders of an acoustical environment. The soundwalk is an important tool of the soundscape technique and is used to measure these perceptions. The core of the soundwalk technique is that local experts experience and evaluate, in context, the acoustical environment where they live, work, and play. The collected soundwalk data can assist in the new design, or an intervention on existing conditions, for an improved quality of life in our communities. This presentation will describe the usefulness of the soundwalk technique and the results which one can obtain. Topics include, planning a soundwalk, ISO Standard 12913 Part 2 procedures, conducting a soundwalk, who are local experts, gathering participants, interaction of participants and problem solving (sound levels and subjective judgments), factors in judgment (sound quality and psychoacoustics), health and healthy soundscapes, interpretation of data, and applications such as ordinances and interventions. Examples of collected soundwalk data, including real-world outcomes are discussed. A soundwalk in Sydney will be conducted following the presentation (about an hour), everyone is invited to participate. After the walk, participants are encouraged to engage in a discussion of the experience, provide additional feedback, and ask questions.

3:40–4:10

Session 4pAB

Animal Bioacoustics: Marine Bioacoustics in the West Pacific

Shane Guan, Chair
BOEM, 45600 Woodland Road, Sterling, VA 20166

Chair's Introduction—12:55

Invited Papers

1:00

4pAB1. Evolution of acoustic methods for assessing and managing exposure of gray whales to sound pulses from seismic surveys off Sakhalin Island, Russian Far East. Roberto Racca (JASCO Appl. Sci., Victoria, BC, Canada) and David E. Hannay (JASCO Appl. Sci., Victoria, BC, Canada, david.hannay@jasco.com)

An effective program of estimation and mitigation of sound exposure from airgun array pulses is of vital importance when an endangered whale population forages in coastal waters near oil and gas production fields, where geophysical exploration campaigns regularly take place. Starting in 2001, increasingly sophisticated acoustic measurement and modeling techniques have been used for the environmental management of airgun array seismic surveys adjacent to the nearshore feeding grounds of the gray whale (*Eschrichtius robustus*) off the NE coast of Sakhalin Island, Russia. These monitoring data are also used in combination with visual observations for the analysis of the effects of sound exposure on whale behavior, distribution, and energetics. This talk follows two decades of evolution of the acoustic monitoring technology used in the Sakhalin studies, from rudimentary analog sonobuoys with limited bandwidth VHF telemetry, to digital stations capable of onboard signal processing and relaying of data via satellite to anywhere on the globe. In parallel, it traces the advancement of modeling techniques capable to fill the gaps between sparse measurement locations to yield high resolution estimates of the sound field from seismic sources operating concurrently at different, moving locations and in various regimes of ramp-up, production, or mitigation pulses.

1:20

4pAB2. Detections of biological sounds and mapping of acoustic presence of marine organisms. Tomonari Akamatsu (Ocean Policy Res. Inst., The Sasakawa Peace Foundation, 1-15-16 Toranomon, Tokyo, Minato-ku 105-8524, Japan, akamatsu.tom@gmail.com)

Many species of marine organisms have been known to produce sounds, which represents species or family characteristics. Passive acoustic monitoring is widely applied for the monitoring of presence and distribution of marine mammals since the species-specific sound characteristics are well identified. In the present study, sounds of damselfish, silver croaker as well as biosonar sounds of high frequency cetacean and impulse sounds potentially produced by snapping shrimp were extracted from towed and fixed passive acoustic monitoring systems off Japanese waters. Results show the scattered spots of damsel fish in a coral reef and dense aggregation of silver croakers in specific area. Acoustic distribution of silver croakers and finless porpoises was similar each other that suggested co-existence of prey and predator in highly productive area due to river discharge. Pulse sounds distribution was different from others. Towed passive acoustic monitoring provided snap shot acoustic distributions of multiple species. Day-by-day change of acoustic distribution recorded by fixed passive acoustic monitoring systems could be used for the adaptive management of marine organisms.

Contributed Paper

1:40

4pAB3. Detecting free-ranging river dolphin using deep learning based algorithms. Gang Qiao (College of Underwater Acoust. Eng., Harbin Eng. Univ., Harbin, China), Suleman Mazhar (Acoust. Sci. and Technol. Lab., Harbin Eng. University, Harbin 150001, China, suleman.mazhar@fulbrightmail.org), Mamoon Masud (BiSMiL Lab, Comput. Sci. Dept., Information Technol. Univ. of the Punjab, Houston, TX), Abubakr Muhammad (LUMS, Lahore, Punjab, Pakistan), Masroor Hussain (FCSE, GIK Inst., Topi, Khyber Pakhtunkhwa, Pakistan), Uzma Khan, and Masood Arshad (WWF-P, Lahore, Pakistan)

Indus river dolphin is an endangered freshwater river dolphin and is included in IUCN red list of endangered species. Canal-strandings is one of the major issues that results in high mortality of the dolphin because the dolphins that are stranded into canals during flood season cannot return to the river during low-water season. As the dolphin uses high frequency echolocation clicks for navigation in muddy river waters, passive acoustic monitoring can be used for detection and localization of the dolphin and can assist in conservation efforts. In this context, we compare performance of variants of deep learning (LSTM) based event detection algorithms for dolphin click detection with conventional algorithms (such as the Teager Energy based click detection and Envelope Derivative Operator based click detection) and analyze the possible scenarios where machine learning based methods may assist in better performance by automatic detection of reflected clicks.

Invited Paper

2:00

4pAB4. Review of PAM studies in the Coastal Waters West of Taiwan during 2013–2022. Wei-Chun Hu (Eng. Sci. and Ocean Eng., National Taiwan Univ., Taipei, Taiwan), Chi-Fang Chen (Eng. Sci. and Ocean Eng., National Taiwan Univ., No. 1 Roosevelt Rd., Section 4, Taipei 106, Taiwan, chifang@ntu.edu.tw), and Pei-Yi P. Lee (Ctr. of Excellence for Ocean Eng., Taiwan Ocean Univ., Keelung, Taiwan)

Taiwan is making efforts to transform its energy system in response to the global trend of climate change, vigorously developing renewable energy to achieve the goal of "Net Zero" emissions by 2050. However, the development of offshore wind farms may have an impact on the ecological environment of the vast western waters. The coastal waters are heavily used for human activities, and the adjacent protected reefs and estuary areas have abundant biodiversity, which may be greatly affected and require priority investigation. In recent years, passive underwater acoustic monitoring has been widely used. In addition to sound levels, it is also used for biodiversity surveys and ocean environmental impact assessments. This paper provides a review of passive acoustic monitoring studies conducted along the western coast of Taiwan in the past decade. The study period encompasses the pre-construction, construction, and operational phases of the first offshore wind turbine demonstration project, as well as the current underwater noise characteristics during the construction phase of offshore wind farms. During the large-scale development of offshore wind farms, passive acoustic monitoring is a great way to maintain a good and sustainable marine environment.

Contributed Papers

2:20

4pAB5. Study of the underwater soundscape in coastal waters of Hsinta Power Plant. Yin-Ying Fang (System Eng. Section, Kang-Lung Project Office, Information and Commun. Res. Div., National Chung-Shan Inst. of Sci. and Technol., P. O. Box 90008-16-11 Lung-Tan, Taoyuan 32599, Taiwan, ininverysmart@gmail.com), Chen-You Shih (Underwater Technol. Section, Information and Commun. Res. Div., National Chung-Shan Inst. of Sci. and Technol., Kaohsiung, Taiwan), Ming-Ru Zhong (Dept. of Appl. Artificial Intelligence, Ming Chuan Univ., Taoyuan, Taiwan), Chia-Yu Liu (Configuration Information Management Section, Naval Shipbuilding Development Ctr., Kaohsiung, Taiwan), and Ting-Jung Kuo (Dept. of Appl. Artificial Intelligence, Ming Chuan Univ., Taoyuan, Taiwan)

This study investigates the acoustic environment surrounding the Hsinta power plant, situated in the southwest coastal waters of Taiwan, in proximity to Xingda Port, Yongan Port, Yongan Liquefied Natural Gas Plant, and an artificial reef area via passive acoustics monitoring (PAM) and Automatic Identification System (AIS). Shipping noise (including 45% fishing boats, 20.8% large ships, and 6% tug) generated by inbound and outbound ships significantly affects areas within a range of 2 to 10 km from the study site. Notably, mechanical signals at 60 and 120 Hz, not originating from PAM, have been observed, likely attributed to the Hsinta power plant. Moreover, a distinct series of periodic biological signals within the frequency range of 500 to 800 Hz is consistently recorded near the artificial reef area between 8 p.m. and 10 p.m. Assessing the impact of local ship noise by comparing ambient noise levels below the 100 Hz band before and after the coal shipping seasons (April to September), revealing a difference of 5 to 10 dB. These findings provide valuable insights into the acoustic characteristics of the study area, emphasizing the importance of understanding soundscape dynamics amidst industrial activities and maritime traffic. [Work supported in part by the National Science and Technology Council, Taiwan under Grant NSTC 112-2222-E-130-002-.]

2:40–3:00 Break

3:00

4pAB6. Underwater acoustic surveys in the eastern Clarion Clipperton Zone. Iain M. Parnum (Ctr. for Marine Sci. and Technol., Curtin Univ., Kent St., Bentley, Western Australia 6102, Australia, i.parnum@curtin.edu.au), Adrian Flynn, David M. Donnelly (Fathom Pacific Pty Ltd, Mordialloc, Victoria, Australia), and Michael Clarke (The Metals Co., Vancouver, BC, Canada)

A drifting, underwater sound recording system was developed and deployed for acoustic surveys of vocalizing marine mammals in the eastern Clarion Clipperton Zone (CCZ), Northern Tropical Pacific. The system was designed to position an Ocean Instruments Sound Trap 500HF recorder and hydrophone at a depth of 450 m. A depressor vane was inserted halfway along the drop line, with shock cord, to minimize hydrodynamic noise. The surface line included a Spotter V2 buoy that provided real-time tracking of the location of the surface line. The recording system was deployed ten times in the CCZ, between November 2020 and October 2021. Deployment length was between two and seven days. All deployments were set to continuously record underwater sound with a sample frequency of 288 kHz. Acoustic data were analyzed using the CHORUS software to manually review recordings, and an FM signal detector (in MATLAB) was used to detect potential biological sounds. A variety of marine mammals were detected, including *Delphinidae* species, *Physeter macrocephalus* (sperm whales); and clicks likely to be from *Ziphiidae* species (beaked whales). Beaked whales have been previously visually sighted in the CCZ, but to our knowledge these might be the first acoustic recordings of them in the CCZ.

Invited Papers

3:20

4pAB7. Whales, wind, and whale watching—Uncovering the secret soundscape of Geographe Bay. Capri D. Jolliffe (Res. Board, Geographe Marine Res., Dunsborough, Western Australia, Australia, capri@marineresearch.org.au), Craig McPherson (JASCO Appl. Sci., Capalaba, Queensland, Australia), and Ian Wiese (Res. Board, Geographe Marine Res., Geographe Bay, Western Australia, Australia)

Quantifying the acoustic soundscape of an area is an important precursor to understanding the significance of that area from an environmental perspective, as well as providing useful environmental context to inform environmental management. Underwater soundscapes provide important contextual information about the use of that habitat by marine fauna species as well as some of the anthropogenic stressors that may be present. Multi-year acoustic data collected from an underwater acoustic recorder near Cape Naturalist in Geographe Bay was used to characterize the underwater soundscape and complete an initial assessment of cetacean presence. Ambient noise conditions within the Bay were characterized, providing an interim baseline of the ambient noise conditions within the Bay, identifying the biological and anthropogenic contributors to the soundscape. This characterization provided a better understanding of species presence within the Bay and general trends in occurrence, and has also been used to investigate the scale and extent of existing anthropogenic pressures. The information from the monitoring program has been applied to better understand the value of the Geographe Bay area to different species, while at the same time setting a benchmark for ongoing monitoring efforts and providing inputs into detection range modeling to assist in enhancing program design.

3:40

4pAB8. Offshore industry and whales: Pygmy blue whale acoustic behavior and movement patterns in the Otway Basin. Craig McPherson (JASCO Appl. Sci. (Australia), Capalaba, Queensland, Australia, craig.mcpherson@jasco.com), Capri Jolliffe (Blue Whale Project Oz, Perth, Western Australia, Australia), Matthew W. Koessler (JASCO Appl. Sci., Victoria, BC, Canada), and Bruce S. Martin (JASCO Appl. Sci., Dartmouth, NS, Canada)

The offshore Otway region of Australia is designated as a Biologically Important Area for foraging for blue whales. Understanding the acoustic behavior of blue whales in these foraging grounds will enable an improved understanding of the residency time of individual animals and facilitate improved management of the population. JASCO deployed a directional AMAR G4 for part of the 2020 pygmy blue whale foraging season. The recorded vocalizations, including song and D-calls, have been analyzed in detail to determine bearings, received level, and estimated ranges. Bearings to detected vocalizations were determined using a maximum-likelihood-estimator, with a fluid only propagation model corrected for the layered elastic seafloor used to determine ranges. An analysis of the distribution of different vocalizations across the listening range of the AMAR over time has been conducted and correlated with AIS information from shipping traffic. The results have been used to provide a brief insight into the acoustic behavior and movement patterns of individual pygmy blue whales within the region, furthering our understanding of spatial habitat use in association with commercial shipping lanes and oil and gas operations.

Contributed Paper

4:00

4pAB9. If these walls could talk: Investigating bottlenose dolphin habitat use in fiord ecosystems. Leah M. Crowe (Marine Sci., Univ. of Otago, 46 Allandale Rd., St. Clair/Dunedin, Otago 9012, New Zealand, leah.crowe@postgrad.otago.ac.nz), Will Rayment (Marine Sci., Univ. of Otago, Dunedin, New Zealand), and Jenni A. Stanley (Univ. of Waikato, Hamilton, New Zealand)

Effective management of protected species requires a comprehensive understanding of their ecology. Within the Te Moana o Atawhenua-Fiordland Marine Area (FMA), Aotearoa-New Zealand, two of the four recognized bottlenose dolphin (*Tursiops truncatus*) sub-populations are considered to exclusively inhabit discrete fiord systems. Opportunistic sightings outside the Patea-Doubtful and Tamatea-Dusky fiord complexes, however, suggest they occupy a larger space than currently recognized. To investigate the presence of bottlenose dolphins in five neighboring fiord systems, passive acoustic monitoring (PAM) was conducted from February 2022 to November 2023. PAM effort included two instrumentation approaches: F-PODs to continuously detect click trains of odontocetes, and SoundTraps to provide broad spectrum recordings (15 of every 30 min, sampling rate: 96 kHz). This study discusses the trade-offs between approaches in terms of cost, recording duration, and data collected. In addition, photo-identification of bottlenose dolphins in these neighboring fiords was used to identify the sub-population of individuals. The results demonstrate that bottlenose dolphins are regularly using fiords that are not formally considered part of their range within the FMA. A greater understanding of the spatial ecology of Fiordland bottlenose dolphin sub-populations necessitates consideration of threats and resources both inside and outside of their namesake fiord complexes.

Session 4pBA

Biomedical Acoustics and Physical Acoustics: Cavitation Therapies for Cancer Treatment II

Mike Averkiou, Cochair
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Tatiana Khokhlova, Cochair
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Contributed Papers

1:00

4pBA1. Real-time cavitation monitoring based on the entropic analysis of ultrasound RF signals. Juan Tu (Acoust., Nanjing Univ., 22 Hankou Rd., Nanjing 210093, China, juantu@nju.edu.cn), Renjie Song, Xiasheng Guo, and Dong Zhang (Acoust., Nanjing Univ., Nanjing, China)

The violent inertial cavitation activity generated in high intensity focused ultrasound (HIFU) treatment has a risk of damaging healthy tissues around the target area, so that it is necessary to monitor the degree of cavitation during the treatment. Entropy, as a non-model-based statistical parameter, can effectively characterize the effect of cavitation on the structure of scatters within tissues. Thus, a real-time cavitation monitoring system was established based on the entropy analysis of ultrasound radio frequency (RF) signals, and the degree of cavitation was evaluated through entropy images. Based on the two-dimensional average filtering method, we achieved real-time monitoring of cavitation through the reconstruction of entropy image and the binarization of cavitation regions, breaking the inherent limitation of current systems that can only perform cavitation monitoring between HIFU pulses. The results showed that the entropy image retained more frequency components, which could determine the time and location of cavitation earlier and display cavitation areas. Moreover, as the acoustic pressure increased, the advance of cavitation indication in the entropy images became earlier. This method might provide a feasible evaluation scheme for improving real-time cavitation monitoring during HIFU treatment and was expected to become an effective reference index for imaging in HIFU surgery.

1:20

4pBA2. Active Doppler monitoring of *de novo* cavitation induced by a dual-mode pulsed high-intensity focused ultrasound array. Randall P. Williams (Dept. of Medicine, Univ. of Washington, 10100 Burnet Rd., Bldg. 160, Rm. 1.108, Austin, TX 78758, r.williams.06@gmail.com), Minho Song (Dept. of Mech. Eng., Univ. of Washington, Seattle, WA), Yak-Nam Wang, Stephanie Totten (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), and Tatiana Khokhlova (Dept. of Medicine, Univ. of Washington, Seattle, WA)

Pulsed high-intensity focused ultrasound (pHIFU) is capable of inducing inertial cavitation in tissues without the need for microbubbles or other ultrasound contrast agents, thereby permeabilizing the tissue to enhance passive diffusion of systemically administered therapeutic agents. Here, we report on the use of ultrafast, synchronous power Doppler imaging to spatially map cavitation activity and resulting tissue disruption in *ex vivo* and *in vivo* porcine tissues using a dual-mode imaging-therapy linear pHIFU array. The array (64 elements, aperture 51.2 mm $\times$ 14 mm, frequency 1.0 MHz, 40% bandwidth) was driven by a power-enhanced Verasonics V1 system to deliver a series of 1 ms pulses at a 1% duty cycle while electronically scanning the pHIFU focus over an azimuthal range of $\pm$1 cm. Plane wave Doppler ensembles were captured immediately after each pHIFU pulse to map the distribution and characteristics of the remnants of cavitation bubbles. The degree of resulting tissue disruption was assessed by histology and was correlated with cumulative Doppler power contours computed over the exposure region. The results show that ultrafast synchronous power Doppler is a promising approach for real-time quantitative monitoring of cavitational therapies. [Work supported by NIH R01EB023910 and T32DK007742.]

1:40

4pBA3. Dependence of sonoporation efficiency on microbubble size: An *in vitro* monodisperse microbubble study. Benjamin van Elburg (Phys. of Fluids Group, TechMed Ctr., Univ. of Twente, Enschede, Netherlands), Joke Deprez (Lab. of General Biochemistry and Physical Pharmacy, Ghent Res. Group on Nanomedicine, Ghent, Belgium), Martin van den Broek (BIOS Lab-on-a-Chip Group, MESA + Inst. for Nanotechnology, Univ. of Twente, Enschede, Netherlands), Stefaan De Smedt (Lab. of General Biochemistry and Physical Pharmacy, Ghent Res. Group on Nanomedicine, Ghent, Belgium), Michel Versluis (Phys. of Fluids Group, TechMed Ctr., Univ. of Twente, P.O. Box 217, Enschede 7500 AE, Netherlands, m.versluis@utwente.nl), Guillaume Lajoinie (Phys. of Fluids Group, TechMed Ctr., Univ. of Twente, Enschede, Netherlands), Ine Lentacker (Lab. of General Biochemistry and Physical Pharmacy, Ghent Res. Group on Nanomedicine, Ghent, Belgium), and Tim Segers (BIOS Lab-on-a-Chip Group, MESA + Inst. for Nanotechnology, Univ. of Twente, Enschede, Netherlands)

Sonoporation is the process where intracellular drug delivery is facilitated by ultrasound-driven microbubble oscillations. Several mechanisms have been proposed to relate microbubble dynamics to sonoporation including shear and normal stress. The present work aims to gain insight into the role of microbubble size on sonoporation by varying the microbubble size of monodisperse microbubble suspensions. Sonoporation experiments were performed *in vitro* on cell monolayers at an ultrasound frequency of 1 MHz at varying acoustic pressures (250–750 kPa) and pulse length (10, 100, 1000 cycles). Sonoporation efficiency was quantified using flow cytometry by measuring the FITC-dextran (4 kDa and 2 MDa) fluorescence intensity in 10,000 cells per experiment. We demonstrate that the bubble oscillation amplitude is nearly independent of the equilibrium bubble radius at acoustic pressure amplitudes that induce sonoporation ($\geq$500 kPa). However, we show that sonoporation efficiency is strongly dependent on the equilibrium bubble size and most efficiently induced by off-resonance 4.7-μm radius bubbles. These 4.7-μm bubbles are an order of magnitude more efficient than the polydisperse bubbles. We discuss that for our system shear stress is highly unlikely the mechanism of action, while we show that sonoporation efficiency correlates well with an estimate of the bubble-induced normal stress.

2:00

4pBA4. Characterization of acute and subacute immune response to boiling histotripsy ablation of pancreatic adenocarcinoma. Tatiana Khokhlova (Dept. of Medicine, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, tdk7@uw.edu), Yak-Nam Wang (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Helena Son (Dept. of Medicine, Univ. of Washington, Seattle, WA), Zhen Xu, Reliza McGinnis (Dept. of Biomedical Eng., Univ. of Michigan, Ann Arbor, MI), Brett Fite, Aris Kare, Katherine Ferrara (Dept. of Radiology, Stanford Univ., Palo Alto, CA), and Frederic Padilla (Focused Ultrasound Foundation, Charlottsville, VA)

Boiling histotripsy (BH) is a cavitation-based pulsed focused ultrasound method of mechanical tissue disintegration. When applied to tumor ablation, BH was shown in a number of preclinical studies to stimulate antitumor immune responses, with the extent and dynamics depending on the tumor microenvironment. In this work, we assessed the response of pancreatic ductal adenocarcinoma (PDAC)—an immunologically "cold" tumor—to BH ablation. C57BL/6J mice (n = 6 per group) were bilaterally subcutaneously grafted with KPC-derived mT4 cell line in the hind limbs. When one of the two tumors reached 8 mm, 80% of its volume was ablated with BH (1.5 MHz, 3 ms pulses, 1% duty cycle, 450 W power, 15 pulses/point). The tumors, inguinal lymph nodes and spleens were harvested at 1-, 3-, and 6-day time points and processed for immune phenotyping through flow cytometry, histology, and immunohistochemistry. At earlier time points, the response within the tumor was dominated by neutrophils, and by day 6—by inflammatory monocytes both in treated and contralateral sides versus sham. Transient increases in B cells, neutrophils, NK, NKT cells, and monocytes were observed in the ipsilateral lymph nodes and in spleens on day 3, and mostly subsided by day 6. These data will inform further studies in optimization of combination therapies for PDAC. [Work supported by Focused Ultrasound Foundation.]

2:20

4pBA5. Towards *in vivo* immunotherapy using high intensity focused ultraosund. Guillaume Lajoinie (Phys. of Fluids Group, Univ. of Twente, TechMed Ctr., U Enschede, Netherlands, g.p.r.lajoinie@utwente.nl), Yanou Engelen (Faculty of Pharmacy, Univ. of Ghent, Ghent, Belgium), Karine Breckpot (Vrije Universiteit Brussel, Brussels, Belgium), Dmitri Krysko (Ghent Univ., Ghent, Belgium), Stefaan De Smedt, and Ine Lentacker (Faculty of Pharmacy, Univ. of Ghent, Ghent, Belgium)

High-intensity focused ultrasound (HIFU) has been regarded as an appealing anti-cancer treatment for over a decade. HIFU indeed offers a combination of unique advantages: it is non-invasive and capable of delivering a precise and local thermal or mechanical dose without affecting the surrounding healthy tissue. HIFU is therefore already being applied as a cancer treatment in the clinics, although mostly as a thermal modality. On the other hand, immunotherapy has demonstrated remarkable promise for durable cancer treatment. However, its cost and complexity are prohibitive. Here, we evaluate (moderate) mechanical HIFU treatment and its capacity to induce immunogenic cell death with the aim to achieve immunotherapy treatment, directly *in vivo*. To that end, we treat a full lummox dish using an automatic scanning setup. Immunogenic cell death is monitored via distinct biomolecular markers (ATP secretion, HMGB1 secretion, calreticulin exposure). Cavitation activity is monitored via passive cavitation detection. Beyond the correspondence between acoustic cavitation and immunogenic cell death, our results also show a major influence of the cell type, namely, standard CT26 or B16F1O cells.

2:40–3:00 Break

3:00

4pBA6. Mechanical disintegration of benign and malignant tumors in *ex vivo* human prostate tissues using boiling histotripsy. Vera A. Khokhlova (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., University of Washington, Seattle, USA, verak2@uw.edu), Pavel B. Rosnitskiy, Sergey A. Tsysar, Maria M. Karzova (Lab. for Industrial and Medical Ultrasound, Phys. Faculty, Lomonosov Moscow State Univ., Moscow, Russian Federation), Sergey V. Buravkov (Faculty of Fundamental Medicine, Lomonosov Moscow State Univ., Moscow, Russian Federation), Natalya V. Danilova, Pavel G. Malkov (Medical Res. and Educational Ctr., Lomonosov Moscow State Univ., Moscow, Russian Federation), Ekaterina Ponomarchuk (Lab. for Industrial and Medical Ultrasound, Phys. Faculty, Lomonosov Moscow State Univ., Moscow, Russian Federation), Oleg A. Sapozhnikov (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., University of Washington, Seattle, USA), Tatiana Khokhlova (Dept. of Medicine, Univ. of Washington, Seattle, WA), Adam D. Maxwell (Dept. of Urology, Univ. of Washington School of Medicine, Seattle, WA), Yak-Nam Wang (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Aleksey V. Kadrev (Medical Res. and Educational Ctr., Lomonosov Moscow State Univ., Moscow, Russian Federation), Andrey L. Chernyaev (Dept. of Fundamental Pulmonology, Pulmonology Sci. Res. Inst., Moscow, Russian Federation), and George R. Schade (Dept. of Urology, Univ. of Washington School of Medicine, Seattle, WA)

The feasibility of boiling histotripsy (BH) to mechanically liquefy *ex vivo* human prostate tissue with benign hyperplasia (BPH) and adenocarcinoma (PCa) was assessed. Volumetric BH lesions were generated in fresh tissue samples under B-mode guidance using a 1.5-MHz focused transducer, 10- and 1-ms pulses with 1% duty cycle. Prior to BH exposures, the samples were analyzed using shear wave elastography (SWE) to determine whether their mechanical properties were clinically representative. Completeness and efficiency of ablation for BH lesions were evaluated grossly, histologically, and using B-mode ultrasound. The SWE measured Young's modulus of the samples was shown to be within the typical range observed clinically. During the exposures, BH-induced hyperechogenic bubbles were visible using B-mode, and post-treatment hypoechoic regions indicated successful tissue fractionation. Gross analysis of the exposed samples with BPH revealed ablation of the targeted tissue inside stiff BPH nodules. The sonications using 1-ms pulses (with 150 pulses-per-focus) were twofold faster comparing to 10-ms (with 30 pulses-per-focus). Histological analysis revealed liquefied lesions containing homogenized cell debris in all BPH samples. A pilot experiment of human PCa tumor fractionation into subcellular fragments was confirmed histologically. [Work supported by R01CA258581, R01DK119310, and RSF 20-12-00145.]

3:20

4pBA7. Inactivation of bacteria using different Histotripsy regimes: Toward the treatment of abscesses. Pratik Ambekar (Univ. of Washington, Seattle, WA), Yak-Nam Wang (Appl. Phys. Lab., Univ. of Washington, Seattle, WA 98105, ynwang@uw.edu), Tatiana Khokhlova (Dept. of Medicine, Univ. of Washington, Seattle, WA), Gilles Thomas, Matthew Bruce, Daniel Leotta (Univ. of Washington, Seattle, WA), Adam D. Maxwell (Dept. of Medicine, Univ. of Washington, Seattle, WA), Pavel Rosnitskii, Stephanie Totten, Jeff Thiel, Shelby Pierson (Univ. of Washington, Seattle, WA), Keith Chan (Vantage Radiology and Diagnostic Services, Seattle, WA), Conrad Liles, Evan Dellinger, Adeyinka Adedipe, Wayne Monsky (Dept. of Medicine, Univ. of Washington, Seattle, WA), and Thomas Matula (Univ. of Washington, Seattle, WA)

Infected abscesses are defined as localized collections of infected purulent material delineated by a capsule composed of granulation and connective tissue. They can affect any part of the body. Current treatment typically involves antibiotics with either long-term catheter drainage or incision and washout. Given the ubiquitous nature of bacteria and the serious threat of multi-drug resistant strains, abscesses have become a persistent global healthcare problem. Using histotripsy, a potential new noninvasive treatment modality, we have evaluated bacterial inactivation of *Staphylococcus aureus* (*S. aureus*) and *Escherichia coli* (*E. coli*) microbes. We have determined the peak negative pressure threshold for inactivation of *E. coli* in solution for cavitation histotripsy (low cycles, high pulse repetition rate) and boiling histotripsy (high cycles, high pulse repetition rate) for different

treatment times. The effect of transducer frequency on bacterial inactivation has also been evaluated. The optimal regimen was used to evaluate the *S. aureus* inactivation. Bacterial inactivation has also been evaluated in an abscess phantom model. Custom transducers have been developed and evaluated in an animal model for abscesses. [Work supported in part by NIH NIBIB #R01EB019365 and NIAID #R01AR080120.]

3:40

4pBA8. Ultrasound-responsive hydrogel microcapsules for on-demand drug release. Rachel D. Field, Margaret A. Jakus (Biomedical Eng., Columbia Univ., New York, NY), Xiaoyu Chen (Dept. of Mech. Eng., Massachusetts Inst. of Technol., Cambridge, MA), Kelia A. Human (Biomedical Eng., Columbia Univ., New York, NY), Xuanhe Zhao (Dept. of Mech. Eng., Massachusetts Inst. of Technol., Cambridge, MA), Parag V. Chitnis (Bioengineering, George Mason Univ., 4400 University Dr., 1J7, Fairfax, VA 22030, pchitnis@gmu.edu), and Samuel K. Sia (Biomedical Eng., Columbia Univ., New York, NY)

Hydrogel-based implantable systems offer viable solutions for localized drug delivery but often lack the ability to easily achieve on-demand actuation or real-time tuning of release kinetics in response to physiological changes. Here, we present a hydrogel microcapsules produced using two-phase microfluidics that can release drugs on demand as triggered by focused ultrasound (FUS). The biphasic microcapsules consist of an outer phase of mixed molecular weight (MW) poly(ethylene glycol) diacrylate that mitigates premature payload release and an inner phase of high MW dextran with payload that breaks down in response to FUS. Compound release from microcapsules could be triggered as desired; 0.4 μg of payload was released across 16 on-demand steps over days. We detected broadband acoustic signals amidst low heating, suggesting inertial cavitation as a key mechanism for payload release. Overall, FUS-responsive microcapsules are a biocompatible and wirelessly triggerable structure for on-demand drug delivery over days to weeks.

4:00

4pBA9. Enhancement of brain hyperthermia via transcranial magnetic resonance imaging-guided focused ultrasound and microbubbles—Heating mechanism investigation using COMSOL. Zhouyang Xu (School of Biomedical Eng., ShanghaiTech Univ., 393 Middle Huaxia Rd., Pudong, Shanghai, Shanghai, Shanghai 201210, China, xuzhy3@shanghaitech.edu.cn), Samuel Pichardo (Cumming School of Medicine, Univ. of Calgary, Calgary, AB, Canada), and Bingbing Cheng (School of Biomedical Eng., ShanghaiTech Univ., Shanghai, China)

Noninvasive methods for enhancing the brain drug delivery has been pursued for years. Previously we developed a new MR-guided focused ultrasound (FUS)-based technique, which can achieve targeted brain hyperthermia for heat-triggered drug release and simultaneously open the blood–brain barrier safely for drug penetration. However, the underline mechanisms were unclear. This study aimed to explore the mechanisms for the enhanced FUS brain tissue hyperthermia with microbubbles via numerical modeling in COMSOL. The acoustic wave equation was employed to describe the FUS propagation. A bubble dynamics equation was adopted for calculating the stable bubble oscillations under FUS exposures. A modified bioheat transfer equation was utilized to compute the heating, with various heating sources including FUS, microbubble acoustic emission (MAE), and viscous dissipation (VD). The microbubbles were randomly distributed within the focal region. The sonication time was 6s with an initial temperature of 41°C. The average temperature in the focal region were 41.65°C, 42.24°C, 42.98°C, and 43.59°C for FUS alone, FUS + MAE, FUS + VD, and FUS + MAE + VD, respectively. Compared with the FUS alone, both MAE and VD made significant contributions to the heating with additional temperature increases of 47.6% and 67.2%, respectively.

4p THU. PM

Session 4pCAa

Computational Acoustics and Physical Acoustics: Computational Aspects of Spatial Audio

D. Keith Wilson, Cochair
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Jonas Braasch, Cochair
School of Architecture, Rensselaer Polytechnic Institute, School of Architecture, 110 8th Street, Troy, NY 12180-3522

Chair's Introduction—12:55

Invited Papers

1:00

4pCAa1. Computational aspects of real-time auralizations of jazz venues. E. K. Ellington Scott (School of Architecture, Rensselaer Polytechnic Inst., Troy, NY) and Jonas Braasch (School of Architecture, Rensselaer Polytechnic Inst., Greene Bldg., 110 8th St., Troy, NY 12180, braasj@rpi.edu)

Historically, architectural acoustics has focused on concert halls for European Classical music, even though other genres, including rock, blues, and Jazz, have large groups of followers as well. Additionally, the acoustic requirements for European Classical music differ fundamentally from Jazz and other music. For example, Jazz venues are typically smaller and dryer than a traditional European Classical concert hall. A recent survey showed that reverberation in these venues plays a secondary role to parameters that describe the affordance of a space to allow communication between improvising musicians. This presentation focuses on the unique computational auralization aspects of Jazz venues, including wave-based approaches for smaller enclosures, low-latency requirements, intelligent accompanying systems that can adapt ad-hoc to jazz soloists, and the use of anechoic jazz recordings as source material. The adequacies of different reproduction systems, from higher-order ambisonics and wave-field synthesis to headphone-based systems, will also be discussed. [Work supported by James West Fellowship, Leo and Gabriella Beranek Scholarship, and NSF HCC-1909229.]

1:20

4pCAa2. High quality low complexity binaural rendering for headphones. Karlheinz Brandenburg (Brandenburg Labs GmbH, Ehrenbergstr. 11, Ilmenau 98693, Germany, khb@brandenburg-labs.com), Nils Merten, Thomas Thron, and Ulrike Sloma (Brandenburg Labs GmbH, Ilmenau, Germany)

Playing back immersive audio via headphones has been a research topic for a long time. Although there have been advances, bigger breakthroughs are still missing. The basic idea of binaural rendering systems can be summarized with the following elements: The emission characteristics of the audio source, the transmission path from audio source to listener as well as suitable head related transfer functions. Systems following this basic paradigm range from dummy head stereophony over purely simulation based approaches to current parametric algorithms including the usage of room impulse responses. According to research in audio cognition, it is important to incorporate positional and rotational tracking data of the head close to real time as well as acoustic information about the listening room. Based on the chosen approach, the effort to acquire the measurements, to prepare the rendering or the computational demand of the rendering itself might be high. In this paper, we report on a low complexity rendering system which allows high quality binaural reproduction. A recent listening test and many reviews from listeners have proven, that the perception of the rendered audio signals is equivalent to listening to real audio sources in a room.

1:40

4pCAa3. Generating 3-D auditory stimuli for the evaluation of spatial impression in musical instruments. Jorge Trevino (Yamaha Corp., 10-1, Nakazawa-cho, Naka-ku, 21 Bldg. 4th Flr. Kansei Group, Hamamatsu, Shizuoka 430-8650, Japan, jorge.trevino@music.yamaha.com), Fusako Ishimura, Masaru Tanaka, and Yasuo Shiozawa (Yamaha Corp., Hamamatsu, Naka-ku, Shizuoka, Japan)

Spatial hearing plays an important, but not well understood, role in the perception of musical instruments and audio equipment. Psychophysical experiments are necessary to elucidate the processes which convey the spatial impression of sound sources. To this end, we propose a way to generate virtual stimuli for research on the spatial impression of musical instruments, specifically pianos. Our proposal, based on acoustical holography, takes a series of measurements over a dense grid and designs a virtual vibrating plate with the same directivity as the piano, as observed from the measuring grid. The number of active modes in the left-right and front-back directions of this virtual source can be controlled to reduce the radiation pattern complexity and generate a wide range of virtual stimuli. The resulting

sound fields are rendered using binaural Ambisonics, using the position of the piano player as the spherical harmonic expansion center. The proposal was used in actual psychophysical experiments looking to shed light into the spatial impression of digital and acoustic pianos. The results show that the proposal does generate spatial sound signals which are consistently associated with different degrees of richness in the spatial impression.

2:00

4pCAa4. Room-aware portable Auditory Augmented Reality: Real-time spatial audio generation with geometric data analysis. Rai Sato (Graduate School of Culture Technol., Korea Adv. Inst. of Sci. and Technol., 291 Daehak-ro, Yuseong-gu, Daejeon 34141, Korea (the Republic of), s.rai@kaist.ac.kr), Yuto Izumi (Dept. of Architecture and Architectural Eng., Graduate School of Eng., Kyoto Univ., Kyoto, Japan), and Sungyoung Kim (Graduate School of Culture Technol., Korea Adv. Inst. of Sci. and Technol., Daejeon, Korea (the Republic of))

This study proposes an innovative and practical approach to spatial acoustic rendering for Auditory Augmented Reality (AAR) in small room usage. AAR aims to enhance user experiences by overlaying spatially appropriate sounds in the real environment, creating the illusion that virtual sounds coexist in the physical world. However, current technological constraints in hardware and AAR rendering present challenges in creating a truly immersive sensation of "being there." To address these challenges, we present a dynamic processing framework for a typical mobile AAR setting. This framework combines a game-audio engine with 3-D visual computation both specialized for real-time processing, providing a plausible spatial impression with lightweight computation. The process involves capturing room information by the Light Detection And Ranging (LiDAR) sensor mounted on the device, followed by estimating its dominant acoustic materials. The generated data will be used to compute dynamic early reflections and late reverberation. Early reflections are calculated up to the fourth reflection using the image source method, considering the acoustic materials. Late reverberation is generated using the Feedback Delay Network (FDN), with room frequency characteristics and spatial directivity varying relative to the listener's position. All components will be represented in fifth-order Ambisonics and binauralized with head-tracking. Notably, this dynamic rendering method could be light enough to handle all processing on a single mobile device.

2:20

4pCAa5. Binaural effects and rendering of edge diffraction in geometrical acoustics. Christoph Kirsch (Medical Phys. and Cluster of Excellence Hearing4all, Carl von Ossietzky Universität Oldenburg, Ammerlaener Heerstr. 114-118, Oldenburg 26125, Germany, christoph.kirsch@uol.de) and Stephan D. Ewert (Medical Phys. and Cluster of Excellence Hearing4all, Carl von Ossietzky Universität Oldenburg, Oldenburg, Germany)

Virtual acoustic environments have many applications in entertainment, architectural planning, and hearing research. Geometrical acoustics (GA) is commonly used, offering high computational efficiency by assuming ray-like sound propagation. However, GA does not account for wave-related effects such as edge diffraction, leading to perceptually dissatisfactory results or discontinuities, e.g., when a sound source is occluded. GA can be extended by constructing edge-diffracted sound paths. Recently, we have suggested the universal diffraction filter approximation (UDFA) to model the spectral effects of diffraction from infinite and finite edges and objects composed thereof using highly efficient recursive filters. This contribution investigates how edge diffraction affects the spatial perception of a sound source in conditions where a flat plate occludes the direct sound or produces a reflection. Binaural recordings were obtained with a head and torso simulator at discrete points of a sound source trajectory in the vicinity of a flat plate, including conditions near the incident and reflection shadow boundary. In a listening test, the perceived source location was matched for conditions with and without plate. Differences in localization were observed, and an approach for binaural rendering of simulated diffraction in the GA context is suggested, replicating the localization results obtained for the dummy-head recordings.

Session 4pCAb

Computational Acoustics and Physical Acoustics: Computational Aeroacoustics

Z. Charlie Zheng, Cochair
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Con Doolan, Cochair
UNSW, School of Mechanical and Manufacturing Engineering, UNSW Sydney, Sydney 2052, Australia

Chair's Introduction—3:15

Invited Papers

3:20

4pCAb1. Wall-pressure fluctuations in weakly compressible turbulent channel flow. Meng Wang (Aerosp. and Mech. Eng., Univ. of Notre Dame, 105 Hessert Lab., Notre Dame, IN 46556, m.wang@nd.edu), Yi Liu, and Kan Wang (Aerosp. and Mech. Eng., Univ. of Notre Dame, Notre Dame, IN)

Wall-pressure fluctuations in turbulent wall-bounded flows are detrimental in many applications because they can cause structural vibrations and acoustic radiation. Their spectral behavior at subconvective wavenumbers are to date poorly understood and predicted, particularly in low-Mach-number flows. In this study, compressible direct numerical simulation is employed to elucidate the low-wavenumber behavior of wall-pressure fluctuations in turbulent channel flow and the effect of flow Mach number in the nearly incompressible regime. Simulations are conducted at bulk Mach numbers of 0.4, 0.2, and 0.1, and friction Reynolds number of 180. In addition to the convective ridge that is virtually Mach-number independent, acoustic ridges representing longitudinal and oblique waves are clearly identified in the two-dimensional wavenumber-frequency spectrum. The acoustic peaks are orders of magnitude weaker than the convective peak and decay with flow Mach number, but remain distinctly identifiable even at Mach 0.1. The acoustic energy in the supersonic wavenumber range is significantly enhanced by the onset of the first oblique mode but not much affected by the higher modes. The effect of a small, two-dimensional surface hump is also considered, which is shown to elevate the spectral level of the fluctuating wall pressure in the subconvective wavenumber range by several decades due to acoustic diffraction by the hump. [Work supported by the U.S. Office of Naval Research.]

3:40

4pCAb2. Acoustic characterization and its relation to POD and SPOD of flow around tandem cylinders. Z. C. Zheng (Mech. and Aerosp. Eng., Utah State Univ., 4130 Old Main Hill, Logan, UT 84322, zzheng@ku.edu) and Jerry Zhou (Mech. and Aerosp. Eng., Utah State Univ., Logan, UT)

Flows around two tandem cylinders are simulated using an immersed-boundary method. Cases in the lock-in, transitional and quasi-periodic regimes are investigated. The flowfield is used to provide sources to analyze acoustic pressure at the far field. Both proper orthogonal decomposition (POD) and spectral POD (SPOD) methods are performed for the flowfield to extract flow physics. The relations between the acoustic characterization and the flowfield POD and SPOD are then discussed.

Contributed Papers

4:00

4pCAb3. Sonic boom near field analysis of a Mach 5 wave-rider configuration. Samuele Graziani (DIMEAS, Politecnico di Torino, Corso DC degli Abruzzi 24, Torino, Turin 10129, Italy, samuele.graziani@polito.it), Nicole Viola, and Roberta Fusaro (DIMEAS, Politecnico di Torino, Turin, Italy)

The following paper discuss the near field sonic boom results obtained with CFD simulations regarding a Mach 5 wave rider configuration derived from the STRATOFLY heritage. Sonic boom near field have emerged as a vital tool for the mitigation and understanding of the impact of supersonic and hypersonic aircraft configurations. The work done discusses the major consideration in the CFD approach for the flow field simulations in the vicinity of the aircraft, including grid generation techniques based on the NASA sonic boom predictions workshop carried out between 2014 and 2020 and the results obtained for different flight conditions and operations. The near field simulation necessitates of accurate predictions of shock wave propagation generated by the aircraft, employing the high-fidelity numerical schemes and turbulence models. These results will be the input for our propagation tool that manage to propagate the shocks from the aircraft to the ground. This work is carried out within the MORE&LESS project, that is an EU-funded project and has the aim to investigate the environmental impact of supersonic aircraft through multi-fidelity simulations and test campaigns for creating multidisciplinary holistic framework for the evaluation of the future supersonic aircraft, trajectories, and operations.

4:20

4pCAb4. Computational aeroacoustics and phase-conjugation technique for localizing flow-induced sources. Akhilesh Mimani (Mech. Eng., Indian Inst. of Technol. Kanpur, MED, IIT Kanpur, Kanpur, Uttar Pradesh 208016, India, amimani@iitk.ac.in) and Shubham Kumar (Mech. Eng., Indian Inst. of Technol. Kanpur, Kanpur, India)

This paper presents a framework comprising computational aeroacoustics (CAA) solvers and a numerical phase-conjugation technique to localize flow-induced noise sources generated by bodies immersed in low Mach number flows. Two test-cases were considered, namely, flow over a two-dimensional (a) circular cylinder at Reynolds number Re = 150 and Mach number M = 0.2 and (b) NACA-0012 airfoil at Re = 5000 and angle of attack (AoA) $\alpha = 5$ deg. The CAA simulations were carried out in OpenFOAM wherein the two-dimensional unsteady, incompressible Navier–Stokes equations was solved using either the pressure-implicit splitting of operators (PISO) algorithm or large eddy simulation (LES). The near-field aerodynamic sources were extracted in terms of the unsteady Lighthill stress tensor terms, following which the far-field acoustic data were computed on boundary nodes using different methods which include the numerical solution of the Lighthill equation, Curle's analogy, and Ffowcs Williams and Hawkings (FWH) analogy as well as linearized Euler equations (LEE). Next, the frequency-domain phase-conjugation (PC) method was implemented in the finite-element (FE) based COMSOL software to reconstruct the radiated acoustic pressure field, whereby the dipole source location was readily identified by pair of focal spots at the cylinder or at the airfoil trailing-edge (TE).

4:40

4pCAb5. Numerical simulation of flow-induced noise generation and propagation of a floating offshore wind turbine with prescribed pitch motion. Ruosi Zha (School of Ocean Eng. and Technol., Sun Yat-sen Univ. & Southern Marine Sci. and Eng. Guangdong Lab. (Zhuhai), B434-1, Haiqin Bldg. No. 3, Zhuhai Campus, Xiangzhou District, Zhuhai, Guangdong 519082, China, zhars@mail.sysu.edu.cn) and Kai Wang (School of Ocean Eng. and Technol., Sun Yat-sen Univ. & Southern Marine Sci. and Eng. Guangdong Lab. (Zhuhai), Zhuhai, Guangdong, China)

It is important to evaluate flow-induced noise emitted due to operations of floating offshore wind turbines since noise pollution can damage the environments of seabirds and aquatic species. This paper studies the aerodynamic and aeroacoustic performance of a laboratory three-blade wind turbine model with an airfoil profile of NREL S826. The wind turbine blade rotation motion is coupled with the prescribed pitch motion of the floating offshore wind power platform. The influence of prescribed pitch motions on the noise generated by wind turbine blades and noise propagation was discussed. Adopting the computational fluid dynamics (CFD) method, the aerodynamic noise field on a blade is simulated by the improved delayed detached eddy simulation (IDDES) model and Ffowcs Williams–Hawkings (FW-H) acoustic analogy. The overset grid technique is adopted to simulate the six degrees of freedom (DoFs) motions of the floating offshore wind turbine model. The sound source distributions on blades for noise generation are obtained, and the flow-induced noise propagation characteristics are analyzed. It is found that the influence of the pitch amplitude and frequency on the flow-induced noise field as well as the wake vortex characteristics should not be neglected.

Session 4pEA

Engineering Acoustics: Topics in Engineering Acoustics

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Tyler McGee, Cochair
Walker Department of Mechanical Engineering, University of Texas at Austin, 204 E Dean Keeton St, Austin, TX 78712

Contributed Papers

1:00

4pEA1. Modal analysis of lithium-ion batteries for estimation of state of charge and state of health. Tyler McGee (Walker Dept. of Mech. Eng., Univ. of Texas at Austin, 204 E Dean Keeton St., Austin, TX 78712, tyler.m.mcgee@gmail.com), Ofodike A. Ezekoye, and Michael Haberman (Walker Dept. of Mech. Eng., Univ. of Texas at Austin, Austin, TX)

The robustness and safety of battery-operated systems will become critically important as society transitions away from fossil fuels. In the foreseeable future, lithium-ion batteries will be used for high-power, high-capacity applications such as electric vehicles and renewable energy storage. In these applications, which require thousands of cells, existing battery management systems do not monitor the operating conditions of each cell. Variability in temperature and pressure can affect a cell's state of charge (SOC) or state of health (SOH). This environmental loading coupled with cell-to-cell variability makes SOC/SOH estimation very difficult. Inaccurate measurements of SOC/SOH can reduce the lifespan of battery systems or lead to accidental overcharge and thermal runaway. We present modal analysis as a viable approach to estimate SOC/SOH for electrically cycled cells. Modal tests are performed using laser Doppler vibrometry on 10 Ah Nickel–Manganese–Cobalt pouch cells at 0% and 100% SOC across more than 30 cycles. Changes in the resonance frequencies of the cell are correlated with SOC/SOH to provide an understanding of how the mechanical properties of cell components change as a function of charge level and aging. This work demonstrates that modal analysis may be used as a tool for regular battery maintenance to improve battery safety.

1:20

4pEA2. Real time GMAW weld bead profile mapping using acoustic sensing. Mitchell Cullen (Univ. of Technol. Sydney, unit 6, 2 Mooney St., Strathfield South, New South Wales 2136, Australia, mitchell.cullen@uts.edu.au), J. C. Ji, and Jeffrey Parnell (Univ. of Technol. Sydney, Sydney, New South Wales, Australia)

Monitoring the penetration profile of Gas Metal Arc Welding (GMAW) is critical in determining the overall quality and structural integrity of the produced weld bead. However, due to the instability and complexity of the GMAW process, it can be difficult to accurately monitor the weld bead formation and penetration profile. Due to this complexity, traditional methods for modeling the welding process, such as CFD analysis, are unable to be computed in real time. In this work, a new analytical method of estimating the weld bead penetration profile in real time using the sound signal is proposed. This method monitors the sound signal generated during the droplet transfer process to estimate both the heat input and material deposition into the weld pool. Using this estimation, a digital twin of the welding process is produced, allowing for the operator to monitor the weld bead formation in real time, guaranteeing the structural integrity of the final weld bead.

1:40

4pEA3. The UNSW tip clearance flow noise test rig. Manuj Awasthi (School of Mech. and Manufacturing Eng., UNSW, Sydney, Sydney, New South Wales, Australia, m.awasthi@unsw.edu.au), Angus Wills (School of Mech. and Manufacturing Eng., UNSW, Sydney, Kensington, New South Wales, Australia), Danielle Moreau (School of Mech. and Manufacturing Eng., UNSW, Sydney, Sydney, New South Wales, Australia), Paul Croaker, and Paul Dylejko (Maritime Div., Defence Sci. and Technol. Group, Melbourne, Victoria, Australia)

A new test rig to measure the far-field sound, surface pressure fluctuations and flow-field in tip clearance flows has been designed at UNSW. The test rig is a part of the UNSW anechoic wind tunnel (UAT) and has been designed to allow studies of the tip clearance flow and sound radiation across a vast parameter space—tip clearance height, Reynolds number, angle of attack, and the tip geometry. This paper will present and discuss the design, characterization, and capabilities of this new test rig, along with some far-field sound results for a tip clearance flow formed by placing a stationary cambered airfoil adjacent to a wall. The measurements were performed for 51 different clearance heights at free-stream velocity of 30 m/s. The geometric angle of attack in the measurements was 5.6 and the clearance height Reynolds number ranged between 2000 and 102,000. The far-field sound was measured using a 64-microphone spiral phased array and the array output was beamformed to reveal the tip clearance noise sources. It is expected that the test rig will aid in a better understanding of the sound radiated by tip clearance flows which may be used to devise noise control strategies in the future.

2:00

4pEA4. The design, characterization, and performance assessment of an aeroacoustic measurement system for a hard-walled wind tunnel. Manuj Awasthi (School of Mech. and Manufacturing Eng., UNSW, Sydney, Sydney, New South Wales, Australia, m.awasthi@unsw.edu.au), Con Doolan (School of Mech. and Manufacturing Eng., UNSW, Sydney, Sydney, New South Wales, Australia), and Paul Croaker (Maritime Div., Defence Sci. and Technol. Group, Melbourne, Victoria, Australia)

A new aeroacoustic measurement system for the large wind tunnel (LWT) at the University of New South Wales (UNSW) has been designed and characterized. The LWT is a closed-return type wind tunnel with a 2.87 m $\times$ 1.28 m $\times$ 0.92 m hard-walled test-section, which contains 0.61 m $\times$ 0.61 m access windows one of which was used to attach a 1.06 m $\times$ 0.61 m $\times$ 0.61 m anechoic chamber lined with 100 mm thick Basotect foam. A 16-microphone spiral phased array embedded in the chamber was used to characterize the acoustical properties of the chamber and localize several acoustic and aeroacoustic sources placed in the reverberant test-section. Acoustic sources considered include single and dual speaker setup,

while a UAV propeller and a forward-backward step mounted on the test-section sidewall were used as an aeroacoustic source. The beamforming output in each case was deconvolved using several existing algorithms to assess their performance under reflective boundary conditions. Additionally, a numerical tailored Greens function approach to beamforming is being considered to exclude the effects of reverberations within the test-section from the array output. It is expected that the present work will help develop strategies and guidelines for aero/hydroacoustic testing under reverberant conditions.

2:20

4pEA5. Acoustic lens that diffuses plane waves composing of multiple spherical obstacles. Hidetoshi Masukawa (Kogakuin Univ., Bldg. 5 #405, 2665-1, Nakano-machi, Hachiouji-shi, Tokyo 192-0015, Japan, em23041@ns.kogakuin.ac.jp) and Yoshinori Takahashi (Kogakuin Univ., Tokyo, Japan)

Recently, a non-electrical smartphone loudspeaker based on acoustic horn has been created and marketed. An acoustic horn is directional and has the advantage of propagating strong sound pressure in the direction the horn is facing. However, it has the disadvantage that sound pressure cannot be expected in any direction other than the direction in which the form facing. This study proposed an acoustic lens in the form of a cylinder composed of multiple spherical obstacles. The density of the sphere was calculated using the phase difference equation and the equation proposed by Kock *et al*. The density of the sphere at the center of the lens is lower than at the circumferential direction. This change in density slows down the apparent sound speed passing through the further from the center. In this work, FDTD simulations were performed to evaluate the sound directivity for the acoustic lens. This work also test manufactured the acoustic lens and performed an experiment in an anechoic chamber to evaluate the characteristics of the lens.

2:40

4pEA6. Acoustic probe for temperature measurement suitable for operation with audio interfaces having random input/output delays. Yuki Fujita (Graduate School of Sci. and Technol., Univ. of Tsukuba, 3M-105, Third Area, Tsukuba Campus, 1chome-1-1 Tennoudai, Tsukuba 305-0006, Japan, fujita.yuki.21@aclab.esys.tsukuba.ac.jp), Tadashi Ebihara, Naoto Wakatsuki, Yuka Maeda, and Koichi Mizutani (Inst. of Systems and Information Eng., Univ. of Tsukuba, Tsukuba, Japan)

Measuring temperature distribution is an important technique in the fields of meteorology, agriculture, and architecture. The acoustic probe emits sound waves from a speaker and measures the time it takes for the waves to reach a microphone, enabling the measurement of the average temperature between the speaker and the microphone. Additionally, by placing multiple acoustic probes around the measurement area, it becomes possible to measure the temperature distribution using fewer sensors compared to point-type sensors. However, acoustic probes can have higher costs due to the requirement for externally controllable setups or interfaces with minimal audio input/output (I/O) delays. To address this issue, we propose an acoustic probe that achieves accurate measurements even with a low-cost audio interface featuring random I/O delays. The proposed acoustic probe measures the delay time from the recording start to the playback start. This design cancels the error of sound wave propagation time caused by random delays in the audio interface. Through temperature measurement experiments using the proposed acoustic probe, it was revealed that precise measurements (the error of the proposed probe was 0.34°C, while the error of the existing probe was 4.88°C) could be achieved by canceling the random delays in the audio interface.

3:00–3:20 Break

3:20

4pEA7. Bringing free weight areas under acoustic control. Max Brahman (None, 1/2-22 Kirkham Rd. West, Keysborough, Melbourne, Victoria 3173, Australia, max.brahman@getzner.com)

Fitness studios that are housed in office, residential, or hotel buildings are repeatedly confronted with the problem that the strong vibrations caused by dropping weights lead to noise and vibrations in the entire building or even cause damage to existing ceiling structures. For these situations, Getzner has developed the g-fit Shock Absorb system with low construction heights and low weight. This is achieved through several layers of polyurethane foam materials including high damping or high elasticity sub-layer. The impact is isolated, the drop energy is absorbed by the material and dangerous rebound of the weights (dumbbell, barbell, kettlebells, etc.) is prevented. But the best flooring system can hardly be used without a proper prognosis of the isolation effect and the achieved sound pressure level in adjacent rooms. This paper is about the development of a prognosis tool that allows to predict the sound pressure level as a function of the impact energy, the type of weight at gym, and the ceiling mass for all g-fit Shock Absorb floor coverings. The extensive drop tests as well as the evaluation and the development of a prediction function is described in detail and explained by examples.

3:40

4pEA8. Application of heat transfer at the combustor wall to attenuate self-excited thermoacoustic oscillations. Molly Evans (Univ. of Canterbury, 20 Kirkwood Ave., Christchurch, Canterbury 8041, New Zealand, molly.evans@pg.canterbury.ac.nz) and Dan Zhao (Dept. of Mech. Eng., Univ. of Canterbury, Christchurch, New Zealand)

Self-sustained thermoacoustic oscillations (also known as combustion instability) arise in combustors when pressure and heat release fluctuations couple to produce high-energy sound waves. The need for diverse strategies to control these oscillations will increase as future design moves towards highly efficient, lean combustors. In this work, the impact of heat transfer at the combustor wall on the generation of thermoacoustic oscillations is examined through computational fluid dynamics. Characteristics of the thermoacoustic oscillations generated across cases with a range of thermal boundary conditions are compared to an adiabatic reference case. The amplitude of limit cycle oscillations in a Rijke tube is seen to decrease as wall temperatures increase. A stable combustor with a negative growth rate is observed when the full outer wall is heated to 900 K, and a sound pressure level reduction of 75 dB is achieved. The most efficient stable configuration is achieved when only the inlet duct wall is heated to 1000 K, requiring a minimum of 430 W of input power. The present work introduces an alternative stable combustor design and demonstrates that optimizing the temperature distribution of the combustor wall can effectively dampen thermoacoustic oscillations.

4:00

4pEA9. Investigation of heat transfer at the outer combustor wall on self-sustained thermoacoustic oscillations in a Rijke tube. Molly Evans (Univ. of Canterbury, 20 Kirkwood Ave., Christchurch, Canterbury 8041, New Zealand, molly.evans@pg.canterbury.ac.nz) and Dan Zhao (Dept. of Mech. Eng., Univ. of Canterbury, Christchurch, New Zealand)

Thermoacoustic oscillations have plagued combustion systems for decades and are becoming increasingly relevant as engines become more powerful, more compact, and more eco-friendly. These oscillations arise from constructive coupling between unsteady heat release and acoustic pressure waves, and result in strong pressure oscillations that can inhibit flame stability and induce component failure. In this work, a simple numerical model is used to study the effects of the temperature distribution of the outer combustor wall on the thermoacoustic stability of a Rijke tube. A Galerkin series expansion of the linearized acoustic governing equations is coupled with a classical G-equation premixed flame model to determine the minimum energy required to maintain stability for various system configurations, and results are compared against computational fluid dynamics simulations for select cases.

4:20

4pEA10. Enhancing heat-driven acoustic characteristics of standing wave thermoacoustic engines by using corrugated stack. Lixian Guo (Dept. of Mech. Eng., Univ. of Canterbury, Univ. of Canterbury, Christchurch, New Zealand, lixian.guo@pg.canterbury.ac.nz) and Dan Zhao (Dept. of Mech. Eng., Univ. of Canterbury, Christchurch, New Zealand)

The present work considers the heat-driven acoustic characteristics and dynamic thermal-fluid flow fields of a standing-wave thermoacoustic engine

(SWTAE) by using sinusoidally shaped corrugated stack surfaces. 3-D numerical SWTAE models are developed and validated, aiming to examine the heat-driven acoustic effects caused by different corrugation amplitudes and wave-lengths of the sinusoidal stack surface. The results demonstrate that corrugated-shaped stack channels increase the contact area of the working gas in the stack area, and the thermos-acoustic conversion is more amplified. Compared to the SWTAE with uniform-shaped stack, the corrugated-shaped stack exhibit enhanced acoustic power output while maintaining a constant acoustic oscillation frequency. With the corrugation peak and wave-length of the sinusoidal stack are 2 and 0.2 mm, respectively, the amplitude and acoustic power of the acoustic pressure oscillations increase by 10.12% and 17.31%, respectively, in comparison with that of the conventional SWTAE. However, when the corrugation peak exceeds 0.4 mm, stronger nonlinear acoustics and flow effects are observed in the stack channels, leading to a reduction in the heat-driven acoustic power output, and thermo-acoustics energy conversion efficiency. The developed model highlights the effects of the corrugated-shaped stack on enhancing acoustic power output and thermo-acoustic conversion efficiency.

4:40

4pEA11. Effects of temperature difference and operating pressures on heat-driven acoustic characteristics and nonlinear behaviors in a looped tube traveling wave thermoacoustic engine. Lixian Guo (Dept. of Mech. Eng., Univ. of Canterbury, Christchurch, NZ, academic, Christchurch, New Zealand, lixian.guo@pg.canterbury.ac.nz) and Dan Zhao (Dept. of Mech. Eng., Univ. of Canterbury, Christchurch, NZ, academic, Christchurch, New Zealand)

In this work, the output heat-driven acoustic characteristics and dynamic thermal-fluid flow fields in a looped-tube traveling-wave thermoacoustic engine (TWTAE) are numerically investigated. Emphasis is placed on optimizing acoustic power output from the TWTAE by varying its operating pressure and temperature difference across the regenerator. For this, a time domain full-scale 3-D traveling-wave TAE model is developed, and then validated by comparing with those results obtained from the experimental data available in the literature. The present results indicate that the acoustic pressure oscillations and the acoustic power are increased with increased operating pressure of the working gas. Furthermore, nonlinear acoustics and flow dynamics in the heat-driven acoustic and flow fields of the TWTAE such as vortex generation around the regenerator and Gedeon streaming are observed. Considering the comprehensive acoustic characteristics, the optimal heat-driven acoustic power output, and thermo-acoustics energy conversion efficiency are achieved, as the working air pressure is set to 0.4 MPa. Increasing the temperature difference across the regenerator can further improve the acoustic power output from the TWTAE. In summary, the present 3-D model can be used as a design tool for predicting and optimizing looped tube traveling-wave TAE performances with detailed thermos-fluid dynamics and acoustics characteristics.

Session 4pMU

Musical Acoustics: General Topics in Musical Acoustics

James Cottingham, Chair
Physics, Coe College, 1220 First Avenue NE, Cedar Rapids, IA 52240

Contributed Papers

1:00

4pMU1. Estimation of tension distribution on drumheads by visualization of high-order modes using Fourier transform profilometry. Ryota Hashimoto (Dept. of Intermedia Art and Sci., Waseda Univ., 3-4-1 Ohkubo, Shinjuku-ku, Tokyo 169–8555, Japan, hashimo_108@fuji.waseda.jp) and Yasuhiro Oikawa (Dept. of Intermedia Art and Sci., Waseda Univ., Shinjuku-ku, Japan)

In this study, higher vibration modes of drumheads were visualized using Fourier transform profilometry (FTP) for simplifying drum tuning process. In our previous work, we confirmed that images of the (1,1) mode of a drumhead had some curved nodal lines due to non-uniform tension and that it provided visual cues on the state of drum tuning. However, when there were two areas with low tension on a drumhead, non-uniform tension did not curve the nodal lines. Therefore, focusing only on the (1,1) mode can lead to misinterpreting a non-uniform state as uniform. In this research, we expand the scope of visualization to higher modes in order to alleviate this problem. If nodal lines can be visualized even in higher modes, information not included in the (1,1) mode can be obtained, which will allow us to know the tuning state more faithfully. In a visualization experiment, we focused mainly on the (2,1) mode, which consists of two nodal lines. FTP-based visualization confirmed that even when the tension distribution had two areas with low tension, those nodal lines curved around each area. Based on this, it became able to estimate tension distribution that had been challenging with only the (1,1) mode considered.

1:20

4pMU2. Jet oscillation model for flute-like instruments allowing overall deflection. Seiji Adachi (School of Marine Sci. and Technol., Tianjin Univ., 92 Weijin Rd., Nankai District, Tianjin 300072, China, seiji_adachi@yahoo.co.jp)

A new jet oscillation model for flute-like instruments is proposed. This is based on a conventional semi-empirical model for a jet in half space disturbed by an acoustic cross flow. The proposed model, however, allows a jet to deflect and to oscillate as a whole with no phase delay in response to the volume flow through the instrument's window (embouchure hole or mouth). The overall jet deflection is essential to match the volume flow swept by the lateral motion of the jet to that through the window. The complete jet oscillation is the sum of the overall oscillation and that caused by fluid dynamical sinuous instability. The latter grows exponentially with a phase delay as the jet travels. The amplitude of acoustic volume velocity triggering this oscillation is much smaller than that through the window because most of it has already contributed to the overall deflection. The proposed model reasonably reproduces the recorder's head reflection coefficient measured in [Price *et al*., J. Acoust. Soc. Am. **138**(5), 3282–3292 (2015)]. Synthesis of recorder sounds using this model is underway.

1:40

4pMU3. An experiment to measure changes in violin frequency response due to playing-in. Andrew A. Piacsek (Phys., Central Washington Univ., 400 E. University Way, Ellensburg, WA 98926, andy.piacsek@cwu.edu) and Seth Lowery (Mech. Eng., Univ. of Texas, Austin, TX)

There is a widespread (but not universal) conviction among luthiers and performers of stringed instruments that a new instrument will sound "better" after it has been played for a certain amount of time. It is not uncommon for instruments to be mechanically stimulated to accelerate the process. The goal of the present experiment is to determine whether sustained mechanical stimulation of previously unplayed violins produces measurable changes in bridge admittance or radiativity. Our test instruments comprised three sibling violins (Andre Tellis, model 200), two of which were excited continuously for 12 weeks using different methods, while the third served as a control. Once per week during the excitation period, we measured bridge admittance with a laser vibrometer and radiativity in an anechoic chamber for each violin using the hammer tap method. The mass of each violin was also recorded prior to each measurement to account for fluctuations in moisture content. The resulting frequency response functions were analyzed within specific bands for changes that followed a trend over time or exceeded the standard deviation derived from a set of baseline measurements. Preliminary results do not show any significant changes in either the bridge admittance or radiativity for any of the violins.

2:00

4pMU4. Wood for guitars II : How the material properties of a spruce guitar top influence the tonal quality of the guitar. David Olson (Pacific Rim Tonewoods, Inc., 505 Ridgeway Dr., Bellingham, WA 98225, d.olson@pacificrimtonewoods.com), Sebastian Merchel (Dept. of Acoust. and Haptics, Tech. Univ. of Dresden, Dresden, Germany), and Trevor Gore (Trevor Gore Guitars, Cottage Point, New South Wales, Australia)

Pacific Rim Tonewoods is the world's largest supplier of spruce tonewood for the soundboard and bracing of the acoustic guitar. A previous study from our research group (JASA 2019) has demonstrated the importance of (1) longitudinal Young's Modulus and (2) density of the spruce soundboard on the tonal quality of steel string guitars. This study used pairwise comparisons of listener preference to rank a particular guitar model (the Taylor 814ce). The present study updates this previous work, adding a third material property: internal friction (damping) of the soundboard, as measured by the logarithmic decrement of the resonating spruce billet. In addition, objective data on tonal quality were measured on guitars recorded in an anechoic chamber at the Technical University of Dresden, Germany. Spectral sequence analysis, transfer function, and tone decay were measured, and are correlated with the perceptual data. With these data, we can demonstrate the relative importance of the three material properties on the

"performance factors" of guitar quality, such as volume, responsiveness, timbre, attack, and sustain. In doing so, we hope to provide the lutherie community with a predictable means of integrating tonewood material properties into their designs, in order to consistently produce their desired tonal outcomes.

2:20

4pMU5. Vibration mode analysis and estimation of sound power of source for acoustic tube driving by single reed. Nanako Shiono (Graduate School of Eng., Kogakuin Univ., Bldg. 5 #405, 2665-1 Nakano-machi, Hachioji-shi, Tokyo 192-0015, Japan, em22014@ns.kogakuin.ac.jp) and Yoshinori Takahashi (Kogakuin Univ., Tokyo, Japan)

Reproduction of wind instrument using additive manufacturing has been studied for the purpose of cultural property by Takahashi *et al*. It is known that the material of air reed instruments has little effect on sound quality, indicating the possibility of reproducing the Shakuhachi, a Japanese air reed instrument. On the other hand, in the case of the reproduction of classical oboe, a listening test using short musical phrases suggested the possibility of distinguishing the difference of the real instrument from that the reproduced instrument. Since the vibration of the tube caused by reed vibration is greater in reed instruments than in air reed instruments, it is considered that the sound power of the source caused by the vibration effective on the tone. In this study, a 3-D printer was used to create an acoustic tube driven by a single reed, and the sound power of the source generated by the vibration was estimated by analyzing the vibration modes generated in the acoustic tube.

2:40–3:00 Break

3:00

4pMU6. Understanding the coherence properties of musical vibrato 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), Erin Driscoll, and Mark Bocko (Elec. and Comput. Eng., Univ. of Rochester, Rochester, NY)

In the field of optics, coherence theory is used to characterize the relative correlation of optical fields at different points in space and time. While this framework has found fewer applications in acoustics outside of an imaging context, recent work has explored the possibility of characterizing rooms and other acoustical systems based on the coherence properties of naturally occurring reverberant sounds. Additionally, the cross-correlation functions used in coherence theory also underly many auditory perceptual models which leverage interaural correlations to estimate interaural time differences (ITD) used in sound localization. In these contexts, musical vibrato constitutes an important type of partially coherent signal. The quasi-periodic frequency, and often amplitude, modulation inherent to natural vibrato serves to reduce the coherence time of the sound, while being defined by a small set of relatively constant control parameters. This presentation will describe the relative impact of vibrato performance parameters, including vibrato rate and depth, on the coherence properties of the signal for various instruments. Analysis of recorded instrument samples will be compared predictions from a sinusoidal synthesis model for both the dry and reverberant cases. Finally, the potential for characterizing other musical systems, such as instruments, using coherence theory will be explored.

3:20

4pMU7. Exploring Gen in MaxMSP: Approaches to virtual analog and wavetable synthesis. Shaikat Hossain (None, 20800 Homestead Rd., Apt. 30A, Cupertino, CA 95014, shossa3@gmail.com)

The proposed poster presentation will explore using the Gen environment in MaxMSP to develop software that implements virtual analog and wavetable synthesis. Gen is a powerful tool that enables one to approach sound synthesis on an elemental level by implementing signal processing on the sample level instead of dealing with audio frames. The talk will explore both graphical patching and using the embedded text-based language Gen-Expr using the codebox operator. Focus will be placed on developing design patterns for developing efficient sound synthesis routines and learning to experiment in realtime with Gen.

3:40

4pMU8. Deep net classification of drum strike location with non-uniform membrane tension. John R. Taylor (The MARCS Inst. for Brain, Behaviour and Development, Western Sydney University, Westmead Innovation Quarter, Bldg. U, 160 Hawkesbury Rd., Westmead, Sydney, New South Wales 2145, Australia, j.taylor@westernsydney.edu.au)

Tuning inaccuracies in snare drum membranes can cause inconsistent spectro-temporal changes (e.g., timbre differences) exacerbated by drumhead strike location. This experimental study assesses the potential for developing a deep net classifier for strike drum location, using a Non-Uniformly Tuned Snare (NUTS) dataset, comprising 12,107 individual snare drum hits recorded in an anechoic chamber. Unique features of this dataset include its substantial size, additional novel features comprising physical attributes of the drum, and a performance-based classification schema. A series of deep net classifiers are trained using Bayesian hyper-parameter optimization, on a subset of 8685 strikes of the NUTS dataset, whose acoustic features are extracted at nine different averaged time-windows (23–2000 ms). A comparison is then made against similar models supplemented by domain knowledge of 15 different combinations of non-uniform membrane tension, whose measures are recorded near the sixteen tuning lugs for each strike. Results found above chance model accuracies, and some performance benefits of including the tuning information as domain knowledge between equivalent trials. Shorter window sizes performed worse than longer time windows, and per-class F1-scores suggest complex spectro-temporal time variant changes caused by dis-uniform membrane tension, which could have implications for performers wishing to broaden the sonic capabilities of the instrument.

4:00

4pMU9. The emotional characteristics of bass drums, snares, and disengaged snares with different strokes and dynamics. Zeyu Huang (Dept. of Comput. Sci. and Eng., The Hong Kong Univ. of Sci. and Technol., Rm. 110B, Tower A, HKUST, Clear Water Bay, Sai Kung, Hong Kong, zhuangbi@connect.ust.hk), Wenyi Song, Xiaojuan Ma, and Andrew B. Horner (Dept. of Comput. Sci. and Eng., The Hong Kong Univ. of Sci. and Technol., Hong Kong, Hong Kong)

Recent research has explored the emotional characteristics of pitched instruments and the voice. This study investigates the emotional characteristics of the unpitched bass drum, snare drum, and disengaged snare drum. The objective is to explore how the choice of instrument, drum strokes (e.g., single stroke and roll), and dynamics influence the perceived emotional characteristics. Listening tests were carried out with 46 participants, who provided absolute judgments on the sounds in terms of Valence and Arousal on a 5-point Likert scale, alongside the presence of 16 emotional categories: Agitated, Angry, Scary, Terrifying, Joyful, Playful, Slapstick, Exciting, Depressing, Spooky, Mysterious, Nervous, Calm, Gentle, Relaxed, and Peaceful. Data from the listening tests were analyzed using linear regression. Overall, the choice of instrument, drum strokes, and dynamics significantly and similarly influenced Arousal, whereas the choice of instrument had a much stronger effect on Valence than the other two. Specifically, the bass drum was strongly correlated with almost all negative emotions and inversely correlated with high-energy positive emotions. The type of drum strokes was relatively strongly correlated with low-energy positive categories and Agitated. Dynamics had a more secondary effect. These findings offer some detailed insights into how specific emotional characteristics are perceived for drums.

4:20

4pMU10. An automated pop song mashup system using drum swapping. Xinyang Wu (Hong Kong Univ. of Sci. and Technol., HKUST, GGT, Sai Kung, Hong Kong, xwuch@connect.ust.hk) and Andrew Horner (Hong Kong Univ. of Sci. and Technol., Sai Kung, Hong Kong)

Music mashups blend elements from two or more songs. They are extremely popular among both composers and listeners since they carry the emotional power of the original songs, plus delightful new surprises. A few automated systems have appeared on the mashup scene, but the results are often heavy-handed and muddy, since they usually combine everything

from both songs, and therefore, lack the nuance and clarity of hand-crafted mashups by professional composers. We propose a mashup system "Rhythm Fusion" that allows users to swap the drum track from one song into another. We use a beat-tracking algorithm to synchronize and align the songs. In our preliminary subjective tests, 10 songs were used to generate 100 combinations. Of these, 27% of the variations were rated more musically satisfying than the original. These results indicate that drum swapping alone has much promise for music mashups, and will guide our future work on swapping both drums and bass stems from different songs in automated music mashups, and eventually other components as well.

THURSDAY AFTERNOON, 7 DECEMBER 2023 ROOM C3.3, 1:00 P.M. TO 4:40 P.M.

Session 4pNSa

Noise: Assorted Topics on Noise

Marion Burgess, Cochair
University New South Wales, Australia, UNSW, Canberra 2610, Australia

Walter Montano, Cochair
Technical, ARQUICUST, Gualeguaychú, Entre Ríos, Argentina

Contributed Papers

1:00

4pNSa1. Methods for addressing variable turbine operations during wind farm noise compliance monitoring. Justin Adcock (Marshall Day Acoust., Collingwood, Victoria, Australia), Christophe Delaire (Marshall Day Acoust., 10/50 Gipps St., Collingwood, Victoria 3066, Australia, cdelaire@marshallday.com), Christopher Guzik (Marshall Day Acoust., Adelaide, South Australia, Australia), and Christian Mesa (Marshall Day Acoust., Collingwood, Victoria, Australia)

Post-construction wind farm noise measurements at dwelling locations are commonly required in Australia to verify compliance with regulatory and contractual noise limits. These measurements are often required as soon as the wind farm has begun operating. However, commissioning issues and electricity network restrictions are a common occurrence during the initial stages of operation. Large wind farm projects are also commonly commissioned in stages. As a result, curtailed modes of operation and shutdowns are a regular occurrence, and large sections of a wind farm may only be operating intermittently. This presents challenges to obtaining noise measurement data that is representative of normal operation of the wind farm and may ultimately limit the conclusions that are able to be reached from measurements during the early stages of operation. To address these challenges, it is necessary to identify and account for any periods when operational noise levels may be lower as a result of turbines being curtailed or shut down. Several methods have been developed for this purpose, and range from simple procedures that remove any data affected by curtailment or shutdown, through to more complex assessments of the magnitude and implications of the uncertainty introduced by variable wind turbine operations. These methods are presented and assessed in this paper with case studies of their application to operational projects.

1:20

4pNSa2. An approach to on/off testing for wind farm noise assessment. Kym A. Burgemeister (Arup, c/o Arup, 699 Collins St., Docklands, Victoria 3008, Australia, kym.burg@arup.com), Helen Searle, Jim White, Kai Fisher, Sophie Gleeson, and Tristan Desombre (Arup, Docklands, Victoria, Australia)

In Victoria, assessment of operational wind farm noise is usually undertaken using a regression approach based on NZS6808. This adopts a comparison of long-term operational sound level and wind speed measurements with background sound levels measured prior to the construction of the wind energy facility (WEF), taken at sensitive receivers near to the site. Where no background noise levels have been measured or there are changes in the environment that may make background noise levels unrepresentative, it can be difficult to positively demonstrate compliance with required noise limits, and the results can be inconclusive. On/Off testing is described in NZS6808 as an alternative to conducting sound measurements correlated with wind data, and relies on the wind farm being turned off for a short period to allow a short-term background noise level to be determined, against which the "adjacent" operational noise levels can be compared. However, the approach is not documented in detail in NZS6808 and on/off testing is rarely undertaken in Australia. This paper describes a detailed approach that has been adopted for recent on/off testing undertaken at a large WEF in Australia and outlines the key aspects of the testing which are necessary to conduct a meaningful assessment.

1:40

4pNSa3. Assessing noise impacts from light aircraft. Martin P. Tonner (Acoust., WSP Australia Pty Ltd, Level 2, 121 Marcus Clarke St., Canberra, Australian Capital Territory 2601, Australia, Martin.tonner@wsp.com)

Typically in Australia, aircraft noise is assessed against the Building Site Acceptability criteria set out in Australian Standard 2021:2015 *Acoustics-Aircraft noise intrusion-Building siting and construction* (AS 2021), which uses the Australian Noise Exposure Forecast (ANEF) system for land use planning around airports. However, the ANEF system and AS 2021 are not intended to present information about potential impacts from aircraft noise (such as from light aircraft operations) that may be experienced within the community and are considered deficient as a useful tool for describing aircraft noise to potential noise sensitive receivers. This paper discusses a methodology that may be used to better subjectively assess the environmental noise impacts from light aircraft taking into account the issues related to light aircraft operations, such as the level, duration and tonal content of noise from the aircraft. The paper will focus on observations made on a recent study of noise monitoring undertaken in March 2023 for light aircraft trial flights in Canberra ACT. This paper discusses a methodology that may be used to better subjectively assess the environmental noise impacts from light aircraft taking into account the issues related to light aircraft operations, such as the level, duration and tonal content of noise from the aircraft. The paper will focus on observations made on a recent study of noise monitoring undertaken in March 2023 for light aircraft trial flights in Canberra ACT.

2:00

4pNSa4. Coal mining noise in Australia. Roman Haverkamp (None, 80 Williams St., Woolloomooloo, New South Wales 2011, Australia, Roman.Haverkamp@rwdi.com) and John H. Wassermann (None, Woolloomooloo, New South Wales, Australia)

Noise pollution in open-cut coal mines in NSW is of great concern and affects large numbers of local communities. The management of noise from coal mines has evolved considerably over the last 30 years, in part due to the significant improvement in mobile fleet sound power levels due to new technology and leading practice noise attenuation components. The mining industry in Australia has also developed an innovative way of managing environmental noise driven by the use of proactive noise management systems. Most large open-cut mining operators in NSW have in recent years adopted and successfully implemented proactive noise management systems to assist with managing noise levels during periods of noise-enhancing meteorological conditions. The system allows the mine operator to prepare and adjust operations to reduce noise levels as far as reasonably and feasibly practical in the event anticipated adverse meteorological conditions are experienced. This paper describes the evolution of the regulatory process, the industry response and the key factors that have shaped the changes.

Invited Paper

2:20

4pNSa5. Lab-scale research on acoustically transparent stacks for hot exhaust jets in cooler cross-flow. Orddom Leav (Defence Sci. Technol. Group, DSTG, Fishermans Bend, Victoria 3207, Australia, orddom.leav@defence.gov.au), Benjamin Cazzolato (The Univ. of Adelaide, Forestville, South Australia, Australia), and Carl Howard (School of Elec. and Mech. Eng., The Univ. of Adelaide, Adelaide, South Australia, Australia)

From the exhaust stack of open cycle gas turbine power stations, the sound radiated is strongly refracted in the near-field by the thermal and velocity gradients of the jet. Research undertaken by the authors has identified that the presence of a cooler cross-flow leads to a bent-over plume downwind of the exhaust stack and a significant increase in sound in the far field by up to 10dB downwind when compared to spherical spreading. This paper is the first in a series that explores a unique "acoustically transparent silencer," which prevents this by separating the sound from the flow. The concept of the approach is demonstrated in this paper, with aero-acoustic simulations conducted on a supercomputer and small-scale experiments in a wind tunnel. The numerical simulations have predicted the effects of using an acoustically transparent exhaust stack to reduce the refraction of sound in the near field, which reduced the lobes in the acoustic directivity downstream of the exhaust by up to 4.5 dB. This was further supported by experimental results in a wind tunnel with a reduction of up to 4 dB. These results were used to assist in the second iteration of designs for the Refrastack in a subsequent paper.

Contributed Papers

2:40

4pNSa6. A review of corona and aeolian noise associated with overhead transmission lines. Tom Evans (Resonate Consultants, Level 4/440 Elizabeth St., Melbourne, Victoria 3000, Australia, tom.evans@resonate-consultants.com) and Lachlan Newitt (Resonate Consultants, Adelaide, South Australia, Australia)

As part of the significant changes occurring in the electricity networks in Australia, there is a need to create significant new transmission infrastructure including in areas relatively close to noise-sensitive land uses. This creates a risk that noise associated with transmissions lines, namely, corona noise and aeolian noise, may occur at a sufficiently high level to result in complaints and compliance risk for the new infrastructure. This paper reviews the mechanisms that can lead to audible corona noise and aeolian noise from transmission lines, as well as information on the nature of the noise including the level and character in the context of legislative requirements that apply in Australia. Guidance is provided on measures to assess and manage the risk associated with it during the design of the infrastructure. Case studies from recent projects are presented where corona noise and aeolian noise has occurred.

3:00–3:20 Break

3:20

4pNSa7. Measurements of ship noise using an acoustic camera: A first survey. Augusto Bocanegra, Davide Borelli (Dept. of Mech. Eng., Univ. of Genoa, Genoa, Italy), and Corrado Schenone (Dept. of Mech. Eng., Univ. of Genoa, Via Opera Pia 15/A, Genoa I-16131, Italy, corrado.schenone@unige.it)

While the acoustic impact of harbors is becoming an increasingly important issue, to the point of limiting their growth, the measurement of noise from ships and port operations is still rough. Two of the main problems that are commonly encountered are source overlapping and large distance from the emitting sources. Source overlapping prevents from identifying the specific contribution coming from each acoustic source. The large distance from the emitting sources, for instance the funnel of a ship or the engine of a transtainer crane, makes the measurement inaccurate. By using the acoustic camera, it is possible to tackle these two problems at the same time. Noise measurements can be made with equipment far from noise source,

while distinguishing the contribution from each single source emitting within the sound field. The paper demonstrates this enhanced measurement technique through a set of surveys performed inside French and Italian ports. The ship emissions are analyzed through the acoustic camera for ship on the way, maneuvering inside the harbor, and berthed at wharf. The effectiveness of beamforming technique is discussed, investigating the measurements accuracy, their qualitative and quantitative significance, and the possibility to introduce specific measurement standards for ships based on this methodology.

3:40

4pNSa8. Active sound transmission reduction across acoustic window.
Pak Yeung Chan (BEEE, The Hong Kong Polytechnic Univ., Rm. ZS867, 8th Fl., Block Z, The Hong Kong Polytechnic University, Hong Kong 999077, Hong Kong, 22096252r@connect.polyu.hk), Chi-Chung Cheung (EEE, The Hong Kong Polytechnic Univ., Hong Kong, Hong Kong), Shiu Keung Tang (School of Eng., The Univ. of Hull, Hull, United Kingdom), and Kwok Wai Mui (BEEE, The Hong Kong Polytechnic Univ., Hong Kong, Hong Kong)

In this study, the experimental method of transfer functions is employed to investigate the possibility of improving the sound transmission loss of acoustic windows using an active control technique. Twenty-five microphones, 20 of them located evenly over the acoustic window's indoor exit (provide error signals) and 5 fixed in the receiver room (capture the transmitted sound), and 19 small loudspeakers were used in the experiments. The loudspeakers acted as the canceling sources, and they were located along the window's interior perimeter (except the bottom part). The receiver room was made relatively "dead" using fiberglass linings. A loudspeaker array located in the source room acted as the primary sound source. The active control is implemented by minimizing the squared sound pressures at selected error microphones. Results show that the active noise control method can improve the sound transmission loss across acoustic window. The performance depends on the arrangement of the cancelling sources and the error microphone locations. It is found that an average of between 6 and 8 dB reduction of sound transmission within the frequency range from 100 to 1000 Hz can be achieved by an active control system with 4 or 6 error microphones and only three cancelling loudspeakers.

4:00

4pNSa9. Validation of an image source method room acoustics software in a highly reverberant room with a directional impulsive noise source.
Steven C. Campbell (U.S. Air Force Res. Lab., 2610 Seventh St., Wright Paterson Air Force Base, OH 45433, steven.campbell.30@us.af.mil), Alan T. Wall, Frank S. Mobley (U.S. Air Force Res. Lab., Wright-Patterson AFB, OH), Gregory Bowers (Ball Aerosp., Fairborn, OH), Mason A. Reeves (Ball Aerosp., Wright-Patterson AFB, OH), and Reese Rasband (Ball Aerosp., Fairborn, OH)

Exposure to impulsive noise sources is common during routine operations military contexts. This type of noise exposure is prevalent in training procedures carried out with small caliber firearms in both indoor and outdoor settings. Typically, hearing protection is required to mitigate hearing damage risk to military personnel. Consequently, proper prescription of hearing protection required bioenvironmental engineers and industrial hygienists to travel and collect data *in situ* which can be time consuming and expensive. As such, the researchers at the 711th Human Performance Wing of the United States Air Force Research Lab have developed a software named the Shooting Range Impulse Noise Calculator (ShRINC). The ShRINC software provides the capability to simulate high fidelity acoustic levels within indoor and outdoor shooting ranges with directional impulsive noise sources. This is accomplished by leveraging principles from Image Source Room (ISM) based room acoustics and combining them with weapon source models with complicated directivity patterns. In this work, validation will be completed by comparing results collected in a highly reverberant facility with the M4A1 firearm operating blank rounds. An investigation into the proper order for the ISM solver will be analyzed and sound exposure levels between the simulation and measured data will be compared.

4:20

4pNSa10. Sound visualization of impact noise using parallel phase-shifting interferometry and development of noise reduction attachment.
Takumi Yamazaki (Dept. of Intermedia Art and Sci., Waseda Univ., 3-4-1 Ohkubo, Shinjuku-ku, Tokyo 169-8555, Japan, takuyama@fuji.waseda.jp), Daiki Sakanoue (Dept. of Intermedia Art and Sci., Waseda Univ., Tokyo, Japan), Yasuhiro Oikawa (Dept. of Intermedia Art and Sci., Waseda Univ., Shinjuku-ku, Japan), and Shoichi Kiya (Komatsu, Ishikawa-ken, Japan)

Impact wrenches are one of the most commonly used tools in factories, and the noise generated by their use is loud. In order to improve the environment for workers, it is desirable to reduce the generation of such noise. In this study, we characterize the sound generated during the use of impact wrenches, one of the noise sources in factories, and propose a method to reduce the generated sound. Specifically, we first made visualization measurements of sound generated by the impact wrench using a high-speed polarization interferometer. As a result, a loud noise was generated from the part where the socket at the tip of the impact wrench contacts the fastening part at the same time as it rotates, and the vibration was transmitted to the fastening part, causing the noise. Next, we developed a sound deadening attachment to be attached to the impact wrench. We designed a sound-absorbing mechanism that matches the frequency band to which humans are most sensitive and the characteristic frequency generated when tightening the impact wrench, and also designed a vibration-absorbing mechanism using anti-vibration rubber and vibration-damping sheets. As a result, the characteristic metallic sound was reduced and the noise level was also reduced.

Session 4pNSb

Noise: Drone Noise

Xin Zhang, Chair
Department of Mechanical and Aerospace Engineering, Hong Kong University of Science and Technology, Kowloon, Hong Kong

Chair's Introduction—1:55

Invited Papers

2:00

4pNSb1. Ducted fan noise control by turbulence grids for drones. Ruichen Wang, Denghui Qin (Peking Univ., Beijing, China), and Xun Huang (College of Eng., Peking University, Beijing 100871, China, huangxun@pku.edu.cn)

A small ducted fan with a rotor-stator assembly is being tested in an anechoic chamber at Peking University. To attenuate fan noise, turbulence grids have been inserted before the rotor to eliminate large-scale ingested flow structures and, therefore, to minimize the flow interactions between the rotors and stators. Several different grids with various geometry and solidity configurations have been considered in the tests. To demonstrate the proposed noise control method, both near-field in-duct on-surface microphones and far-field microphones are used to measure acoustic performance. In addition, a microphone array is deployed to acoustic images at several working conditions. The corresponding thrust performance, with and without the grids in between the rotors and stators, is being measured simultaneously. The possible fluid mechanics is presumably caused by the reduced strength of ingested turbulence structures. More results help to reveal this physical understanding will be given in the final manuscript. Through this work, we hope to investigate a new strategy that could possibly reduce UAV fan noise at several representative working conditions.

2:20

4pNSb2. Near-field sound propagation and the installation effects of an urban air mobility vehicle. Yuhong Li (The Hong Kong Univ. of Sci. and Technol., Hong Kong, Hong Kong, ylilg@connect.ust.hk), Sinforiano Cantos, Etienne Spieser, Zhida Ma, Peng Zhou (The Hong Kong Univ. of Sci. and Technol., Hong Kong, Hong Kong), and Siyang Zhong (The Hong Kong Polytechnic Univ., Hong Kong, Hong Kong)

A unique challenge of the urban air mobility noise assessment is the configuration-dependent acoustic signature of different designs, of which the installed noise sources must be considered. This work presents a computational aeroacoustics study of a four-rotor urban air mobility vehicle, focusing on near-field propagation considering the installation effects of different vehicle components. The aerodynamic noise sources are computed using the delayed detached eddy simulations, in which the interactional aerodynamics of the rotating propulsion sources and the airframe are addressed. The acoustic propagation effects are calculated by solving Pierce's equation using the computed mean flow fields and the surface noise sources. The proposed approach is expected to capture the noise scattering from the rotor supports and the fuselage, which cannot be simulated by conventional computational fluid dynamics. The near-field results are then extrapolated to the far field and compared with those from the Ffowcs-Williams and Hawkings acoustic-analogy-based solver. This research provides an integrated aerodynamic and acoustic effects evaluation which should benefit low-noise vehicle design.

Contributed Papers

2:40

4pNSb3. The acoustics of multirotor platforms for urban air mobility. Charles Tinney (Appl. Res. Labs., Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, charles.tinney@arlut.utexas.edu), John Valdez, and Irene Zhao (Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX)

The sound field of hovering rotors has generated tremendous interest in recent years with the prospect of urban air mobility platforms comprising eVTOL type designs. In this presentation, we will review a number of key advancements that have been made with regard to an understanding of the sound field produced by multirotor systems in hover and how it relates to the aerodynamic performance of the rotor. Much of this work is based on laboratory-scale measurements of different configurations starting with multirotor drones comprising quad-copter, hexacopter, and even octocopter arrangements (Tinney and Sirohi, AIAA J., 2018) followed by stacked corotating rotors (Valdez and Tinney, AIAA J., 2022) where the trade-space based on stacking distance, index and angle and rotor speed are evaluated. Of particular interest is the constructive and destructive interference of sound waves from neighboring rotors which augment the first few fundamental rotor harmonics. Methods for characterizing the first few rotor harmonics based on POD-based Vold-Kalman order tracking filters (Tinney *et al*., Int. J. Aeroacoust., 2023) is presented as well as methods for collapsing the high frequency rotor broadband noise using a moving source model (Tinney *et al*., AIAA Paper 2023-3222).

3:00

4pNSb4. Acoustic signature of sUAS rotors undergoing edgewise forward flight. Justin Malkki (Univ. of New South Wales, Sydney, New South Wales, Australia, j.malkki@unsw.edu.au), Yendrew Yauwenas, Con Doolan, and Danielle Moreau (Univ. of New South Wales, Sydney, New South Wales, Australia)

This presentation shares preliminary experimental results from a new sUAS rotor test rig installed in UNSW's Anechoic Wind Tunnel (UAT). This new test rig allows acoustic and aerodynamic performance testing of small drone-scale rotor blades under edgewise forward flight conditions at a range of shaft angles. The study of sUAS rotors under edgewise forward flight presents asymmetric inflow to the rotor and unsteady loading on the blades themselves, with different flow conditions experienced by the advancing and retreating rotor blades. This means the acoustic signature of a rotor undergoing edgewise flight is significantly changed compared to static thrust or axial flight conditions of drone rotors or propellers. This presentation aims to outline changes to the acoustic signature of a collection of conventional off the shelf (COTS) rotor and propeller designs between static, axial flight and edgewise flight conditions, coupling this with rotor performance metrics.

3:20

4pNSb5. Acoustic effects of drone flights during sudden decelerations. Julia Treichel (Noise Abatement, Noise Impact, German Environment Agency, Wörlitzer Platz 1, Dessau-Roßlau 06844, Germany, julia.treichel@uba.de), Teemu Joonas Lieb, Andreas Volkert (Inst. of Flight Guidance, German Aerosp. Ctr., Braunschweig, Germany), and Jan Foerster (Inst. of Aeronautics and Astronautics, TU Berlin, Berlin, Germany)

With introduction of U-Spaces into European urban airspaces, unmanned aircraft traffic is expected to increase in the coming years. The growing number of unmanned aircraft, common known as drones, raises the question of the noise impact of these vehicles. So far, this new noise source has only been investigated acoustically in a rudimentary way. The German Environment Agency has, therefore, launched a measurement campaign with various small drone models to investigate acoustic effects. Since drone flights must be designed to be safe, a measurement campaign was conducted in which the models performed a full braking maneuver just before the measurement microphone. This is the case for collision avoidance or when strong gusts of wind occur. In addition, binaural measurements were recorded to provide psychoacoustic findings, as these parameters may be related to the degree of annoyance perceived by the public. In this paper, the measurements and results are presented. Noise levels of the overflights are compared with noise levels of the deceleration maneuvers. In addition, psychoacoustic quantities such as loudness or sharpness are used and compared for both maneuvers. Finally, conclusions are drawn for the evaluation of drone noise, which can be used for future noise regulations or standardization.

3:40

4pNSb6. Noise assessment and low-noise flight path planning platform for urban air mobility. Qichen Tan (Mechanial and Aerosp. Eng., Hong Kong Univ. of Sci. and Technol., 5013, CYT Bldg., HKUST, Clear Water Bay, Hong Kong 999077, China, qtanaf@connect.ust.hk), Yuhong Li, Hongsen Bao, Zhida Ma, Han Wu, Peng Zhou (Mechanial and Aerosp. Eng., Hong Kong Univ. of Sci. and Technol., Hong Kong, Hong Kong), Siyang Zhong (The Hong Kong Polytechnic Univ., Hong Kong, Hong Kong), and Xin Zhang (Mechanial and Aerosp. Eng., Hong Kong Univ. of Sci. and Technol., Kowloon, Hong Kong, Hong Kong)

The rapidly emerging market of urban air mobility has raised considerable public concern about noise exposure and annoyance, which is likely to limit the development of future aerial transportation. This study presents a comprehensive approach to evaluate the flight noise of an urban air mobility vehicle under various operating conditions and to provide practical flight strategies for low-noise path planning. Two methods with different fidelity levels are considered and compared in evaluating sound sources. The first approach is based on an efficient semi-empirical model to predict the rotor noise and a boundary element method solver to account for the noise scattering from the vehicle fuselage. The second approach is based on high-fidelity computational aeroacoustic simulations considering a practical full-scale urban air mobility vehicle. Then, a Gaussian beam method is used to calculate the far-field sound propagation considering practical physical phenomena in complex environments, including sound refraction attenuation in the atmosphere and reflection from obstacles. A low-noise flight planning strategy is proposed using a heuristic method to optimize the vehicle flight path in realistic urban scenarios. The proposed methods should contribute to future green aerial transportation in low-altitude operations.

4:00

4pNSb7. Construction of source description database for machine learning analysis. Frank S. Mobley (Human Effectiveness Directorate, U.S. Air Force, 2610 Seventh St., Bldg. 441, Wright-Patterson AFB, OH 45433, frank.mobley.1@us.af.mil), Steven C. Campbell (Human Effectiveness Directorate, U.S. Air Force, Wright-Patterson AFB, OH), and Mason A. Reeves (Ball Aerosp. Inc., Wright-Patterson AFB, OH)

A large set of acoustic measurements were conducted in conjunction with Sinclair Community College Center for UAS Training and Certification at the Flying Pavilion in Dayton, Ohio. Ten aircraft were characterized at multiple altitudes above a circular array placed on the ground. A subset of these data, with two biologic signals, were presented to a sound jury who provided pairwise similarity judgments. These human perception data were organized into a similarity matrix that then passed through a multi-dimensional scaling analysis. The output of the scaling analyses were examined for each subject and the average across the data collection. This dimensional analysis provides a basis for machine learning analysis to determine the salient features.

Session 4pPA

Physical Acoustics: Acoustical Methods for Materials Analytics

Dimitrios Bessas, Cochair
Experimental Division, European Synchrotron Radiation Facility, 71 avenue des Martyrs, Grenoble 38000, France

Nobuto Kaitoh, Cochair
Industrial Technology and Innovation, Tokyo University of Agriculture and Technology, Nakacho 2-24-16, Koganei-shi 184-8588, Japan

Invited Paper

1:00

4pPA1. Acoustically stimulated electromagnetic method for sensing electric and magnetic properties. Nobuto Kaitoh (Biomedical Eng., Tokyo Univ. of Agriculture and Technol., Nakacho 2-24-16, Koganei-shi, Tokyo 184-8588, Japan, n-kaitoh@st.go.tuat.ac.jp) and Kenji Ikushima (Biomedical Eng., Tokyo Univ. of Agriculture and Technol., Koganei-shi, Tokyo, Japan)

A measurement technique for detecting acoustically induced polarization is introduced. Ultrasonic irradiation can generate alternating electric or magnetic polarization in materials via electromechanical or magnetomechanical coupling, respectively. It follows that electromagnetic fields are often emitted to the surrounding environment when materials are acoustically stimulated. The first harmonic response of the acoustically stimulated electromagnetic (ASEM) field is detected by a resonant antenna tuned to the ultrasound frequency. Ultrasound can temporally modulate the magnetic polarization (magnetization) in ferromagnetic materials, resulting in magnetic imaging and magnetic hysteresis measurements via ultrasonic stimulation. Ultrasonic probing of local magnetic properties gives unique magnetic measurements and is a promising tool in steel inspection. Furthermore, the ASEM response is generated in not only inorganic crystals but also biological tissues such as bones, tendons, and the aortic wall. The response signal is well explained by stress-induced electric polarization of biological tissues, which depends on the crystallinity of fibrous proteins. Therefore, the ASEM method opens possibilities for unique noninvasive sensing in medical fields. In this talk, we will discuss the origin of the ASEM response in various materials and its applications including steel and human measurements.

Contributed Papers

1:20

4pPA2. Microstructure quality assessment for hybrid additive manufactured Ti6Al4V components via ultrasonics. Luz D. Sotelo (Purdue Univ., 585 Purdue Mall, West Lafayette, IN 47907, lsotelo@purdue.edu), Cody Pratt (Univ. of Nebraska-Lincoln, Lincoln, NE), Rakeshkumar Karunakaran, Michael P. Sealy (Purdue Univ., West Lafayette, IN), and Joseph A. Turner (Mech. and Mater. Eng., Univ. of Nebraska-Lincoln, Lincoln, NE)

Metal components with functionally organized microstructures for specific applications are emerging thanks to hybrid additive manufacturing (AM). The customization of these high value components accentuates the need for nondestructive methods to characterize their microstructural functional patterns. Nondestructive evaluation (NDE) methods that are economical, fast, energy efficient, and easy to integrate into routine component inspections are preferred. Most importantly, NDE methods must be sensitive to changes in the microstructure such that regions that do not satisfy the design requirements (i.e. out-of-spec regions) can be detected. In this work, ultrasonic NDE methods grounded in diffuse backscatter modeling were used to detect and quantify spatial property variations resulting from a hybrid AM process. The manufacturing process coupled directed energy deposition (DED) with milling in a cyclical manner. These methods were successfully implemented to evaluate the microstructural uniformity of Ti6Al4V samples as well as to make comparisons across an ensemble of samples manufactured with identical parameters. Out-of-spec regions were mapped with respect to the sample geometry on a layer-by-layer basis. The results of this work are expected to inform future NDE strategies for both research and practitioner contexts, and limitations are discussed.

1:40

4pPA3. Development of a thermal aging model for sound-absorbing polyurethane foam. Sung Soo Yang (Mech. Eng., Seoul National Univ., Gwanakro 1, Gwanakgu, Seoul 08826, Korea (the Republic of), ssyang10@snu.ac.kr), Yeon June Kang (Mech. Eng., Seoul National Univ., Seoul, Korea (the Republic of)), Jung Wook Lee, and Sung Hwan Park (Hyundai Motor Co., Gyeonggi-do, Korea (the Republic of))

This research aims to establish a reliable thermal aging model of sound-absorbing polyurethane foam (PUF). To achieve this objective, accelerated testing was conducted in a heat chamber under various conditions. The transport parameters (open porosity, high-frequency limit of the dynamic tortuosity, viscous characteristic length, thermal characteristic length, and static thermal permeability) of original and aged PUF were characterized and investigated. Curve fitting was performed to predict the value of each transport parameters corresponding to the level of thermal aging from the collected data. The thermal aging model of PUF was constructed using the Layton model and Arrhenius equations, which showed good agreement with the measured data. By employing the proposed thermal aging model, it became possible to forecast the acoustic behavior of PUF at different aging temperatures and aging periods. Subsequent testing of PUF sample materials confirmed the validity and efficiency of the developed aging model. These

findings provide a crucial reference for creating a predictive model for the acoustic behavior of PUFs undergoing thermal aging and offer valuable insights for designing an accelerated testing approach.

2:00

4pPA4. Optimal design of fibrous materials for sound absorption. Tao Yang (Tech. Univ. of Munich, Boltzmannstrasse 15, Munich 85748, Germany, tao.yang@tum.de), Marcus Maeder, and Steffen Marburg (Tech. Univ. of Munich, Garching, Bavaria, Germany)

Fibrous materials, which are typical porous materials, play a crucial role in noise control and improving sound quality in the building and automotive industries. Researchers have extensively investigated the acoustic properties of fibrous materials, including both natural and synthetic fibers. Additionally, various numerical and semi-phenomenological models have been employed to accurately predict their acoustic properties. Despite the existence of accurate models, most research on the acoustic properties of fibrous materials has been experiment-oriented. Furthermore, very few studies have focused on reliable and straightforward approaches to optimize these properties and guide the preparation process of fibrous materials. This work aims to develop a robust optimization approach for single-layer and multi-layer fibrous materials by combining numerical optimization methods with numerical and semi-phenomenological acoustic models. The ultimate goal of this optimization approach is to provide designed parameters for fabricating fibrous materials with excellent sound absorption properties. To validate and refine the optimization approach, the fabricated fibrous materials will be thoroughly measured.

2:20

4pPA5. An improved inversion method for shale elastic parameters measurement based on ultrasonic resonance spectroscopy. Chao Li (Inst. of Acoust., Chinese Acad. of Sci., No. 21 North 4th Ring Rd., Haidian District, 100190 Beijing, China, Beijing, Beijing 100190, China, chaoli@mail.ioa.ac.cn), Peng Wang, Hao Chen, and Yuyu Dai (Inst. of Acoust., Chinese Acad. of Sci., Beijing, China)

The accurate measurement of elastic parameters of shale cores has important guidance for the exploration and development of shale oil and gas. The traditional shale acoustic experiment requires multiple samples from different angles and its preparation process is complex. Resonant Ultrasound Spectroscopy has the advantages of repeatable measurement, high accuracy, and the ability to obtain all elastic parameters from a single small sample. However, the influence of clamping force and measuring angle needs comprehensive analysis to ensure that formant can be measured as many and accurately as possible. This research first conducts sensitivity analysis on the elastic parameters of different shale to guide the experiment. In addition, due to the more complex inversion objective function of the anisotropic shale compared to isotropic samples, the traditional Levenberg–Marquardt method has a strong dependence on the initial value and is prone to falling into local optimal solution. The L-BFGS (limited memory BFGS) method is introduced into the inversion of anisotropic elastic parameters of shale, which can significantly improve the inversion efficiency. Finally, the robustness and reliability of the inversion method are validated through theoretical and experimental measurement results. [Work supported by the National Natural Science Foundation of China (42004117 and 42127807).]

2:40–3:00 Break

Invited Paper

3:00

4pPA6. Determination of sound velocity by resonant ultrasound spectroscopy and complementary inelastic scattering. Dimitrios Bessas (Nuclear Resonance Beamline, European Synchrotron Radiation Facility, 71 Ave. des Martyrs, Grenoble 38000, France, bessas@esrf.fr)

In this talk, I am going to highlight the complementarity of resonant ultrasound spectroscopy and (nuclear) inelastic scattering of synchrotron radiation for determining the sound velocity in condensed matter systems. Following a very brief introduction, characteristic examples of condensed matter systems relevant to material science in which the sound velocity is extracted using both experimental methods at ambient pressure and low temperature will be shown. The strengths of (nuclear) inelastic scattering of synchrotron radiation for obtaining the sound velocity at extreme pressure (up to ca. 200 GPa) and temperature (up to ca. 2000 K) relevant to geosciences will be discussed and relevant examples will be presented.

Contributed Papers

3:20

4pPA7. Transport theory and the founding of condensed matter physics. Woon S. Gan (Acoust. Technologies Singapore Pte Ltd, 33 Oxford Rd. #04-03, Singapore 218816, wsgan5@gmail.com)

In 1966, Woon Siong Gan coined and invented the name transport theory in condensed matter physics. Today, transport theory is the backbone theory of condensed matter physics and the whole condensed matter physics can be represented by transport theory. His PhD thesis pioneered the application of statistical mechanics to ultrasound propagation in semiconductor in the presence of high magnetic fields and low temperatures with the phase transition from the spherical energy surface of metal to the warped energy surface of semiconductor. The usual treatment is using the many-body theory of quantum field theory. Phase transition is an important topic in condensed matter physics. Thus, his PhD thesis also played a role in the founding of condensed matter physics. In this paper, transport theory is applied to phase transition, an important topic in condensed matter physics. The advantages of using transport theory is its broad coverage of both particles interaction and description of the singularity characteristics of phase transition. An example for illustration is that of magnetization which has Ising model describing spins interaction and the Lee Yang theory describing the singularity behavior of the partition function at the critical point of phase transition.

3:40

4pPA8. Higher-order scattering of high-frequency ultrasound in strongly heterogeneous microstructures. Anubhav Roy (The Penn State Univ., 409 B Earth and Eng. Sci. Bldg., The Penn State Univ., University Park, PA 16802, aroy_esm@psu.edu) and Christopher Kube (The Penn State Univ., University Park, PA)

Ultrasonic waves get scattered from grain boundaries while propagating through polycrystals, thereby experiencing attenuation and a change in wavespeed. These grain boundaries act as the sole scattering sites for

untextured aggregates of randomly oriented crystallites. In addition, porosity, texture, defects, precipitates, or inclusions also serve as scatterers when present in a microstructure. Existing analytical models rely on the first-order smoothing approximation (FOSA)-based homogenization to the mass operator series in the governing Dyson equation. They can accurately predict ultrasonic dispersion in some weakly inhomogeneous metals like Aluminum. However, while comparing against more realistic Finite Element (FE) predictions, serious discrepancies, as high as 70%, are found in the attenuation estimates for metals, like Lithium, that possess strong elastic fluctuations in their microstructures. Thus, the current model analytically investigates higher-order scattering effects corresponding to the third-order smoothing approximation (TOSA) for the first time. Current results reveal no higher-order scattering effects for Aluminum at any frequencies. However, the longitudinal attenuation estimates for Lithium are found to improve by 1.27% and 0.37% in the "stochastic" (high-frequency) and "Rayleigh" (low-frequency) regimes, respectively. The presentation will highlight the effects of including TOSA on analytical estimates of high-frequency ultrasonic dispersion in different strongly heterogeneous microstructures.

4:00

4pPA9. Characterizing high-frequency wave interaction in rigid porous bilayer materials: Insights from acoustic parameter sensitivity analysis. Mustapha Sadouki (Acoust. and Civil Eng. Lab., Khemis-Miliana University, Khemis-Miliana 44001, Algeria, mustapha.sadouki@univ-dbkm.dz), Abdelmadjid Mahiou (Theor. Phys. and Radiation Matter Interaction Lab., Blida, Algeria), and Roumaissa Oubadji (Acoust. and Civil Eng. Lab., Khemis-Miliana, Algeria)

This study aims to investigate the sensitivity of high-frequency acoustic parameters in rigid air-saturated porous bilayer materials using the equivalent fluid theory. The interaction between the fluid and solid phases of the porous medium incorporates visco-inertial and thermal exchange, characterized by two functions: the dynamic tortuosity $\alpha(\omega)$ proposed by Johnson *et al*. and the dynamic compressibility $\beta(\omega)$ proposed by Allard, refined by Sadouki for the low-frequency domain of ultrasound. Various parameters, including porosity, tortuosity, viscous characteristic length, thermal characteristic length, as well as viscous and thermal shape factors, are examined in this study. A 10% variation in these parameters is considered to assess their impact on the amplitudes of transmitted waves. The findings of this study contribute to a deeper understanding of high-frequency wave behavior in porous bilayer materials. The investigation sheds light on the sensitivity of the parameters and provides valuable insights for the design and optimization of such materials in engineering, technology, and applied sciences.

4:20

4pPA10. Cortical bone properties assessment using axially transmitted low frequency (<500 kHz) guided waves. Aubin A. Chaboty (PULÉTS, École de Technologie Supérieure, Montréal, QC, Canada), Vu-Hieu Nguyen (Laboratoire Modélisation et Simulation Multiechelle, Université Paris-Est Créteil, Créteil, France), Guillaume Haiat (Laboratoire Modélisation et Simulation Multiechelle, Ctr. National de la Recherche Scientifique, Créteil, France), and Pierre Belanger (PULÉTS, École de Technologie Supérieure, 1100 Rue Notre Dame O, Montreal, QC H3C 1K3, Canada, pierre.belanger@etsmtl.ca)

The early diagnosis of osteoporosis through bone quality assessment has been extensively studied in the past decade. Research in axial transmission using ultrasonic guided waves has shown the method to be sensitive to intrinsic properties of long cortical bone. The aim of this work is, therefore, to show the capability of low frequency guided waves to enable the inversion of dispersion curves into bone properties. The proposed inversion scheme relies on dispersion curves simulated using the semi-analytical isogeometric analysis (SAIGA) method. The model used in simulation comprised a bone phantom plate with a layer of soft tissue attached to the top surface in accordance with experimental bone phantom plates. A proprietary axial transmission multielement ultrasonic transducer specifically designed to excite ultrasonic guided waves under 500 kHz was used for measurements. Acquired data were processed using the 2-D Fast Fourier Transform to extract dispersion curves. The mechanical properties of the bone phantom plates were obtained by minimizing the difference between the experimental and simulated dispersion curves. The inversion was based on the dispersive properties of ultrasonic guided waves as well as their amplitudes. Results show a difference around 5% between the mechanical properties found with the SAIGA based inversion and those provided by the manufacturer of the plates.

4:40

4pPA11. Quantitative inversion of the elastic parameters outside the borehole based on exact Zoeppritz equations. Chao Li (Inst. of Acoust., Chinese Acad. of Sci., No. 21 North 4th Ring Rd., Haidian District, Beijing 100190, China, chaoli@mail.ioa.ac.cn), Hao Chen, Xiao He, and Yuyu Dai (Inst. of Acoust., Chinese Acad. of Sci., Beijing, China)

The acoustic remote detecting method in the borehole has developed rapidly in the past few decades. It uses the reflected waves recorded by array receivers based on monopole or dipole sources in the fluid filled borehole to image geological structures outside the well. The result can provide guidance on the exploration and evaluation of oil and gas reservoirs and mineral resources. However, the present structure imaging results lack of the elastic information, which limit the quantitative evaluation of the resources. The AVO (amplitude variation with offset) inversion theory is introduced into the analysis of borehole waves. The interested interfaces (fractures or minor faults) with a generally larger wave impedance difference makes the widely used approximate equation in the surface exploration no longer applicable. Therefore, the exact Zoeppritz equations are used as the forward equations in the inversion. To reduce the nonlinearity of inversion, the initial model combining structure tensor and mode wave velocity analysis is constructed. The simultaneous inversion of the wave velocity and density of the reflector provides a theoretical basis for quantitatively evaluating the elastic parameters of the reservoir outside the borehole. [Work supported by the National Natural Science Foundation of China (42004117 and 42127807).]

Session 4pPP

Psychological and Physiological Acoustics and Animal Bioacoustics: Comparative Models of Hearing Loss

Amanda Lauer, Cochair
Johns Hopkins University SOM, 521 Traylor, 720 Rutland Ave., Baltimore, MD 21205

Micheal L. Dent, Cochair
Psychology, University at Buffalo, SUNY, B76 Park Hall, Buffalo, NY 14260

Chair's Introduction—1:35

Invited Papers

1:40

4pPP1. The enduring importance of comparative models for understanding hearing loss. Amanda Lauer (Johns Hopkins Univ. SOM, 521 Traylor, 720 Rutland Ave., Baltimore, MD 21205, alauer2@jhmi.edu)

Comparative hearing research has historically focused on diverse organisms' solutions to sensory problems such as localizing sounds in space, responding to conspecific vocalizations, and using sound for navigation and prey capture. These studies have advanced the bioacoustics and auditory neuroscience fields for many decades, despite decreasing funding for comparative auditory research whose primary intent is not to model human communication disorders. However, animals from across many taxa can experience hearing loss, and to varying degrees, recover their hearing abilities after exposure to damaging events. The genetic diversity of these animals, species-specific variations in evolved protective mechanisms, the rich repertoire of acoustic behaviors, and differences in the capacity for self-repair present important opportunities for discovery of the fundamental mechanisms of normal and disordered hearing. No one species best models humans. It is critical that we support and promote analysis of the physiological, anatomical, and behavioral manifestations of hearing loss using a comparative approach. This presentation will highlight some of the opportunities and challenges in working with diverse species as models of hearing dysfunction.

2:00

4pPP2. The effects of noise exposures on complex sound perception by laboratory mice. Micheal L. Dent (Psych., Univ. at Buffalo, SUNY, B76 Park Hall, Univ at Buffalo, Buffalo, NY 14260, mdent@buffalo.edu), Payton Charlton (Psych., Univ. at Buffalo, SUNY, Buffalo, NY), Kali Burke (Psych., Univ. at Buffalo, SUNY, Belcamp, MD), Anastasiya Kobrina (Psych., Univ. at Buffalo, SUNY, Buffalo, NY), and Amanda Lauer (Johns Hopkins Univ. SOM, Baltimore, MD)

Using laboratory mice as models for humans with age-related or noise-induced hearing loss has been a widely accepted practice for many years. However, physiological and reflex-based methodologies are often limited in the stimuli that can be used to measure auditory processing. Behavioral tasks are informative for understanding acuity for ecologically relevant and naturalistic sounds and are more closely matched to measurements of auditory acuity by humans reporting hearing difficulties. Our laboratory has extended the utility of the aging mouse model by (1) assessing hearing across the lifespan using behavioral procedures in awake and trained mice, (2) determining how aging, blasts, and noise exposures affect hearing in mice, and (3) measuring the detection, discrimination, and categorization of complex sounds by mice including ultrasonic vocalizations. We have found differences in auditory acuity of laboratory mice across age, sex, stimulus type, noise exposure, traumatic blasts, and measurement techniques, highlighting the importance of studying complex sound perception in animal models using behavioral approaches. [Work supported by R01-DC012302 and R01-DC016641.]

2:20

4pPP3. Neural and behavioral binaural hearing impairment and its recovery following noise-induced cochlear synaptopathy. Monica A. Benson, John Peacock, Matthew D. Sergison, Dominik Stitch, and Daniel J. Tollin (Physiol. & Biophys., Univ. of Colorado School of Medicine, Aurora, CO, daniel.tollin@cuanschutz.edu)

Noise-induced cochlear synaptopathy refers to a loss of synapses in the inner ear but no change in audiometric thresholds. Animal studies have focused primarily on peripheral hearing measures to diagnose synapse loss, while suggesting binaural listening deficits such as speech-reception-in-noise result from this disorder, but have not accounted for the possible regeneration of synapses. To address this, we measured binaural physiological and behavioral function in adult guinea pigs with noise-induced synaptopathy. Physiological measurements extended to two months post noise exposure. While common audiological assessments showed temporary threshold loss, reduced evoked potential amplitudes indicative of synaptopathy and measurable binaural electrophysiological hearing deficits post exposure, all measures recovered by two months including binaural behavior. Suspected regeneration of synaptic ribbons occurred by two

months post exposure and cochlear histology revealed no synaptic loss two months post exposure. We show that the same noise exposure protocol that caused synaptic loss in prior studies causes physiological binaural processing deficits in the brainstem and that the regeneration of synapses corresponds with recovered binaural processing including behavior. Results suggest that functional recovery of ribbon synapses post a single moderate noise exposure is sufficient to restore binaural hearing abilities.

2:40

4pPP4. Birds: A unique animal model for studying hearing loss and recovery. Robert Dooling (Psych., Univ of Maryland, Baltimore Ave. College Park, MD 20742, rdooling@umd.edu), Micheal L. Dent (Psych., Univ. at Buffalo, SUNY, Buffalo, NY), and Amanda Lauer (Johns Hopkins Univ. SOM, Baltimore, MD)

The 40-year-old discovery that birds regenerate hair cells in the inner ear after damage has yet to lead to treatments for hearing loss in humans, but there are other unique ways in which birds contribute to understanding the mechanisms and consequences of hearing loss. Like humans, many birds learn their vocalizations by reference to auditory information. Deafened as adults, birds experience a degradation in vocal output. Budgerigars trained to match a specific call lose precision in call production with temporary hair cell loss but regain that vocal precision with even a partial return of hearing. Canaries bred for low-pitched songs over hundreds of years inherit hair cell abnormalities, deficits in complex sound perception, and genetic mutations corresponding to mutations underlying human hair cell abnormalities. Treatment with aminoglycosides in these birds, which causes hair cell loss followed by regeneration and recovery of hearing, leads to improved hearing in birds with these hair cell abnormalities but not in those with normal hair cells suggesting that inducing supporting cell division leads to hearing recovery. Taken together, these and other findings suggest that birds should continue to be a fertile animal model for investigating hearing loss and recovery well into the future. [Work supported by NIH.]

3:00

4pPP5. Zebrafish models of sensorineural hearing loss. Allison Coffin (Integrative Physiol. and Neurosci., Washington State Univ. Vancouver, 14204 NE Salmon Creek Ave., Vancouver, WA 98686, allison.coffin@wsu.edu)

Sensorineural hearing loss negatively impacts over 450 million people worldwide. While we know that insults such as ototoxic drugs and noise damage sensory hair cells, our understanding of damage mechanisms is incomplete. Zebrafish are an exciting vertebrate model for research on fundamental mechanisms of hair cell damage. In addition to the inner ear, zebrafish have a lateral line system containing clusters of externally located hair cells. These hair cells are homologous to vertebrate inner ear hair cells and respond similarly to ototoxic damage. Our lab uses the lateral line system to determine the ototoxic potential of COVID-19 therapies; with over 900 medications currently in clinical trials, it is highly likely that a subset of those medications causes mild hearing loss that is not detected in clinical settings. In addition, while ototoxicity is a significant concern for patients exposed to these medications, acoustic damage and age-related hearing loss are far more prevalent in the general population. Our lab developed a zebrafish model of acoustic trauma and showed that noise exposure causes synaptic damage and hair cell loss in the larval lateral line analogous to the damage observed in the mammalian cochlea. Finally, we show that aged adult zebrafish lose both lateral line and inner ear hair cells similar to hair cell loss seen in aging mammals. Collectively, our research demonstrates that zebrafish are a tractable model to understand mechanisms of sensorineural hearing loss and to test therapeutic strategies.

3:20–3:40 Break

3:40

4pPP6. Mouse models reveal P2X2 receptor-mediated hearing adaptation is an indicator of noise-induced hearing loss vulnerability. Gary D. Housley (Dept. Physiol., School of Biomedical Sci., UNSW Sydney, Sydney, New South Wales 2052, Australia, g.housley@unsw.edu.au), Prathamesh T. Nadar Ponniah, Frederic von Wegner, and Jennie M. Cederholm (Dept. Physiol., School of Biomedical Sci., UNSW Sydney, Sydney, New South Wales, Australia)

Transgenic mouse models targeting the *P2rx2* gene encoding the ATP-gated ion channel P2X2R subunit have revealed a cochlear humoral control mechanism where sustained elevation of background noise drives adaptation of hearing sensitivity over several minutes, requiring hours to reset (evident as "temporary threshold shift" in ABR thresholds, and reduction in DPOAE amplitudes; Housley *et al*. PNAS 2013, Cederholm *et al*. Purinergic Signalling 2019, Housley *et al*. Curr. Opinion. Physiol. 2020). Exposure of *P2rx2* knockout (KO) mice to sustained noise (85 dB SPL) fails to elicit the ~15 dB reduction in ABR thresholds that wildtype (WT) controls demonstrate. While acute hearing sensitivity is maintained, when these *P2rx2*KO mice are exposed to long-term moderate (75 dB SPL) white noise, they develop high frequency hearing loss not seen in WT controls. This "Purinergic Hearing Adaptation" otoprotection postulate is supported by human studies, where *P2RX2* loss of function mutations lead to autosomal dominant progressive hearing loss that is exacerbated with exposure to environmental noise (DFNA41; Yan *et al*. PNAS 2013). Thus, noise-mediated release of ATP into the cochlear partition drives sustained suppression of the "cochlear amplifier," supporting cochlear sensori-neural function across a lifetime that complements dynamic medial olivocochlear efferent-based otoprotection.

4:00

4pPP7. The biology of hearing and hearing loss: New insights from naked mole-rats into mechanisms and adaptive responses. Sonja Pyott (Univ. Medical Ctr. Groningen and Univ. of Groningen, Hanzeplein 1 BB21, Groningen 9713GZ, Netherlands, s.pyott@umcg.nl), Joëlle Jagersma, Marcel van Tuinen (Univ. Medical Ctr. Groningen and Univ. of Groningen, Groningen, Groningen, Netherlands), Ursula Koch (Free Univ. of Berlin, Berlin, Germany), Amanda Lauer (Johns Hopkins Univ. SOM, Baltimore, MD), Thomas Park, Anna Lysakowski (Univ. of Illinois Chicago, Chicago, IL), Joseph Santos-Sacchi (Yale Univ., New Haven, CT), Sudhir Kumar (Temple Univ., Philadelphia, PA), and David Ryugo (Garvan Inst. of Medical Res., Sydney, New South Wales, Australia)

Understanding the biology of hearing and hearing loss requires not only examination of the existing structure and function of the auditory system but also consideration of its evolutionary legacy. In this context, research in my group, in collaboration with others, utilizes a comparative approach to investigate hearing and hearing loss in various rodent models, including Naked mole-rats—a highly

vocal, eusocial, subterranean species exhibiting surprisingly poor hearing for reasons that long remained elusive. Our findings reveal that the comparatively poor hearing in Naked mole-rats results from a combination of factors, including the absence of cochlear amplification, disrupted hair bundles, and hair bundle proteins bearing deafness-associated amino acid substitutions. Intriguingly, evidence of positive selection in some bundle proteins suggests that altered hearing in Naked mole-rats may be an adaptive response to their subterranean and eusocial lifestyles. More recent work investigates the comparative organization of central auditory structures in Naked mole-rats to identify the mechanisms of central compensation that support auditory-vocal communication in this species despite their poor peripheral hearing. Ultimately, the Naked mole-rats serve as a naturally occurring disease model to investigate hearing loss and gain valuable insights into mechanistic approaches to treat hearing loss.

4:20

4pPP8. Clinical relevance of the cat as a model of hearing loss. James Fallon (Bionics Inst., 384-388 Albert St., East Melbourne, Melbourne, Victoria 3002, Australia, JFallon@bionicsinstitute.org), Ella Trang, and Andrew Wise (Bionics Inst., Melbourne, Victoria, Australia)

Reliable, clinically relevant, animal models of hearing loss are vital to both increase our understanding of changes that occur during hearing loss and to test the safety and efficacy of interventions designed to treat hearing loss. This presentation will outline the variety of deafening techniques than can be used in the cat to induce either profound or partial hearing loss over a range of ages (from neonatal to adult). We will also discuss the range of behavioral measures, long-term and acute electrophysiological measures (including DPOAEs, CAPs, ABRs), and histological techniques (including whole cochlea imaging) that are available when using the cat as a model that make it an invaluable resource for the hearing field.

4:40

4pPP9. Investigating the effects of hearing loss in rats and guinea pigs. Wilhelmina Mulders (School of Human Sci., Univ. of Western Australia, University of Western Australia, 35 Stirling Hwy., Crawley, Western Australia 6009, Australia, helmy.mulders@uwa.edu.au), Carlos Jimena, Jack W. Zimdahl, Jennifer Rodger, and Kristin M. Barry (School of Human Sci., Univ. of Western Australia, Crawley, Western Australia, Australia)

Experimentally induced hearing loss in laboratory animals can lead to subsequent changes in the central nervous system. These central changes following hearing loss may lead to alterations in cognition and anxiety. In our laboratory, rats and guinea pigs are used to study the effects of cochlear trauma with the selected species dependent on the parameters of interest. Guinea pigs' relatively large and easily accessible cochlea makes the species well suited for cochlear physiological and neural measurements, whereas rats have the advantage of well-established performance on a wide variety of behavioral tests. In a recent study, we investigated whether these well-established tests in rats for stress (open field test; OFT), and learning and memory (novel object recognition; NOR; Morris Water Maze; MWM) could be used in guinea pigs to investigate changes in behavior following conductive hearing loss. Analysis showed that guinea pigs could be trained in the MWM, and showed a similar range of behaviors as described for rats in OFT; however, NOR tests could not be interpreted as most guinea pigs did not interact with the novel objects. Overall, anxiety, spatial learning, and memory were not affected by 8 weeks of conductive hearing loss.

5:00

4pPP10. The hunt for hidden hearing loss—Linking animal and human studies. David McAlpine (Macquarie University Hearing, Macquarie Univ., 16 University Ave., Australian Hearing Hub, Sydney, New South Wales 2109, Australia, david.mcalpine@mq.edu.au), Jessica J. Monaghan (Signal Processing, National Acoust. Labs., Sydney, New South Wales, Australia), and Jason Mikiel-Hunter (Macquarie University Hearing, Macquarie Univ., Sydney, New South Wales, Australia)

Many otherwise normal-hearing individuals experience listening problems not apparent from their hearing thresholds. Carefully controlled animal studies using invasive techniques in a range of species have established a clear link between exposure to loud sounds and a range of pathologies underlying this *hidden* hearing loss (HHL), including cochlear synaptopathy and elevated central gain. Despite progress in these pre-clinical models, however, evidence supporting the existence of HHL in humans remains inconclusive. Here, I review studies of animal models of HHL demonstrating evidence consistent with the types of issues reported by humans who experience problems listening in background noise despite having normal to near-normal hearing thresholds. I suggest that the nature of hearing loss reported by humans—often gradual and multi-factorial—requires us to conceptualize HHL as a system-level listening problem, rather than a coding deficit in sensory processing within the inner ear. I hypothesize that listening deficits in HHL arise from the brain's homeostatic response to altered sensory input modifying total system gain to stabilize long-term neural activity. Few animal studies have approached the problem from this perspective, while homeostatic responses, such as mal-adaptation to the longer-term statistical structure of acoustic environments, are unlikely to be captured by clinical assessments.

Contributed Paper

5:20

4pPP11. Aging leads to impairment of spatial hearing abilities in the Mongolian Gerbil. Matthew D. Sergison (Physiol., Univ. of Colorado Anschutz Medical Campus, 128000 E 19th Ave., RC1 7401G, Aurora, CO 80045, matthew.sergison@cuanschutz.edu), John Peacock, Monica A. Benson, Nathaniel T. Greene (Otolaryngol., Univ. of Colorado Anschutz Medical Campus, Aurora, CO), Achim Klug, and Daniel J. Tollin (Physiol., Univ. of Colorado Anschutz Medical Campus, Aurora, CO)

Aging in humans is known to affect spatial hearing and speech in noise recognition, often even when hearing thresholds are normal. The Mongolian Gerbil (*Meriones unguiculatus*) can provide a model organism to investigate these changes. We performed auditory brainstem responses (ABRs), prepulse inhibition of the acoustic startle (PPI) behavior, and cochlear histology to assess age-related changes of the gerbil auditory brainstem. Aging (>33 month) gerbils showed reduced ABR waveform amplitudes compared to young gerbils (2-10 months). Aging gerbils also show a reduced amplitude of the binaural interaction component (BIC), a biomarker for spatial hearing. However, ABR thresholds were not significantly different between cohorts. Aging gerbils showed impaired performance in our behavior tasks: young gerbils detected shorter gaps in noise and smaller minimum audible angles than aging animals, indicating that aging impairs temporal and spatial hearing abilities. Cochlear histology revealed an increase in cochlear synaptopathy in aging animals, revealing a potential mechanism for age-related dysfunction in the auditory pathway. Taken together, these data reveal structural and physiological changes in the auditory brainstem that underlie spatial hearing deficits in the gerbil. [Work supported by R01-DC017924.]

Session 4pSA

Structural Acoustics and Vibration and Physical Acoustics: Structural Acoustics and Vibration in Buildings

Benjamin M. Shafer, Cochair
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Invited Papers

1:00

4pSA1. Investigation of alternative methods for measuring and evaluating airborne sound insulation. Wayland Dong (Paul S Veneklasen Res. Foundation, Santa Monica, CA), Benjamin M. Shafer (PABCO Gypsum, Tacoma, CA), Sunit Girdhar (Paul S Veneklasen Res. Foundation, Santa Monica, CA), and John LoVerde (Paul S Veneklasen Res. Foundation, 1711 16th St., Santa Monica, CA 90404, jloverde@veneklasen.com)

The reproducibility of airborne noise insulation testing is poor, as documented by various interlaboratory studies. This is compounded by the definition of Sound Transmission Class (STC), the dominant single-number rating used in North America, which over-emphasizes the wall performance at and around the 125 Hz third-octave band. Small variations in performance at these frequencies can result in substantial changes in STC rating. To address these concerns, changes to both the measurement method and the single number ratings may be appropriate. For the former, the authors have begun an investigation of measuring the vibro-acoustic characteristics of walls, including measuring the vibration on the surfaces of the separating assembly and the flanking assemblies. This may result in increased precision of sound insulation measurements. For the latter issue, the authors have performed statistical evaluation of published wall sound insulation tests to clarify the effects of idiosyncrasies in the definition of STC and investigated whether different rating methods may better describe the performance of wall assemblies.

1:20

4pSA2. Predicting trends in the sound transmission loss performance of steel-framed wall partitions using yield strength. Benjamin M. Shafer (Tech. Services, PABCO Gypsum, Tacoma, CA, ben.shafer@quietrock.com)

Mechanical properties of beams and panels are commonly used for calculating structural loads and deformation under stress and strain. Stress and strain analysis for steel beams may result in the determination of yield strength, a commonly reported steel framing property in the steel industry. This characteristic of steel framing is rarely, however, linked to trends in sound transmission loss (STL). This paper provides an analytical basis for linking the yield strength of steel framing to the STL of vertical double-leaf partitions with a focus on acoustic impedance of the beam. Experimental data are shown that validates this analytical basis. These results may be used as evidence for including the reporting of the yield strength in standardized test reports.

Contributed Papers

1:40

4pSA3. Noise and structural vibration assessment of the Qubit 360 partition wall system. Pablo Reboredo Gasalla (Acoust. & Vib., ACOR Consultants (Vic) Pty Ltd, Level 2, 6 Palmer Pde, Cremorne, Victoria 3121, Australia, PReboredoGasalla@acor.com.au)

High-rise buildings can be highly susceptible to the action of dynamic loads produced by the action of wind and earthquakes. Creaky noise towers can be a major, and often overlooked, consideration in the design and operation of new developments. A reliable and considered approach to assessing creaky towers is needed to achieve relevant standards and to minimize environmental impacts on buildings and structures. In collaboration with Swinburne University of Technology and Traxx Metal Framing Systems, our team tested the structure-borne noise and vibration responses of a lightweight prototype implementing a patented vibration isolation system for building movements, Qubit 360. The primary goal of this research work was to refine, develop, and test this innovative system that can be used in multistorey constructions and confirm the structural and vibro-acoustic behavior during a loading series of lateral drift simulating the movement of a building. Monitoring a structure with the Multi-Axis Substructure Testing System under the dynamics effects of different loading events allowed our team to understand the dynamic behavior of a lightweight prototype, and improve its safety, reliability, performance, and comfort. The proposed abstract will focus on an overview of the process involved in the laboratory and will

4p THU. PM

present the vibro-acoustics results for a number of different specimens that implement the isolation system.

2:00

4pSA4. The development of the AAAC Guideline for Gymnasium & Exercise Facility Assessment and case study information. Richard Haydon (USYD, 2/174 Willoughby Rd., St Leonards, New South Wales 2065, Australia, rh@acousticdynamics.com.au) and Matthew Ottley (Sydney, Marshall Day Acoust., Ultimo, New South Wales, Australia)

Association of Australasian Acoustical Consultants (AAAC) member representatives began discussing the need for AAAC guidance on impact noise from gyms in early 2016. This was in the absence of an agreed position or standard assessment criteria. It was identified that in addition to the overall target criteria the Guideline also needed to provide a standardised method to assess and test impacts and to determine compliance as well as guidance on suitable locations for the siting of gyms. This presentation provides an overview of the development of the AAAC Guideline for Gymnasium & Exercise Facility Assessment. The presentation also includes case study information to illustrate the application of the Guideline.

2:20

4pSA5. Comparison between BS 7385 and DIN 4150 and their suitability for application in high-level vibration impact assessments. Aaron Miller (Acoust. Studio, Stanmore, New South Wales, Australia), Dominik Duschlbauer (SLR Consulting Australia, 120 High St., North Sydney, New South Wales 2060, Australia, dduschlbauer@slrconsulting.com), and Joseph Spagnol (SLR Consulting Australia, North Sydney, New South Wales, Australia)

In Australia, there are two standards that are regularly applied to determine appropriate cosmetic damage criteria with respect to vibration from external sources: BS 7385-2:1993 and DIN 4150-3 (both the 1999 and 2016 versions). DIN 4150-3 is being increasingly applied by consultants and specified by regulators, primarily due to having more conservative guideline values and secondarily due to containing specific guideline values for structures that are "of great intrinsic value" (i.e., heritage buildings). However, this presents challenges when trying to undertake a high-level or preliminary vibration assessment, where the resonance frequencies of the buildings requiring assessment or the operating frequencies of the proposed construction equipment are not known. Applying the "long-term" guideline values from DIN 4150-3, which only apply in the horizontal plane of the highest floor of the building, complicates these types of assessments and presents practical challenges for vibration monitoring in practice. This paper compares BS 7385-2:1993 and DIN 4150-3 with the aim of clarifying the differences between the two standards and provides commentary on how these standards should be applied for high-level vibration assessments.

2:40

4pSA6. Comparison of human comfort vibration metrics as applied to Sydney passenger trains. Jordan McMahon (SLR Consulting Australia, North Sydney, New South Wales, Australia), Dominik Duschlbauer (SLR Consulting Australia, 120 High St., North Sydney, New South Wales 2060, Australia, dduschlbauer@slrconsulting.com), and Aaron Miller (Acoust. Studio, Stanmore, New South Wales, Australia)

This paper discusses train vibration from 400 passenger train passbys measured at two locations in the Sydney metropolitan area with respect to human comfort. The paper focuses on the assessment outcomes as they result from the application of different Standards. The considered Standards include BS 6472, NS 8176, DIN 4150.2, ISO 2631, and the VC curve framework.

3:00–3:20 Break

3:20

4pSA7. Prediction of low frequency vibration propagation from operating rail tunnels. Graham Brown (Eng. Sci., Mott Macdonald, Level 10, 383 Kent St., Sydney, New South Wales 2000, Australia, graham.brown@mottmac.com), David Timms (Eng. Sci., Mott MacDonald, Sydney, New South Wales, Australia), and David Larner (Eng. Sci., Mott MacDonald, Docklands, Victoria, Australia)

Predicting the propagation of vibrations from rail tunnels is important for quantifying and mitigating the environmental impacts of operating railways. In the general case, vibrations propagating from rail tunnels are attenuated by material damping and geometric effects. Often these effects are aggregated in empirical or numerical models which offer little insight into the physical behaviors that are involved and may be limited in terms of their applicability according to frequency range and tunnel-receiver geometry. This paper examines the aspects of low frequency vibration propagation from rail tunnels that are of relevance to predicting tactile vibration impacts and the influence of operating railway vibrations on sensitive imaging equipment. Fundamental geometric considerations and numerical models are used to determine the influence of pseudo-static effects, source-receiver geometry, layered ground and material damping effects, and the relationship between average and maximum vibration levels that are dependent upon source-receiver distance.

3:40

4pSA8. Investigating the impact of tunnel geometry on ground-borne noise and vibration from underground railways. David Timms (Mott MacDonald, Level 10, 383 Kent St., Sydney, New South Wales 2000, Australia, david.timms@mottmac.com) and Neil Mackenzie (Mott MacDonald, Adelaide, South Australia, Australia)

Existing methodologies offer a robust basis for predicting ground-borne noise and vibration effects arising from typical railway tunnel structures constructed using tunnel boring machines. However, the literature offers less clarity on how variations in tunnel geometry—such as those found in mined sections or crossover caverns—may influence these vibration levels. This paper addresses this knowledge gap, employing finite element modeling to conduct a comprehensive parametric study of variations in tunnel geometry. The research objective is to discern the influence these deviations from standard tunnel cross sections have on predictions of ground-borne noise and vibration. The results from this investigation offer valuable insights, challenging the notion that existing methodologies can be universally applied across all tunnel configurations. Our findings underscore the critical role that nuanced tunnel geometries play in shaping ground-borne noise and vibration profiles, emphasizing the need for more granular predictive models that consider these variations. This study is poised to significantly contribute to the understanding and prediction of ground-borne noise and vibration in railway systems, facilitating improved tunnel design, minimizing potential disturbances, and enhancing overall system performance.

4:00

4pSA9. Comparison of FEA and SEA methods for predicting railway-induced vibration propagation through over-station developments. David Timms (Mott MacDonald, Level 10, 383 Kent St., Sydney, New South Wales 2000, Australia, david.timms@mottmac.com), Graham Brown (Eng. Sci., Mott Macdonald, Sydney, New South Wales, Australia), and Damian McGuckin (Pacific ESI, Glebe, New South Wales, Australia)

The design process for Over-Station Developments (OSDs) includes consideration of the potential impacts of railway-induced vibrations to ensure that structure-borne noise and occupant comfort standards are maintained during railway operations. Should an OSD be identified as sensitive to railway operational structure-borne noise or vibration impacts, the

inclusion of adequate track vibration isolation measures becomes essential for mitigation. This paper presents a comparative study of two methodologies for predicting vibration propagation through structures and consequent structure-borne noise levels—Finite Element Analysis (FEA) and Statistical Energy Analysis (SEA). Specifically, this study provides an indication of the range of predicted results and emphasizes the computational efficiency and scalability of these methods when determining track isolation requirements for structures.

4:20

4pSA10. Design and development of SiteHive MEMS based system for real-time vibration monitoring. Michael M. Darroch (SiteHive, Ste. 5B, 2-12 Foveaux St., Surry Hills, New South Wales 2010, Australia, michael@sitehive.co), Ben Cooper-Woolley (SiteHive, Surry Hills, New South Wales, Australia), and Benjamin J. Halkon (School of Mech. and Mechatronic Eng., Univ. of Technol. Sydney, Botany, New South Wales, Australia)

Construction projects need to proactively manage their works that may cause vibration impacts to nearby structures and stakeholders. Risks of vibration include cosmetic and structural damage to buildings and threats to human comfort. The advent of MEMS accelerometers offers significant opportunities to improve on traditional vibration monitoring practices based on geophones. Geophones measure velocity which preclude acceleration based measurements and calculations like vibration dose value (the measure for human comfort), groundborne noise, and auto-levelling. The inability to capture these results means additional monitoring devices are required to capture all key measurements. MEMS-based vibration monitoring systems can be much cheaper, smaller, and more power efficient than traditional vibration monitoring systems. This enables easier installation, greater mobility, and more monitoring to be conducted. SiteHive has worked extensively with the National Measurement Institute (NMI) and the University of Technology Sydney (UTS) to test and validate the efficacy of the MEMS-based accelerometers and develop a calibration system for MEMS-based devices. This paper will outline the design research and findings that have gone into this development, results from field testing, and details on the value offered by this innovation.

4:40

4pSA11. Estimation of input power to track energy propagation through a small air handling unit. Matt Smither (Univ. of Kentucky, 506 Administration Dr., #151 RGAN Bldg. Rm. 006, Lexington, KY 40506, Matt.Smither@uky.edu), David Neihguk, and David Herrin (Univ. of Kentucky, Lexington, KY)

Statistical energy analysis (SEA) is dependent on having accurate inputs. The inputs to an SEA model are structural and acoustic input powers. In this work, blocked force analysis is utilized to determine the structural and acoustic input powers to a small air handling unit (AHU). An electromagnetic shaker connected to a small stiff plate bolted to a panel of the AHU is used as an input force to the unit. The acoustic input consists of a calibrated volume velocity source placed inside the air handler. The inversely determined input powers are then used as inputs to the SEA model to track the structure-borne and airborne energy propagation through the small air handler. The results demonstrate that the inversely determined input powers can be used to reliably estimate inputs for the SEA model.

5:00

4pSA12. Piping riser isolation to prevent transmission of noise and vibration in high rise structures. Douglas Valerio (Mason Industries, 350 Rabro Dr., Hauppauge, NY 11788, dvalerio@mason-ind.com)

Preventing the transmission noise and vibration in high rise construction through risers utilizing Spring Isolators is the easiest way to achieve the acoustic criteria for the structure and eliminate weak links (expansion compensators and flexible connectors) in the piping system. We first published literature on the topic in the late 1980s. By utilizing standard calculations for thermal expansion and contraction and selecting a proper anchorage point, isolating risers with spring mounts is simple. This paper is written as a design and installation guideline which addresses concept, design, estimating, and installation.

Session 4pSP

Signal Processing in Acoustics: Signal Processing Potpourri III

Manton J. Guers, Chair
Acoustics, Penn State Univ, PO Box 30, State College, PA 16804

Chair's Introduction—12:55

Contributed Papers

1:00

4pSP1. Sonar echo detection based on two-dimensional matched filtering and convolutional neural network. Meng Zhao (Inst. of Acoust., Chinese Acad. of Sci., No. 21 North 4th Ring Rd. Haidian District 100190 Beijing 100190, China, zhaomeng@mail.ioa.ac.cn), Qunyan Ren, and Li Ma (Inst. of Acoust., Chinese Acad. of Sci., Beijing, China)

In underwater acoustics, the performance of sonar echo detection is limited when the echo-to-noise ratio or echo-to-reverberation ratio is low. As an attempt to improve the echo detection rate and maintain a low false alarm rate, a method based on two-dimensional matched filtering (2-D-MF) and convolutional neural network (CNN) is proposed. The 2-D-MF divides the replica signal into multiple sub-replicas, each with different frequency components, and utilizes the sub-replicas to perform the matched filtering individually, obtaining 2-D-MF features that better represent the amplitude-frequency characteristics of the echo signal. The CNN is utilized as an echo detector to extract echo information from the 2-D-MF features and determine the presence of an echo. The proposed method is tested using data collected in the South China Sea, 2021. During the experiment, a transducer transmitted linear frequency modulation (LFM) signals, and a transponder, acting as an analog target, forwarded the LFM signals as echoes. The detection results demonstrate that this method can improve the echo detection rate by approximately 7% while maintaining a constant false alarm rate of 1‰.

1:20

4pSP2. Simulation-based training of neural networks for low-frequency sonar target classification. Yeon-Seong Choo (Korea Univ. of Sci. and Technol., Shinsung-dong, Daejeon 341166, Korea (the Republic of), chooys@kriso.re.kr), Sung-Hoon Byun, Sea-Moon Kim (Korea Res. Inst. of Ships and Ocean Eng., Daejeon, Korea (the Republic of)), and Jeong-Bin Jang (Korea Univ. of Sci. and Technol., Daejeon, Korea (the Republic of))

Artificial Intelligence (AI) is widely used in various fields, but obtaining sufficient underwater acoustic data for AI training remains challenging. Even if simulation data are used for training, the classification accuracy could be lowered due to changes in the environment around the target. This study presents the research findings on training neural networks for target classification using data generated through simulations. We trained neural networks using target scattering signals obtained through finite element analysis for various environments. Subsequently, we utilized trained neural networks to classify the spherical shells on the seabed acquired in water tank experiments. To solve the problem of lack of data for AI training, we generated training data by applying realistic variations of parameters such as sound speed of water, target material properties, and thickness. Such training data provide more meaningful information than noise-added data. The targets considered were spherical shells with and without internal spaces. Although the target classification accuracy is not yet high enough, it is expected to be used to overcome the lack of data by applying to untrained targets. [Work supported by KIMST funded by the Agency of Korea Coast Guard (KIMST-20210547) and by KRISO funded by Ministry of Oceans and Fisheries (PES4380).]

1:40

4pSP3. Classification of underwater soundscapes using raw hydroacoustic signals. Leixin Nie (State Key Lab. of Acoust., Inst. of Acoust., Chinese Acad. of Sci., No. 21 North 4th Ring Rd., Haidian District, Beijing 100190, China, nieleixin@mail.ioa.ac.cn), Yonglin Zhang, and HaiBin Wang (State Key Lab. of Acoust., Inst. of Acoust., Chinese Acad. of Sci., Beijing, China)

Automatic classification of underwater soundscapes remains an open problem, mainly due to the challenges posed by multi-source interferences. Accurately classifying anthropogenic sounds in the marine environment, specifically those emanating from surface-ships, is concerned in this study. To achieve this, the convolutional neural network (CNN) with sine-cardinal-like constrained convolutional kernels is proposed, where kernels represent unsolved filter coefficients. Raw signals of sound pressures passively received by the hydrophone are used directly in this method without the need for the routine time-frequency analysis (TFA) beforehand. The convolutional layer with constrained kernels plays a crucial role in extracting features from hydroacoustic signals, effectively acting as a learnable extension of the bandpass filter groups with flexible bandwidth. One significant advantage of the proposed approach is its adaptive filtering capability based on the real-world data, which makes it effectively filter out irrelevant interference sounds. Such heterogeneous interferences often exhibit diverse unknown spectral ranges, making them challenging for conventional TFA with fixed parameters. By leveraging this adaptability, the output of the convolutional layer serves as a task-specific spectrogram, customized to the classifier after being trained on hydroacoustic data. The experiments on the ShipsEar dataset demonstrate the promising results of our solution than TFA and vanilla CNN.

2:00

4pSP4. Seabed classification using acoustic signals: A decision tree approach. Diego Rios (Mathematical Sci., New Jersey Inst. of Technol., Newark, NJ) and Zoi-Heleni Michalopoulou (Mathematical Sci., New Jersey Inst. of Technol., 323 ML King Blvd, Newark, NJ 07102, michalop@njit.edu)

Decision trees are versatile machine learning algorithms that are frequently used in classification and regression tasks. In this work, decision trees are employed as tools for sediment classification using sound waves propagating in the ocean and their behavior and features. We first analyze the structure of received time series at deployed hydrophones, extracting characteristic features (kurtosis and skewness, for example). Feature values then form vectors that are used as input patterns to the trees. A training step is the first stage of the machine learning approach with the trees trained to recognize sediment types based on feature values. The method is

subsequently tested on feature vectors obtained from noise-corrupted time series. The performance depends on Signal-to-Noise Ratio values as expected and the method is found to be superior to conventional machine learning approaches. The addition of tools such as principal component analysis as well as spectrogram processing and time-frequency curve fitting further enhances the method. The decision tree technique provides an effective and efficient solution to the problem of sediment classification using acoustic data. [Work supported by ONR.]

2:20

4pSP5. Platform motion estimation in multiple-input multiple-output synthetic aperture sonar with coupled variational autoencoders. Angeliki Xenaki (CMRE, STO-NATO, Viale San Bartolomeo 400, La Spezia 19126, Italy, Angeliki.Xenaki@cmre.nato.int), Yan Pailhas, and Alessandro Monti (CMRE, STO-NATO, La Spezia, Italy)

Synthetic aperture sonar (SAS) utilizes the motion of the platform carrying the sonar system to synthesize an aperture that is much longer than the physical antenna by coherently combining data from several pings. Coherent processing in SAS requires platform motion estimation and compensation with sub-wavelength accuracy for high-resolution imaging. Micronavigation, i.e., through-the-sensor platform motion estimation from spatio-temporal coherence measurements of diffuse backscatter on overlapping recordings between successive pings, is essential when positioning information from navigational instruments is absent or inadequately accurate. Representation learning with a variational autoencoder (VAE) offers an unsupervised data-driven micronavigation solution. In this study, we introduce a hierarchical variational model implemented with coupled VAEs to relate the common latent features between datasets of coherence measurements in broadband multiple-input multiple-output SAS systems. We show that self-supervising the training process of independently parameterized but coupled VAEs improves significantly the accuracy of the micronavigation estimates.

2:40–3:00 Break

3:00

4pSP6. Graph Neural Networks for source localization using Ships-of-Opportunity spectrograms. Jhon A. Castro-Correa (Elec. and Comput. Eng., Univ. of Delaware, 139 The Green, 3rd Fl., Newark, DE 19716, jcastro@udel.edu), Mohsen Badiey (Elec. and Comput. Eng., Univ. of Delaware, Newark, DE), Jhony H. Giraldo (LTCI, Télécom Paris - Institut Polytechnique de Paris, Palaiseau, France), and Fragkiskos D. Malliaros (Université Paris-Saclay, CentraleSupélec, Inria, Ctr. for Visual Computing (CVN), Gif-Sur-Yvette, France)

Traditional underwater source localization methods have primarily relied on optimization techniques, matched-field processing, beamforming, and, more recently, deep learning approaches. However, these methods often fail to exploit the spatial correlation of data due to their representation in a regular domain. Nowadays, data collection commonly occurs in complex domains, like sensor networks, where signals and features are best represented as graphs based on feature similarity metrics. In a graph representation setting, each sensor or feature corresponds to a node in the graph, accompanied by a feature vector that may or may not vary with time. As edges define spatial relationships, spatio-temporal information is considered simultaneously during the learning process. This work proposes a novel graph learning module for source localization using Ships-of-Opportunity (SOO) spectrograms, which represent mid-frequency acoustic broadband signals (360-1100Hz) collected during the 2017 Seabed Characterization Experiment (SBCEX17). The proposed approach employs a semi-supervised learning algorithm on a graph constructed through a *k*-nearest neighbor (*k*-NN) algorithm, incorporating features extracted from the spectrograms using a backbone architecture. The efficacy of the proposed approach is demonstrated through model evaluation on both synthetic and measured data, validating the generalization power of the architecture. [Work supported by ONR under Grant No. N00014-21-1-2760.]

3:20

4pSP7. High-frequency source localization based on steered frequency-wavenumber analysis using sparse array. Y. H. Choi (Dept. of Ocean Eng., Korea Maritime and Ocean Univ., 727 Taejong-ro, Youngdo-Gu, Busan 49112, Korea (the Republic of), hwa1470@gmail.com), J. S. Kim (Dept. of Ocean Eng., Korea Maritime and Ocean Univ., Busan, Korea (the Republic of)), and Gihoon Byun (Dept. of Convergence Study on the Ocean Sci. and Technol., Korea Maritime and Ocean Univ., Busan, Korea (the Republic of))

In this study, we address the problem of estimating the position of a nearby target emitting a high-frequency signal using sparse array. One of the representative localization techniques, Matched Field Processing (MFP), is known to demonstrate reliable performance below 1 kHz. However, at frequencies higher than the design frequency of the array sensor, the localization becomes challenging due to spatial aliasing. Nevertheless, Frequency-Difference Matched Field Processing (FDMFP) is a technique that successfully estimates the position of the target even in the presence of spatial aliasing. It performs signal processing in the low-frequency band using low-frequency components obtained from two high-frequency components. To ensure reliability, we compared and validated the results of this study with the results of FDMFP. In this research, we completed the algorithm by controlling the phenomenon of striation shifting caused by spatial aliasing in the conventional frequency-wavenumber(f-k) analysis through beam steering. We steered the received signal to the location of signal reception and identified the target's position using the f-k spectrum of the steered signal. The data used in the study is the sound of snapping shrimp from SAVEX15, with a frequency range of 5–24 kHz wideband impulse signals.

3:40

4pSP8. Bayesian source localization in an urban environment using scattered signal distributions. D. Keith Wilson (US Army Engineer Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755, d.keith.wilson@usace.army.mil), Aaron C. Meyer, Matthew J. Kamrath, Rachel A. Romond, Cody M. Best (US Army Engineer Res. and Development Ctr., Hanover, NH), Chris L. Pettit (Aerosp. Eng. Dept., US Naval Acad., Annapolis, MD), and Vladimir E. Ostashev (US Army Engineer Res. and Development Ctr., Hanover, NH)

Acoustic signals propagating in urban environments are influenced by rough-surface scattering, multipath reflections, and diffraction. Conventional source localization algorithms often perform poorly when these effects are present. Bayesian approaches, however, are particularly well suited to incorporating physics-based statistical models for the signal propagation. Previously, we found that the complex Wishart distribution, which describes fully saturated scattered signals across a network of receivers, can be readily employed in a Bayesian framework. This approach is very general, as it includes source triangulation and trilateration as special cases. Feasibility was initially demonstrated using simulations. In the present work, we describe an experimental demonstration of Bayesian source localization using data recorded in an urban-like environment. The experimental data were collected as part of a NATO urban acoustics-seismics experiment in Walenstadt, Switzerland, in May 2023. A network of four acoustic nodes, each with 12 microphones, was deployed and recorded emissions from a variety of sources. Initial results from the data processing and localization are described.

4:00

4pSP9. Deep learning for source localization and seabed classification under dynamic oceanographic environments using explosive sound signals. Christian D. Escobar-Amado (Elec. and Comput. Eng., Univ. of Delaware, 139 The Green, Newark, DE 19716, escobarc@udel.edu) and Mohsen Badiey (Elec. and Comput. Eng., Univ. of Delaware, Newark, DE)

A deep-learning approach is proposed for simultaneous source localization and seabed classification of underwater signals generated by SUS (Signal, Underwater Sound) MK64 explosives. The charges were deployed in

two separate shallow water acoustic experiments on the New England Mudpatch region in 2017 (SBCEX17) and 2022 (SBCEX22) conducted under different oceanographic conditions. The proposed approach uses a multitask learning (MTL) methodology, and convolutional neural networks (CNNs) trained on the spectrogram representation of SUS signals in the 10–199 Hz frequency band where the time-frequency modal dispersion curves exhibit distinct patterns. CNNs were trained on simulated data generated using the normal modes model, ORCA. This synthetic dataset is composed of several sound speed profiles measured in the water column in both experiments and representative seabeds ranging from muddy to sandy sediments. CNNs were tested on the signals measured in the two experiments where the source localization achieved a low mean squared error (0.35–0.8 km), and the seabed classification closely matched existing inversion results in the Mudpatch area. Notably, our method effectively handled oceanographic variations, showcasing the utilization of modal dispersion for training deep learning algorithms. [Work supported by ONR Grant N00014-21-1-2760.]

4:20

4pSP10. Beam-domain deconvolution beamforming algorithm based on compressive sensing. JianLi Huang (State Key Lab. of Acoust., Inst. of Acoust., Chinese Acad. of Sci., Beijing, China), Yu Wang (State Key Lab. of Acoust., Inst. of Acoust., Chinese Acad. of Sci., 21 North Fourth Ring West Rd., Haidian District, Beijing 100190, China, wy@mail.ioa.ac.cn), ZaiXiao Gong, Jun Wang, and HaiBin Wang (State Key Lab. of Acoust., Inst. of Acoust., Chinese Acad. of Sci., Beijing, China)

Deconvolution beamforming is often used to improve the azimuth resolution in target detection. However, most traditional deconvolution beamforming algorithms require the array directivity function to be shift-invariant, which are suitable for specific arrays such as linear arrays and circular arrays. A beam-domain deconvolution beamforming method suitable for arbitrary array is proposed based on compressive sensing. First, conventional beamforming is used to obtain several complex beam outputs, then the Sparse Bayesian Learning (SBL) reconstruction algorithm is applied to the beam-domain model to achieve deconvolution of complex beam outputs. The deconvolution process leads to more accurate estimation for the Direction Of Arrival (DOA) for target signals. The proposed method can also effectively reduce the computational complexity of the algorithm by controlling the number of output beams from the conventional beamforming and is applicable for both non coherent and coherent signals. The simulation and experiment results show that the proposed algorithm has azimuth resolution performance comparable to the traditional element-domain SBL beamforming algorithm, and is superior to the conventional beamforming and Minimum Variance Distortionless Response (MVDR) algorithms. When applied to short and dense arrays, the computation complexity of the proposed algorithm is significantly lower than that of the traditional element-domain SBL beamforming algorithm.

4:40

4pSP11. Striation-based beamforming using a horizontal array. Donghyeon Kim (Korea Maritime and Ocean Univ., 727 Taejong-ro, Yeongdo-Gu, Busan 49112, Korea (the Republic of), donghyeon.ual@gmail.com), Gihoon Byun, J. S. Kim, Daehwan Kim (Korea Maritime and Ocean Univ., Busan, Korea (the Republic of)), and Heechun Song (Scripps Inst. of Oceanogr., San Diego, CA)

Conventional beamforming (CBF) based on plane waves is widely used for direction finding with a horizontal array deployed in ocean waveguides. However, CBF suffers from loss of spatial coherence due to the inhomogeneous field produced by constructive and destructive interference between multipath arrivals. Moreover, the grazing angle propagation leads to a bias in the estimated bearing. In this work, we revisit the striation-based beamforming (SBF) that exploits the waveguide invariant representing the complex wave propagation [Zurk and Rouseff, J. Acoust. Soc. Am. **132**, EL264 (2012)]. Our approach involves a preliminary processing to restore the spatial coherence across the horizontal array, followed by CBF. Experimental results illustrate the performance improvement of SBF over CBF in terms of array gain, resolution, and bias.

Session 4pUWa

Underwater Acoustics and Signal Processing in Acoustics: Mobile Underwater Acoustic Sensor Networks: Communication, Localization, and Networking Challenges II

Aijun Song, Cochair
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Fumin Zhang, Cochair
Hong Kong University of Science and Technology, Rm. 114, University Center, HKUST, Clear Water Bay, Kowloon, Hong Kong

Chair's Introduction—12:55

Invited Papers

1:00

4pUWa1. Source localization in deep water with a towed array from UUV. Zhenglin Li (School of Ocean Eng. and Technol., Sun Yat-sen Univ., Rm. B410 Haiqing 3 Bldg., Zhuhai, Guangdong 519000, China, lizhlin29@mail.sysu.edu.cn), Chao Ming, Haiqiang Niu (State Key Lab. of Acoust., Inst. of Acoust., Chinese Acad. of Sci., Beijing, China), Linji Xu, and Peng Xiao (School of Ocean Eng. and Technol., Sun Yat-sen Univ., Zhuhai, Guangdong, China)

Unmanned Underwater Vehicle (UUV) can work in larger depth. If a towed array applied on UUV, it can detect the acoustics signals from a target in the direct arrival zone with lower transmission loss in deep water. Based on sound field characteristics in deep water, a source detection scheme with multiple UUVs is proposed. The aperture of sonar array is limited by the carrying capacity of UUV. Therefore, a passive aperture extension with sparse Bayesian learning (SBL) is proposed for a moving short array to improve the performance of direction-of-arrival estimation by constructing a larger virtual aperture. The phase correction factors and bearings of different targets are estimated simultaneously by SBL. The experiment results show that the method extends the aperture effectively and obtains higher azimuth resolution and accuracy for the direction-of-arrival (DOA) estimation than the Conventional Beamforming (CBF). The detection of an underwater target by using a towed array from UUV is possible. [Work supported by the National Natural Science Foundation of China, Grant No. U22A2012.]

1:20

4pUWa2. Waveguide invariant navigation of an autonomous underwater vehicle. Junsu Jang (Scripps Inst. of Oceanogr., UC San Diego, 8820 Shellback Way, La Jolla, CA 92037, jujang@ucsd.edu) and Florian Meyer (Scripps Inst. of Oceanogr., UC San Diego, La Jolla, CA)

Reliable navigation of an autonomous underwater vehicle (AUV) remains a challenge due to the unavailability of a global positioning system underwater. However, the self-localization uncertainty of an AUV with an acoustic receiver can potentially be reduced by leveraging passive acoustic ranging techniques. In particular, in shallow water, the range between an acoustic source and a passive receiver can be estimated using the waveguide invariant. In this work, we develop a sequential Bayes navigation filter that fuses passive recordings of an acoustic source with the measurements of an inertial sensor. The acoustic source can either be cooperative or a source of opportunity, e.g., a container ship. Using particle-based processing, our filter can either estimate the waveguide invariant or the AUV position by making use of a detailed nonlinear statistical model of the received signal. If the position of the AUV is accurately known, e.g., in the beginning of the AUV's mission, the waveguide invariant can be estimated. Alternatively, assuming that accurate waveguide invariant information is available, range information can be obtained and used to improve the estimate of the AUV's position. The navigation capability provided by the proposed sequential Bayes filter is demonstrated using simulated and real data.

1:40

4pUWa3. Pioneering time-of-arrival localization with cable-free and battery-free transponders. Lina Pu (Comput. Sci., Univ. of Alabama, 245 7th Ave. Tuscaloosa, AL 35401, lina.pu@ua.edu), Yu Luo (Elec. and Comput. Eng., MS State Univ., MS State, MS), and Aijun Song (Elec. and Comput. Eng., Univ. of Alabama, Tuscaloosa, AL)

In traditional underwater acoustic positioning methods, reliance on systems such as Long Baseline (LBL) and Short Baseline (SBL) localization utilize transponders that actively transmit acoustic pulses. These systems are typically powered by cables and often result in constrictive deployment. The adoption of cable-free and battery-free transponders can greatly enhance the flexibility of deployment. This change facilitates large-scale implementation, thereby providing precise positioning services across extensive areas. However, localization with passive transponders faces unique challenges. It has been widely accepted that battery-free nodes are unsuitable for

time-of-arrival (ToA) localization. The primary reason is the wakeup delay varies based on location and environment, which introduces an unpredictable offset into the time lapse between the transmitted and received pulses. To overcome this challenge, we model the wakeup delay of passive node, considering the hardware characteristics of the passive node and the multipath acoustic channel. Moreover, we propose a ToA-based localization framework specifically tailored for battery-free transponders. In this paper, we will verify our wakeup delay model and assess the performance of the proposed localization method through experimental results. This new approach marks a significant departure from the existing effort, making localization with passive transponders feasible.

2:00

4pUWa4. An open source acoustic modem for communication between autonomous underwater vehicles. Fumin Zhang (Hong Kong Univ. of Sci. and Technol., Hong Kong, Hong Kong, fzhang37@gatech.edu)

Miniature underwater robots (MUR) are becoming popular for underwater robotics research. Existing acoustic modems are typically designed for larger vehicles, which requires relatively high power and high cost. We present the Bluebuzz, a open source acoustic modem with reduced size and low power consumption aiming for the applications of MURs. We will report the detailed technical solutions and the experimental results collected and presents future improvements. The modems have also been tested to support higher level networking functions among multiple MUR.

Contributed Papers

2:20

4pUWa5. Multi-function underwater acoustic networks combining communication, navigation, and sonar. David A. Brown (ECE/CIE, Univ. of Massachusetts Dartmouth, 151 Martine St., Fall River, MA 02723, dbAcousticsdb@gmail.com)

Autonomous underwater vehicles often have on-board acoustic modems and navigation systems that work in connection with multiple vehicles. These capabilities may be combined or enhanced with using multifunction transducers that transmit communication and navigation information. The use of multimode transducers to produce acoustic spiral waves for communication, navigation and sonar are presented.

2:40

4pUWa6. Direct acoustic communication between underwater and airborne nodes. Shaojian Yang (Zhejiang Univ., 1 Zheda Rd., Zhoushan 316021, China, sjyang@zju.edu.cn), Yimu Yang (The Univ. of Edinburgh, Edinburgh, United Kingdom), Xingbin Tu (Zhejiang Univ., Zhoushan, China), Xuesong Lu (Hangzhou Dianzi Univ., Hangzhou, China), Wei Yan, and Fengzhong Qu (Zhejiang Univ., Zhoushan, China)

Direct acoustic communication between underwater and airborne nodes has always been considered unfeasible due to the energy loss caused by the strong surface reflection of sound waves. However, contrary to popular belief, our study demonstrates that underwater transducers can effectively transmit detectable acoustic signals in the air. To investigate this phenomenon, we conducted an experiment involving the deployment of an underwater transducer at a depth of 1 m, while an unoccupied aerial vehicle equipped with a voice recorder was positioned at various altitudes ranging from 2 to 30 m and horizontal distances of 0–30 m. Sound pressure levels were measured at 20 different positions within the frequency range of 10–20 kHz, and orthogonal frequency division multiplex acoustic communication signals were recorded at specific positions. Our findings reveal the successful establishment of a direct acoustic communication link between the water and air interface, achieving a data rate of 4.565 kbps. This study opens up new possibilities for practical applications in underwater-to-air communication systems.

3:00

4pUWa7. Machine learning transmission loss simulations in complex undersea environments with range-dependent bathymetry. Ryan A. McCarthy (Marine Physical Lab., Scripps Inst. of Oceanogr., Univ. of California San Diego, 9500 Gilman Dr #0214, La Jolla, CA 92093, r1mccarthy@ucsd.edu), Jit Sarkar, Sophia Merrifield, Ryan Bednar, Derek Ung, Andrew Nager, Charles Brooks, and Eric Terrill (Marine Physical Lab., Scripps Inst. of Oceanogr., Univ. of California San Diego, San Diego, CA)

Range-dependent bathymetric variability introduces diverse scattering and multipath that can impact optimal acoustic communication ranges, and will impact autonomous underwater vehicle (AUV) operations. We are working towards building a sound-aware framework in which the AUV has knowledge of the transmission environment and can exploit or react with intended outcomes. Prior knowledge of the environment coupled with acoustic propagation modeling on-board AUVs can improve sound-awareness of a vehicle but is computationally intensive, and costly from a power-budget perspective. This work addresses the computational burden by training a machine learning (ML) model to interpret range-dependent acoustic propagation through environmental inputs to predict transmission loss (TL) from the ray tracing model, BELLHOP. A decision tree model is trained to predict locations in range and depth of acceptable TL using feature representations to reduce bathymetric information within a region. Models are trained and tested with TL field realizations for acoustic communications at 10 and 25 kHz for environments with varying bathymetry at ranges up to 3 km through BELLHOP. We present simulations and results from field experiments performed off the coast of Southern California. The communication paths are between a surface vehicle (Liquid Robotics Waveglider, Herdon, VA) and an AUV (REMUS UUV, Huntington Ingalls, Falmouth, MA). Results for model predictions and collected field data will be discussed.

3:20–3:40 Break

3:40

4pUWa8. Theoretical investigation of sound propagation from a moving directional source in a shallow-water waveguide. Tengjiao He (Key Lab. of Marine Intelligent Equipment and System, Shanghai Jiao Tong Univ., Shanghai Jiao Tong University, No 800,Dongchuan Minxing District, Shanghai City, Shanghai 201100, China, hetengjiao@sjtu.edu.cn), Bin Wang, and Ruixin Nie (Key Lab. of Marine Intelligent Equipment and System, Shanghai Jiao Tong Univ., Shanghai, China)

Doppler phenomena resulting from a moving directional source can be complicated in shallow water environments. This study presents a semi-analytical method to calculate the Doppler-shifted field (DSF) caused by a directional source moving horizontally in a shallow-water waveguide. First, an improved normal-mode (NM) model is developed to comprehensively account for the Doppler-induced changing modal shapes, eigenvalue shifts, and the contribution of the branch cut integral. Next, an analytical, modal expression for the DSF is derived based on the principles of Huygens and Taylor series expansion. The solution is expressed as a sum over propagating eigenmodes, with the modal excitation determined by the source directivity depending upon the grazing angle of each pair of modal plane waves. This modal expression is applicable to any type of radiators moving in a shallow-water environment with an arbitrary sound speed profile. By employing the proposed theoretical model, we analyze the beam Doppler shift produced by a piston-like radiator moving in a two-layer waveguide. The simulation indicates that an apparent asymmetry arises in the DSF when the beam grazing angle sweeps over the critical angle of the seafloor. This finding confirms that the beam Doppler shift acts as the third mechanism causing significant differences between the field generated by a directional moving source and that by the same source at rest, which may have practical applications in underwater acoustic detection for moving sources.

4:00

4pUWa9. Feature extraction and classification of deep-sea mobile underwater acoustic channels. Chenyu Pan (College of Underwater Acoust. Eng., Harbin Eng. Univ., Harbin, Heilongjiang 150001, China, panchenyu@hrbeu.edu.cn), Songzuo Liu, Xin Qing, and Gang Qiao (College of Underwater Acoust. Eng., Harbin Eng. Univ., Harbin, China)

The reliable acoustic path (RAP) is one of the crucial channels for deep-sea sound propagation, which is affected weakly by the interface and has lower transmission loss, enabling long-distance communication. However, RAP-based deep-sea acoustic communication may face channel model mismatch issues. In order to analyze the dynamic characteristics of spatial-temporal variability channels, deep-sea mobile underwater acoustic channel measurement experiments were conducted. This work proposes a deep learning method based on multi-dimensional properties to classify deep-sea channels. Specifically, the sound ray convergence zone leads to a complex multipath structure and severe delay spread in the RAP channel. The fuzzy c-means (FCM) algorithm is used for multipath clustering to extract accurate channel features, and then the Markov chain (MC) is introduced to track the evolution characteristics of multipath clusters. Finally, the coupling features of channel time-variant impulse response (TVIR) and multi-dimensional statistical properties are used as the input of convolutional neural networks (CNNs) to obtain the quantitative evaluation index as the channel classification to build a channel feature dataset for underwater mobile platforms. This dataset can effectively assist in identifying deep-sea mobile channels and promote the development of adaptive underwater acoustic communication systems on mobile platforms.

4:20

4pUWa10. Uncrewed undersea vehicle embedded autonomous target recognition in synthetic aperture sonar imagery. Bryan D. Todd (Littoral Acoust. and Target Phys., Naval Surface Warfare Ctr. Panama City Div., 341 Emerald Cove St., Panama City Beach, FL 32407, Bryan.D.Todd3.civ@us.navy.mil), Ivan I. Rodriguez-Pinto (Littoral Acoust. & Target Phys., Naval Surface Warfare Ctr. Panama City Div., Panama City, FL), and Daniel D. Sternlicht (Naval Surface Warfare Ctr. Panama City Div., Panama City, FL)

The opacity of the ocean to electromagnetic signals makes communication with Uncrewed Undersea Vehicles (UUVs) extremely difficult, especially for situations where the vehicle is operating far from its center of command and control, and makes mid-operation data exfiltration for external processing untenable. This leaves operator intervention and control limited necessitating the need for advanced and adaptive autonomy that can create situational awareness from mission data provided by vehicle sensors. Long range high-resolution imagery generated by synthetic aperture sonar (SAS) is a high fidelity data product that can be further processed by both machine learning techniques and traditional image processing techniques to create computer aided detection and classification (CAD/CAC) capabilities to feed that developing autonomy need. This work examines two separate detection and classification pipelines that have been used in parallel to create contact sets that are actionable for onboard decision processes.

4:40

4pUWa11. Mobile AUV localization using multiple static transceivers equipped with linear arrays. Yonghua Chen (School of Electron. and Information Eng., South China Univ. of Technol., No. 381, Wushan Rd., Tianhe District, Guangzhou 510640, Guangdong Province, China, 15079124913@163.com), Hua Yu, Jie Li, Fei Ji, and Fangjiong Chen (School of Electron. and Information Eng., South China Univ. of Technol., Guangzhou, China)

Most existing research on multi-base sonar AUV positioning systems typically assumes that the AUVs are stationary and only observe the Time of Arrival (TOA) for positioning. If the AUV is assumed to be moving at medium to high speed and its motion during measurement is not considered, this will decrease positioning accuracy. Meanwhile, combining multiple positioning parameters, such as Angle of Arrival (AOA) and TOA, can improve target positioning accuracy. However, using planar arrays to observe two-dimensional AOA (2-D AOA) will consume more computational resources. To alleviate these issues, this study uses multiple static transceivers equipped with linear arrays to observe one-dimensional AOA (1-D AOA) and TOA to achieve moving AUV positioning. Firstly, the 1-D AOA and TOA parameter models under motion effects are pseudo-linearized, transforming the positioning problem into a constrained least squares problem. Then, a semi-definite programming (SDP) solution algorithm with noise tolerance is proposed. In addition, the Cramér–Rao Lower Bound (CRLB) under different observation models with motion effects is derived, and bias analysis under different observation models is conducted. Simulations validate the effectiveness of the proposed positioning method.

5:00

4pUWa12. Cooperative positioning method of underwater autonomous vehicle formation based on extended Kalman filter. Jingxu Zhao (College of Underwater Acoust. Eng., Harbin Eng. Univ., 145 Nantong St., Nangang District, Harbin City, Heilongjiang Province,China, Harbin 150001, China, zhaojingxu0521@163.com), Feng Zhou (College of Underwater Acoust. Eng., Harbin Eng. Univ., Harbin, China), and Chen Zhao (Beijing Inst. of Remote Sensing Equipment, Beijing 100854, China)

Autonomous Underwater Vehicles (UAVs) have shown significant potential application value in marine environmental surveillance, development, and utilization of resources. However, due to the complex underwater acoustic channel environment and the relative motion between AUVs, this may lead to heavy-tailed non-Gaussian process noise and measurement noise, leading to increased error and the Extended Kalman Filter (EKF) may fail. In prior studies, we propose a multi-AUV formation hierarchical target location algorithm based on the EKF, which can realize multi-AUV self-localization and stationary target localization. The AUV formation consists of one high-precision piloting AUV and several low-precision following AUVs. The following AUVs are divided into two levels, the reference AUVs and the AUVs to be tested. According to the designed positioning period, the reference AUV receives the position parameters from a high-precision piloting AUV and transmits its own position parameters to the AUV to be measured. Then use the EKF to complete the cooperative position correction of the AUV cluster. In this paper, we will study the influence of heavy-tailed noise on this method and study the EKF based on Huber estimation to improve the antijamming capability.

Session 4pUWb

Underwater Acoustics: Underwater Noisescapes

Daniel Castro, Chair
Volta Acoustics, L11 470 Collins Street, Melbourne 300, Australia

Contributed Papers

3:00

4pUWb1. Soundscapes from deep-water moored receivers in the vicinity of the New England Seamounts. Matthew Walters (Dept. of Phys., Naval Postgrad. School, 833 Dyer Rd., Bldg. 232, Monterey, CA 93943, matthew.walters@nps.edu), Oleg A. Godin (Dept. of Phys., Naval Postgrad. School, Monterey, CA), John Joseph (Dept. of Oceanogr., Naval Postgrad. School, Monterey, CA), and Tsu Wei Tan (Dept. of Marine Sci., ROC Naval Acad., Kaohsiung, Taiwan)

Acoustic noise interferometry in the ocean relies on synchronized measurements of ambient sound at spatially separated points to passively measure the acoustic travel times between receiver locations. A network of four autonomous, moored, near-bottom acoustic receivers were deployed in the vicinity of the New England Seamounts for 52 days as part of the larger New England Seamount Acoustics (NESMA) Pilot experiment. Receiver depths ranged from 2500–4475 m. The moored receivers provided stable observation platforms in a region with highly variable and occasionally strong currents. The noise interferometry network aimed to investigate the feasibility of utilizing passive acoustic remote sensing methods to study a highly dynamic ocean region with complex bathymetry. Strong ocean variability at the experimental site was induced by its proximity to the Gulf Stream. This paper presents the initial analysis of water-depth dependence and intermittency of ambient sound spectra and spectrograms on near-bottom receivers. The relation between the spectral features of ambient sound and ocean dynamics is explored. Additionally, the feasibility of using the onboard Chip Scale Atomic Clocks to synchronize the experimental data and calculate broadband noise cross-correlation functions for each of the receiver pairs is discussed.

3:20

4pUWb2. Floors of Heaven: A case study of an underwater acoustic soundscape for recreational purposes. Daniel Castro (Volta Acoust., L11 470 Collins St., Melbourne, Victoria 300, Australia, daniel.castro@voltaacoustics.com.au) and Sophie Gleeson (Acoust., Arup, Docklands, Victoria, Australia)

This paper presents a case study of an underwater music piece titled “Floors of Heaven” by British artist Leon Vynehall. This installation was delivered in a swimming pool as part of the Mona Foma 2023 festival, representing an example of the use of soundscapes as a means of artistic expression. The study delves into contemporary trends in soundscapes, with a focus on their utilization as a recreational tool, as the writers observe the growing interest in creating immersive and experiential art installations that engage the viewer on a sensory and emotional level. Subsequently, the paper discusses how soundscapes can be delivered in underwater environments, which present unique challenges in terms of acoustics, sound propagation, and environmental risk factors such as hearing damage. The technical considerations involved in creating an underwater sound installation, including the use of hydrophones and underwater speakers, and the importance of selecting appropriate hardware to achieve acceptable noise levels underwater, are thoroughly described. The case study of “Floors of Heaven,” which uses a combination of field recordings and synthesized sounds to explore themes of ecological collapse and the impact of human activity on underwater environments, is then presented.

3:40

4pUWb3. Numerical investigation of shipping noise in the Red Sea. Rihab Larayedh (Comput., Elec. and Mathematical Sci. and Eng. Div., King Abdullah Univ. of Sci. and Technol., Kaust, Towal, Jeddah 23955-6900, Saudi Arabia, rihab.larayedh@kaust.edu.sa), George Krokos (Physical Sci. and Eng. Div., King Abdullah Univ. of Sci. and Technol., Anavyssos, Greece), Bruce Cornuelle (Scripps Inst. of Oceanogr., San Diego, CA), and Ibrahim Hoteit (Physical Sci. and Eng. Div., King Abdullah Univ. of Sci. and Technol., Jeddah, Saudi Arabia)

Underwater noise pollution is a significant environmental issue that can have detrimental effects on marine ecosystems. One of the main sources of underwater noise pollution is ship traffic, which has been shown to negatively impact marine animals by masking communication signals and altering their behaviors. This study represents the first comprehensive analysis of underwater ship noise in the Red Sea. It aims to generate noise maps of ships sailing through the main shipping lane in the Red Sea. The Range-dependent Acoustic Model (RAM), incorporating anthropogenic and environmental inputs, was utilized to predict maps of underwater ship noises. The application of RAM yielded maps showcasing the spatial and temporal distribution of underwater ship noise in the Red Sea, providing valuable insights for policy makers and facilitating targeted mitigation efforts, with implications for future research on the impacts of underwater noise pollution on marine life.

4:00

4pUWb4. Estimation of ambient marine noise levels in New South Wales. Marco Velasco (GHD, 64 Thomson St., Darlinghurst, New South Wales 2011, Australia, marco.velasco@ghd.com)

Marine and offshore infrastructure is increasingly becoming more prominent in NSW resulting in a requirement to investigate the potential behavioral and physiological impacts to marine fauna during construction and operational phases. Typically, detailed modeling is undertaken to compare the predicted underwater noise levels against noise thresholds with limited consideration to the existing ambient noise levels specific to the site. Vessel movements are known to be the primary contributor to ambient underwater noise levels from anthropogenic sources whereas prevailing noise is generally dominated by natural sources such as wind, waves, precipitation, and seismic activity. Underwater noise monitoring of baseline conditions is not always feasible and can also be cost-prohibitive to undertake. Additionally, there is currently limited publicly available underwater noise measurement data along the eastern coastline of NSW, specifically in shallow waters at depths less than 50 m. This study includes simple procedures to allow for preliminary estimates of the key contributors to the marine noise environment in the context of the Eastern coastline of NSW, with the aim of producing maps estimating existing marine noise levels for common descriptors such as SPL(rms) and SEL(24hour).

Session 5aAA

Architectural Acoustics: General Topics in Architectural Acoustics

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Contributed Papers

8:00

5aAA1. A report on an anechoic room design using metal wedges. Vicky H. Maulana (PT. Akustika Swara Indonesia, Ruko Amber 2, Blok B. No.3, Tangerang, Banten 15132, Indonesia, vicky@akustika.co.id), Denny Hermawanto (PT. Akustika Swara Indonesia, Tangerang, Indonesia), Benjamin Sunarko (PT. Akustika Swara Indonesia, Bandung, Indonesia), and Dodi Rusjadi (PT. Akustika Swara Indonesia, Tangerang, Indonesia)

An anechoic room is important in acoustic measurements to simulate free-field conditions. The room is designed with walls that are covered with wedges that absorb as much sound as possible to eliminate reflections. Conventional wedges are constructed from a high-density sound-absorbing material such as rock wool and attached to a wire frame to maintain the shape of the wedges. However, these wedges are fragile and easily deformed by physical disturbances. In this paper, a design of an anechoic room using metal wedges is presented. Glass wool with a density of 32 kg/m^3 is enclosed within a perforated metal casing, which provides durability and resistance to physical disturbances. The wedges are installed in a room with a dimension of 6.17 m $\times$ 5.87 m $\times$ 4.85 m. It is shown in a performance assessment based on ISO 3745 that wedges are capable of absorbing sound well, thus allowing sound to propagate according to the inverse square law. Additionally, a comparison study of the absorption performance of glass wool wedges compared to metal wedges is discussed with a view to improving the system in the future.

8:20

5aAA2. Exploring sound absorption properties of porous wood-based eco-friendly materials for noise reduction in buildings. Nishant Kumar (Dept. of Electronics and Commun. Eng., Koneru Lakshmaiah Education Foundation (KLEF) (Deemed to be University), Green Fields, Vaddeswaram, Guntur, Andhra Pradesh 522302, India, kumarnishant.kumar9@gmail.com), Kirti Soni (Water Resources Management and Rural Technologies, CSIR – Adv. Mater. and Process Res. Inst. (AMPRI), Bhopal, Madhya Pradesh, India), and Mahavir Singh (Acoust. and Vib. Metrology, Ex. CSIR- National Physical Lab., Delhi, India)

In the last few years, there has been an increase in the use of eco-friendly materials in the area of acoustics made from natural fibers. These materials are preferable to synthetic materials because of their benefits, such as raw material availability, cost-effective processing costs, safe handling, and equivalent acoustic qualities. The main objective of this study is to find out how well the material made from vetiver grass absorbs sound. Following the International Standard ISO 354-2003 and ASTM 423-90, sound absorption tests were done in a reverberation chamber to find the Sound Absorption Coefficient (SAC) and Noise Reduction Coefficient (NRC) over a wide range of frequencies from 100–4000 Hz. To enhance the way the material absorbs noise, several ASTM E795-16 mountings were examined and tested. These mountings, which have air cavity sizes ranging from 0 mm (hard base) to 50, 100, and 300 mm, include J, E600, C50, D10, and C10. The most effective technique to enhance the material's acoustic qualities to lower noise was identified by comparing the SAC and NRC readings of samples with various-sized air gaps. The fitting of C10 with a SynthPF 10 $\times$ 50 back and a 50 mm air gap produced the best results (NRC 0.82). The produced porous pine boards with a density of 400 kg/m^3 and a width of 20 mm met all the physical and mechanical property standards of the Indian Standards. The study shows that noise-absorbing materials made from vetiver grass could be used in buildings in ways that are better for the environment.

8:40

5aAA3. Practical applications at field environments of sound absorption measurement method using the ensemble averaging technique. Toru Otsuru (Sci. and Technol., Oita Univ., 700 Dannoharu, Oita 870-1192, Japan, otsuru@oita-u.ac.jp), Reiji Tomiku, and Noriko Okamoto (Sci. and Technol., Oita Univ., Oita, Japan)

To construct appropriate boundary conditions for wave-based room acoustics simulations, the authors have proposed a measurement method for sound absorption characteristics, surface impedance, and sound absorption coefficient of materials using the ensemble averaging technique. We have also shown the robustness as well as the reproducibility of the results obtained by the method at various environments. The method consists of two kinds according to sensor types employed so far: i.e., two-microphone and PU-sensor. Generally speaking, the method using a PU-sensor is more advantageous than one using a two-microphone because of its geometrical straightforwardness. However, at various field environments where environmental noise and/or wind blowing exist, the method using a two-microphone is expected to give more favorable results. In this paper, several applicational measurements of the method conducted at field environments using the two sensor types are summarized to show the plausibility of the method.

9:00

5aAA4. Sound absorption coefficient measurement for various building elements. Yang Ki Oh (Dept. of Architecture, Mokpo National Univ., 1666 Yeongsanro, Muan, Joennam 58554, Korea (the Republic of), oh.duoh@gmail.com) and Min Woo Kang (Dept. of Architecture, Mokpo National Univ., Muan, Korea (the Republic of))

The current sound absorption coefficient testing measurement method, ISO 354, primarily focuses on materials used for surface coverings or equivalent sound absorption objects. However, there are other building elements

such as lightweight partition walls and windows that do not have solid structures behind them. These elements may allow sound to transmit to adjacent interior spaces or escape to the outside. On the other hand, the transmitted sound is expected to be rebounded into the measurement room when the test material is placed on a hard and thick surface as described in ISO 354. The test was conducted in a reverberation chamber located at the center of three consecutive acoustic testing chambers: reverberation-reverberation-anechoic. The testing chambers are equipped with test openings between them and are fully isolated from one another. The purpose of the test was to confirm whether the current measurement method reflects the influence of the assumed rebounded sound. The sound absorption coefficients were measured with the specimen installed on the floor, representing the ISO 354 condition, and comparisons were made to the conditions where the specimen was installed in another reverberation chamber opening and in the anechoic chamber opening.

9:20

5aAA5. The impacts of sustainable design practices and cross laminated timber structures on acoustic design. Nick Wedd (Arup, Perth, L14 Exchange Tower, 2 The Esplenade, Perth, Western Australia 6000, Australia, Nick.Wedd@Arup.com), Daniel Jimenez, and Laura Cowie (Arup, Melbourne, Docklands, Victoria, Australia)

More projects in Australia are prioritizing sustainable design practices throughout their building life cycle. Environmental Sustainable Design (ESD) rating tools such as WELL, Green Star, and the Living Building Challenge (LBC) have existed for a decade or more, with some receiving wide adoption in Australia. These rating tools require the assessment of the ingredients, volatile organic compounds (VOCs), and embodied carbon of the materials specified on the project. In addition, cross-laminated timber (CLT) buildings are becoming a more attractive development option due to lower embodied carbon, a reduction in the structural requirements of the foundations, and an increased speed of construction. These sustainable practices are reviewed using a recent project example that combines these recent developments in sustainability: a commercial office fitout in a hybrid steel and cross-laminated timber structure that is targeting WELL Platinum and the Living Building Challenge. The challenges of specifying materials to meet sustainability targets, potential gaps in available products, and the acoustic design challenges of achieving high levels of sound insulation in cross-laminated timber structures are discussed. With a better understanding of sustainable design practices, acoustic designers can improve sustainability outcomes on all projects.

9:40

5aAA6. A method for evaluating the spatial uniformity of sound pressure in a small room by using power spectra at fewer measurement points. Ryoichi Suzuki (Nihon Univ., 7-24-1, Narashinodai, Funabashi, Chiba 274-8501, Japan, suzuki.ryoichi@nihon-u.ac.jp), Kazuma Hoshi, and Toshiki Hanyu (Nihon Univ., Funabashi, Chiba, Japan)

Sound pressure levels in low frequencies vary greatly depending on a measurement point, especially in a small room. The standard deviation of the spatial distribution of sound pressure level (σ_{SPL}) is used as one of the parameters to evaluate the spatial uniformity of sound pressure. However, many measurement points are required for an accurate evaluation using σ_{SPL}. In this study, we investigate a method for evaluating the spatial uniformity with fewer points. We had proposed the frequency-domain coefficient of variation (*FCV*) to evaluate the diffuseness of a sound field focusing on the frequency domain. The *FCV* is defined as the coefficient of variation of a power spectrum. The lower density of room modes in low frequencies would give large σ_{SPL}. Thus, larger *FCV* should be obtained when larger σ_{SPL} due to room modes occur. Generally, the power spectrum is obtained from a room impulse response (RIR). However, the bandwidth of each mode is changed depending on the decay of the RIR, namely, a reverberation time. To obtain the *FCV* avoiding the change of the bandwidth, the decay of RIR is canceled. The results of numerical simulation showed that the *FCV* at fewer points can evaluate the spatial uniformity than the σ_{SPL}.

10:00–10:20 Break

10:20

5aAA7. The comprehensive role of acoustic materials in healthy sustainable building design. Robert Jones (285 Swan St., Melbourne, Victoria 3121, Australia, robj@autex.com.au)

In today's world, the acoustic performance of buildings is increasingly evaluated in tandem with its environmental, health, and well-being implications. Design choices related to noise management, speech privacy, speech security, and reverberation time not only influence the occupant's comfort but also impact the environment and the people involved in the construction process. As such, acoustic materials are pivotal in creating sustainable and healthy buildings. Beyond traditional acoustic considerations, broader aspects impacting occupants' experience, such as sensory perception and indoor air quality, must be evaluated. A significant emphasis is placed on biophilic design to foster a connection between the built environment and nature for enhanced well-being. Similarly, the environmental impacts of material choices and the transition to recycled products to reduce the industry's environmental footprint are essential considerations. Acoustic materials have a far-reaching impact on sustainable and healthy building design by addressing occupant comfort, experience, air quality, and the environmental implications of material choices. Moreover, a comprehensive approach to acoustic design and material selection can pave the way for a greener, healthier, and more sustainable future.

10:40

5aAA8. Revision of Sabine's reverberation theory by following a different approach of Eyring's theory. Toshiki Hanyu (Dept. of Living and Architectural Design, Nihon Univ., 7-24-1 Narashinodai, Funabashi, Chiba 274-8501, Japan, hanyu.toshiki@nihon-u.ac.jp)

The room acoustic theory was established based on Sabine's reverberation theory. However, in Sabine's theory, the reverberation time does not become zero even if the absolute absorption condition is satisfied. This is considered to be a contradiction of Sabine's theory, and Eyring revised the reverberation theory to resolve this contradiction. In this study, the theoretical framework for a consistent reverberation theory is first presented. By using this framework, it is shown that Eyring's theory has a contradiction between the sound energy density in the steady state and the energy decay from the steady state, which is absent in Sabine's theory. Based on the proposed theoretical framework, Sabine's reverberation theory is revised by following a different approach of Eyring's theory. The reverberation time obtained using the revised theory is shorter than that obtained using Sabine's theory and longer than that obtained using Eyring's theory. Finally, computer simulations using the ray-tracing method are performed to verify the revised theory. The average sound pressure levels and reverberation times obtained using the computer simulations are in better agreement with the values calculated using the revised theory compared to those calculated using Sabine's and Eyring's theories.

11:00

5aAA9. An assessment of the effectiveness of acoustic treatments for balconies and open spaces as required by Queensland State Assessment and Referral Agency. Burak Ayva (Acoust. and Vib. Div., Trinity Consultants Australia, Trinity Consultants Peel St., Level 3/43, South Brisbane, Queensland 4101, Australia, burak.ayva@trinityconsultants.com), Witold Mazur, Samuel Wong, and Beau Weyers (Acoust. and Vib. Div., Trinity Consultants Australia, Brisbane, Queensland, Australia)

This study examines the effectiveness of acoustic treatments required to fulfill "Queensland State Assessment and Referral Agency (SARA)" conditions for balconies or private open spaces in apartment units above the ground floor. SARA SDAP State Code 1 Performance and Acceptable Outcome PO42 mandates developers to use a combination of continuous, solid, gap-free structures or balustrades and highly acoustically absorbent material treatments for such locations. The study explores the relevance of SARA conditions to balconies and highlights the absence of specific statistical noise level criteria for upper-floor private open spaces affected by traffic noise from state-controlled roads in Queensland. Furthermore, the Code requires these measures for balconies "adjacent" to state-controlled roads, which DTMR defined to be within 100 m of a state transport corridor and/or infrastructure. To investigate these aspects, computational noise modeling

was undertaken to predict and assess the impact of road traffic noise on balconies, considering factors such as diffraction, reflection, and direct noise propagation paths. The study also analyzes the potential benefits of acoustic treatments, which were found to be dependent on sight angles to and distances from state-controlled road. In certain cases, minimal acoustic benefits are observed. It is recommended that there is a shift from prescriptive acoustic requirements to performance-based and building-specific requirements within the SARA framework.

11:20

5aAA10. Soundscape of Sandiaoling Ecological Tunnel. Shiang-I Juan (Architecture, National Taiwan Univ. of Sci. and Technol., 43 Keeling Rd. Sec. 4, Taipei, Taiwan, S.Juan@mail.ntust.edu.tw), ChihHsuan Chen, Quynh X. Dinh, Kevin Harsono, Evan Hezekiah, Mahad Moussa Houssein, Fatma Kayatekin, Stirena Rossy Tamariska, Felicia Wagiri, Hao-Chun Yang, and Lucky Tsaih (Architecture, National Taiwan Univ. of Sci. and Technol., Taipei, Taiwan)

The Sandiaoling Ecological Tunnel, originally an abandoned railway, has been transformed into a 1852 m long ecological bike path. Constructed with stones and bricks, the tunnel now features an elevated bike path supported by threaded iron rods, with a gentle water flow beneath. As the tunnel has two main sections and an open transition area in between it, the entrance (open), the middle of the tunnel (enclosure), near the tunnel exit (semi-open), and the transition area between the two main sections are selected as the four soundscape study locations. The study follows ISO-12913 soundscape study method to document essential sound sources, paths, and levels. Odeon Room Acoustic Software is used to study the simulated sound source propagations. The research documents a taxonomy of sound sources, encompassing both natural and man-made sounds. The tunnel's atmosphere is characterized by quietude, monotony, coldness, and resonance. The average equivalent sound pressure level in the four measured locations is 59 dBA. Frequency spectrums and simulated reflection patterns are also examined. During the soundwalk, the overall listening impression of the tunnel is described as delightful, relaxing, peaceful, and harmonious with nature. The atmosphere inside the tunnel evokes a sense of serenity and tranquility.

11:40

5aAA11. Effect of background noise on speech privacy. Lucy Torr (Melbourne, Victoria, Australia) and Vaishnav Balaji (Resonate Consultants, Level 4, 440 Elizabeth St., Melbourne, Victoria 3000, Australia, vaishnav.balaji@resonate-consultants.com)

There is a need for speech privacy in many building environments, which is often affected by various factors such as human perception, acoustic separation of partitions, and background noise levels. Previous studies have classified speech privacy rating between spaces based on partition separation and internal noise in receiving spaces. However, the typical design process often overlooks a holistic approach to speech privacy. Green buildings, focused on sustainability, have recently gained global attention. They aim to reduce environmental impact while ensuring comfort and health for occupants through energy-efficient HVAC systems, sustainable insulation, and low emissivity glazing. Despite their benefits, energy-efficient HVAC systems can emit minimal noise levels. The low internal noise levels in green buildings can often adversely affect the speech privacy rating between spaces. The Green Building Council of Australia have recently addressed the requirement of speech privacy by introducing the need for minimum internal noise levels in the Green Star Buildings, in addition to the acoustic separation requirements between spaces. This paper aims to investigate the need for minimum internal noise levels to address the requirement of speech privacy holistically rather than addressing acoustic separation and internal noise levels as individual components of acoustic design.

Session 5aAB

Animal Bioacoustics, Acoustical Oceanography, Underwater Acoustics, and Signal Processing in Acoustics: Fisheries and Marine Park Management I

Xavier Mouy, Cochair
Northeast Fisheries Science Center, National Oceanic and Atmospheric Administration, National Marine Fisheries Service, 166 Water Street, Woods Hole, MA 02543

Miles Parsons, Cochair
AIMS, 71a Aurelian St, Palmyra, Perth 6157, Australia

Chair's Introduction—7:55

Contributed Paper

8:00

5aAB1. Developing a Global Library of Underwater Biological Sounds and a World Oceans passive acoustic monitoring day. Miles Parsons (Australian Inst. of Marine Sci., 71a Aurelian St., Palmyra, Perth, Western Australia 6157, Australia, m.parsons@aims.gov.au), Lucia Di Iorio (Univ. of Perpignan, Perpignan, France), Audrey Looby (Fisheries and Aquatic Sci., Univ. of Florida, Cedar Key, FL), T A. Mooney (Woods Hole Oceanogr. Inst., Woods Hole, MA), Steve Simpson (Univ. of Bristol, Bristol, United Kingdom), and Sierra Jarriel (Woods Hole Oceanogr. Inst., Woods Hole, MA)

The volumes of "Ocean Sound" recordings now being collected in aquatic systems around the world have reached a level that has necessitated a new wave of data sharing, processing, and analysis techniques to help detect, identify, and assess an increasing number of known and unknown sound sources and sound types. These new tools are matched with a need for reference material to assist the rapidly growing bioacoustics research community, and a process to standardize how they are characterized and reported. A working group of the International Quiet Ocean Experiment proposed the development of a Global Library of Underwater Biological Sound (GLUBS) to integrate new and existing applications to help address these needs. Two initial objectives of GLUBS have been to increase awareness of unknown sounds and to develop a community-wide project that can broaden our understanding of aquatic soundscapes around the world. To that end, we discuss recent developments in designing GLUBS, incorporating underwater sonifery as a searchable trait on the World Register of Marine Species, initiating a recent global community effort to simultaneously record aquatic soundscapes on World Oceans Day, and a drive to agree on a standard process for characterizing newly described sounds.

Invited Papers

8:20

5aAB2. FishSounds: A data-sharing website of global soniferous fish diversity. Audrey Looby (Fisheries and Aquatic Sci., Univ. of Florida, 552 1st St., Cedar Key, FL 32625, alooby101@gmail.com), Kieran Cox (Dept. of Biological Sci., Simon Fraser Univ., Vancouver, BC, Canada), Amalis Riera (Dept. of Biology, Univ. of Victoria, Victoria, BC, Canada), Sarah Vela (MERIDIAN, Halifax, NS, Canada), Santiago Bravo (Instituto Oceanográfico, Universidade de São Paulo, São Paulo, Brazil), Rodney Rountree (Dept. of Biology, Univ. of Victoria, East Falmouth, MA), Francis Juanes, Hailey L. Davies, Brittnie Spriel (Dept. of Biology, Univ. of Victoria, Victoria, BC, Canada), Laura K. Reynolds (Soil, Water, and Ecosystem Sci., Univ. of Florida, Gainesville, FL), and Charles W. Martin (Nature Coast Biological Station, Univ. of Florida, Gainesville, FL)

Active fish sound production is geographically and taxonomically widespread—though not homogenous—among fishes, including numerous commercially and recreationally important fisheries species. Despite the ecological importance of fish sounds, their passive acoustic monitoring (PAM) applications, and extensive endeavors to document them, the field of fish bioacoustics has been historically constrained by the lack of an easily accessible, comprehensive inventory of fish sound production. To create such an inventory while simultaneously assessing the global extent of known soniferous fish species, we extracted information from almost 1000 references from the years 1874–2021 to determine that over 900 fish species have been shown to produce active (i.e., intentional) sounds. Our information is collated on the FishSounds website at FishSounds.net along with representative recordings of fish sounds that can be easily searched through and accessed by our users. FishSounds has since launched a new initiative to develop an acoustic catalog for Canadian-specific fisheries species and explore their ecological characteristics, spatial distribution, and taxonomy. The data available on FishSounds can be similarly adapted to meet other regional management needs, facilitate the application of PAM, and aid in the discovery of novel soniferous behaviors across fishes globally.

8:40

5aAB3. Abstract withdrawn.

9:00

5aAB4. Fixed-station and glider-based passive acoustic monitoring reveals spatiotemporal spawning dynamics of Atlantic cod and their potential interaction with offshore wind energy. Rebecca Van Hoeck (National Oceanic and Atmospheric Administration, 1401 Constitution Ave., Washington, DC 20009, rebecca.vanhoeck@noaa.gov), Timothy J. Rowell (Northeast Fisheries Sci. Ctr., National Oceanic and Atmospheric Administration, Woods Hole, MA), Micah J. Dean (Annisquam River Marine Fisheries Field Station, Massachusetts Div. of Marine Fisheries, Gloucester, MA), Aaron N. Rice (K. Lisa Yang Ctr. for Conservation Bioacoustics, Cornell Lab of Ornithology, Cornell Univ., Ithaca, NY), and Sofie M. Van Parijs (Northeast Fisheries Sci. Ctr., National Oceanic and Atmospheric Administration, Woods Hole, MA)

Atlantic cod, which produce sounds associated with courtship behaviors and are overfished in the US, are potentially vulnerable to disturbance from offshore wind energy (OWE) construction and operation during their spawning period. We used a combination of fixed-station and glider-based passive acoustic monitoring methods to evaluate the spatiotemporal spawning dynamics of Atlantic cod and assess their potential interaction with OWE at the extreme southern extent of the species' range in Southern New England waters of the Western North Atlantic Ocean. Generalized linear modeling of call presence and rates suggests that spawning was concentrated in November and December and was greatest near the new and full moons. The results from both fixed-station and glider-based methods suggest that spawning overlaps with planned OWE construction in time and space. Comparison of these temporal spawning dynamics in Southern New England to analogous PAM data from the geographically separated Massachusetts Bay winter-spawning Atlantic cod subpopulation revealed that the seasonality of inferred spawning was the same between the two regions. By leveraging multiple methods and datasets, this work demonstrates the effectiveness of PAM for understanding spawning phenology of and potential disturbance to an economically important fishery.

Contributed Paper

9:20

5aAB5. Monitoring haddock in the Stellwagen Bank National Marine Sanctuary using passive acoustics. Xavier Mouy (Northeast Fisheries Sci. Ctr., National Oceanic and Atmospheric Administration, National Marine Fisheries Service, 166 Water St., Woods Hole, MA 02543, xavier.mouy@outlook.com), Rodney Rountree (Biology Dept., Univ. of Victoria, East Falmouth, MA), Matthew Brander (Biology Dept., Univ. of Rhode Island, Kingston, RI), Anne Smith (Biology Dept., College of the Holy Cross, Worcester, MA), and Sofie M. Van Parijs (Northeast Fisheries Sci. Ctr., National Oceanic and Atmospheric Administration, National Marine Fisheries Service, Woods Hole, MA)

Haddock (*Melanogrammus aeglefinus*) are important commercial resources in the western North Atlantic. Their distribution ranges from Newfoundland to North Carolina and are most abundant on Georges Bank and in the Gulf of Maine. Over the course of a decade, the Gulf of Maine warmed faster than 99% of the global ocean and these changes are likely to impact the distribution and reproduction of haddock. It is therefore increasingly important to closely monitor this species and ensure a sustainable fishery. The objective of this study is to monitor the spatial and temporal occurrence of haddock using the sound they produce. We manually annotated over 20 000 haddock calls at five different sites and four different years and used them to train a convolutional neural network that can automatically detect haddock calls in acoustic recordings. The detector was then used to analyze several months of passive acoustic data collected in the Stellwagen Bank National Marine Sanctuary. The analysis revealed that haddock calls were mostly detected during the spawning season (January to March) but were also detected as late as August. We discuss how passive acoustics can complement existing monitoring methods and support the management of this commercially and ecologically important species.

Invited Papers

9:40

5aAB6. The search to identify the fish species chorusing along the Southern Australian Continental Shelf. Lauren A. Hawkins (Ctr. for Marine Sci. and Technol., Curtin Univ., Kent St., Bentley, Western Australia 6102, Australia, laurenhawkins799@gmail.com), Benjamin J. Saunders (School of Molecular and Life Sci., Curtin Univ., Perth, Western Australia, Australia), Christine Erbe (Ctr. for Marine Sci. and Technol., Curtin Univ., Perth, Western Australia, Australia), Iain M. Parnum (Ctr. for Marine Sci. and Technol., Curtin Univ., Bentley, Western Australia, Australia), Chong Wei, and Robert D. McCauley (Ctr. for Marine Sci. and Technol., Curtin Univ., Perth, Western Australia, Australia)

Fish produce sound to support life functions. Some species produce choruses, vocalizing continuously over a prolonged period, significantly raising background sound levels within a characteristic frequency band. Fish choruses demonstrate distinctive spatiotemporal patterns, the identification of which can provide information on fish distribution, spawning behaviors, habitat use, and responses to environmental and anthropogenic stressors. Many fish choruses have been reported in Australian waters, yet, very few source species have been identified. Analysis of previously collected acoustic recordings was undertaken to identify fish chorus presence along the southern Australian continental shelf. Three fish chorus types were identified with frequencies centered at approximately 1000, 2300, and 350 Hz. Acoustic recorders and unbaited underwater video recorders were then deployed simultaneously in an attempt to identify the source species of these fish choruses. The 1000 and 2300 Hz fish choruses were detected on the acoustic recordings, corresponding with video recordings of large aggregations of Red Snapper (*Centroberyx gerrardi*) and Deep Sea Perch (*Nemadactylus macropterus*). This pilot study was the first step in an attempt to develop an effective methodology that could be used to identify the source species of fish choruses present in offshore environments.

10:00–10:20 Break

10:20

5aAB7. The marine fish *Terapon theraps* calling in tropical Australia. Robert McCauley (Ctr. Marine Sci. and Technol., Curtin Univ., PO U1987, Perth, Western Australia 6854, Australia, R.McCauley@curtin.edu.au) and Douglas H. Cato (School of Geosciences, Univ. of Sydney, Sydney, New South Wales, Australia)

Calling by fish of the family Terapontidae, primarily *Terapon theraps*, is a dominant component of sea noise recorded in northern Australian tropical coastal waters. These fish produce nightly choruses over a few hours at frequencies from 50 Hz to a few kilohertz, with spectral peaks up to 30 dB above background. Choruses are heard over muddy bottoms <30 m depth. The two commonly heard calls range from 73–260 ms in length and comprised 11–21 pulses with repetition rates of 87–121 Hz. Calls were produced by muscle action on a two chambered swim bladder, and differed by muscle contraction rates, damping, and possibly the opening state of a sphincter separating chambers. Swimbladder carrier frequencies varied from 570–1465 Hz. Source levels varied from 141–154 dB re 1 mPa @ 1 m. Heavily damped alarm calls were sometimes heard. Chorusing fish schools were up to 2 km across although they were diffuse and actively calling fish extended beyond this region. A chorus could be detected out to 8 km from its center, well beyond the detection range of an individual call. Choruses may advertise school location during spawning, be used for mate attraction or mediate gamete release in dark turbid waters.

10:40

5aAB8. Two-dimensional mapping of impulsive ambient sound on a coral reef using vector sensors. Aaron M. Thode (Scripps Inst. of Oceanogr., Univ. of California San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0206, athode@ucsd.edu), Katherine Kim (Greeneridge Sci., San Diego, CA), and Lauren A. Freeman (Naval Undersea Warfare Ctr., Newport, RI)

Coral reef health is an important focus of tropical marine park management. Evidence exists that passive acoustic measurements of reef sound are associated with reef health and ecodiversity, but its utility would be enhanced by an ability to localize and map source distributions. Since much reef noise is impulsive, arising from both fish and crustaceans, localization could be achieved by collecting relative time-of-arrival information across synchronized spatially separated hydrophones. However, the sheer number of bioacoustic pulses in many reef environments can overwhelm attempts to associate sounds between sensors, Here we discuss how directional acoustic vector sensors sidestep these issues and permit 2-D mapping of thousands of pulses an hour, using data collected from two locations off the Big Island of Hawaii in 2020. At one location, three sensors spaced 20 m apart found acoustic activity was only present along the current-facing side of a 100 m long block reef. A second location set in the middle of more complex topography found several acoustic “hotspots” that remained stable over a month’s worth of measurements. A new generation of vector instrumentation is permitting this approach to be extended to crustacean noise in the kilohertz range. [Work sponsored by ONR and DARPA.]

11:00

5aAB9. Using quantitative soundscape analyses for coral reef microhabitat discrimination. Juan Carlos Azofeifa Solano (Ctr. for Marine Sci. and Technol., Curtin Univ., 85 South Terrace, Fremantle, Western Australia 6160, Australia, eazofeifa2@gmail.com), Christine Erbe (Ctr. for Marine Sci. and Technol., Curtin Univ., Perth, Western Australia, Australia), Rohan Brooker (AIMS, Crawley, Western Australia, Australia), Robert McCauley (Ctr. Marine Sci. and Technol., Curtin Univ., Perth, Western Australia, Australia), Daniel Pygas (AIMS, Crawley, Western Australia, Australia), William Feeney (Doñana Biological Station, Seville, Spain), Steve Simpson (Univ. of Bristol, Bristol, United Kingdom), Sophie L Nedelec (Univ. of Exeter, Exeter, United Kingdom), Evie Croxford (Univ. of Bristol, Bristol, United Kingdom), and Miles Parsons (AIMS, Perth, Western Australia, Australia)

The use of Ocean Sound to describe ecosystems and address ecological questions has increased remarkably during the last decades. The current capacity of recording, storing, and analyzing large amounts of data urges efforts toward developing standards and guidelines to advance underwater soundscape analyses. The development and use of metrics to summarize soundscape attributes have become widespread, but further field-testing and validation are still necessary to identify their ecological relevance resolution. We explore a combination of approaches to determine the performance of ecoacoustic indices and the recently proposed “Soundscape Code” in combination with ecological metrics (habitat and fish assemblage composition) for discrimination between coral reef microhabitats. We collected ecological and acoustic data from Lizard Island (Great Barrier Reef), where we established five transects each with five sampling stations, covering a range of microhabitats from the reef flat zone to the forereef zone, and from high live coral cover to rubble-dominated areas. Conspicuous differences were observed between the soundscapes of the live coral cover and the rubble-dominated areas, demonstrating the potential value of using soundscapes to discriminate between and monitor changes in coral reefs. [Work supported by the BHP-AIMS Australian Coral Reef Resilience Initiative.]

Session 5aBA

Biomedical Acoustics: General Topics in Biomedical Acoustics II: Bone and Soft Tissue Properties

Davide Fontanarosa, Cochair
QUT, Brisbane, Australia

Maria Antico, Cochair
Australian e-Health Research Centre, CSIRO, 296 Herston Rd, Brisbane 4006, Australia

Contributed Papers

8:20

5aBA1. Change of piezoelectric signal in cancellous bone with ultrasound attenuation. Atsushi Hosokawa (Dept. of Elec. and Comput. Eng., National Inst. of Technol., Akashi College, 679-3, Akashi 6748501, Japan, hosokawa@akashi.ac.jp)

Bone fracture healing can be accelerated by ultrasound irradiation. The bone formation can be accompanied by the piezoelectric effect in the bone. To establish the healing method for a joint bone, which is mostly occupied by cancellous bone, the piezoelectric properties in cancellous bone are required to sufficiently understand. Because of large ultrasound attenuation in cancellous bone, the ultrasound wave may be weakly transmitted to the deep part, and the piezoelectric signal may be weakly generated. In this study, the change of the piezoelectric signal in cancellous bone due to the ultrasound attenuation was investigated by numerical simulation. A cubic cancellous bone model was reconstructed from the X-ray microcomputed tomographic image of bovine bone. The cancellous bone models with different thicknesses were created by reducing the size in each direction parallel and perpendicular to the main trabecular orientation. Using a piezoelectric finite-difference time-domain (PE-FDTD) method, the piezoelectric signals generated in cancellous bone by ultrasound irradiation in the thickness direction were simulated, together with the ultrasound signal waveforms propagated through the bone. The variations of the piezoelectric and ultrasound signal amplitudes with the thickness showed that the piezoelectric signal could be scarcely detected at the deep part in cancellous bone.

8:40

5aBA2. Ultrasonic backscatter measurements of cancellous bone in the near field of a planar transducer. Blake Lawler (Phys., Rhodes College, 2000 North Parkway, Memphis, TN 38112, lawbc-24@rhodes.edu), Brent Hoffmeister, Ann Viano (Phys., Rhodes College, Memphis, TN), and Joel Mobley (Phys. and Astronomy, Univ. of MS, Oxford, MS)

Ultrasonic backscatter techniques may be used to detect changes in bone caused by osteoporosis and other diseases. Typical measurement locations include peripheral skeletal sites such as the heel, which may place the interrogated region of bone tissue in the acoustic near field of the transducer. The purpose of this study is to investigate how measurements in the near field of a planar transducer affect backscatter parameters used for ultrasonic bone assessment. Ultrasonic measurements were performed in a water tank using a planar 2.25 MHz transducer. Signals were acquired for five transducer-specimen distances: N/4, N/2, 3N/4, N, and 5N/4 where N is the near-field distance, a location that represents the transition from the near field to far field. Three backscatter parameters previously identified as potentially useful for ultrasonic bone assessment purposes were investigated: AIB, FSAB, and FIAB. All three parameters depended on transducer-specimen distance to varying degrees with FSAB exhibiting the greatest dependence on distance. These results suggest that laboratory studies of bone should evaluate the performance of backscatter parameters using clinically relevant transducer-specimen distances including distances where the ultrasonically interrogated region is in the near field of the transducer.

9:00

5aBA3. An ultrasonic phantom for cortical and cancellous bone. Kate E. Hazelwood (Phys., Rhodes College, 2000 North Parkway, Memphis, TN 38112, katehaze4@gmail.com) and Brent Hoffmeister (Phys., Rhodes College, Memphis, TN)

There is interest in developing ultrasonic techniques for diagnosing osteoporosis. Many techniques perform measurements at skeletal sites such as the heel, hip, and spine where the bone tissue consists of a non-porous outer layer of cortical bone that surrounds a porous interior region of cancellous bone. The goal of the present study was to develop an ultrasonic phantom that simulates this tissue configuration. A block of polymer open cell rigid foam (OCRF) was partially embedded in a thin layer of clear epoxy casting resin to create the phantoms. The resulting phantoms were 40 mm $\times$ 40 mm $\times$ 20 mm with one 40 mm $\times$ 40 mm face embedded in $\sim$3 mm of resin. Ultrasonic measurements were performed to characterize the speed of sound and attenuation of the resin and the OCRF separately and together, as configured in the phantom. Backscatter measurements were also performed. The resulting specimens can be used to investigate how non-normal incidence of an ultrasonic wave on the bone cortex may produce errors in these measurements.

9:20

5aBA4. Comparison of deep learning algorithms for quantitative diagnosis of fatty liver using multidimensional-moments heatmaps. Akiho Isshiki (Dept. of Medical Eng., Graduate School of Sci. and Eng., Chiba Univ., 1-33, Yayoicho, Inage-ku, Chiba, Chiba 263-8522, Japan, akiho.i@chiba-u.jp), Daisuke Okazaki (Dept. of Medical Eng., Faculty of Eng., Chiba Univ., Chiba, Japan), KIsako Fujiwara, Takayuki Kondo (Dept. of Gastroenterology, Graduate School of Medicine, Chiba Univ., Chiba, Japan), Kenji Yoshida, Tadashi Yamaguchi, and Shinnosuke Hirata (Ctr. for Frontier Medical Eng., Chiba Univ., Chiba, Japan)

Non-alcoholic fatty liver disease (NAFLD) carries a high risk of progressing to non-alcoholic steatohepatitis (NASH), cirrhosis, and hepatocellular carcinoma. Therefore, the quantitative diagnosis of NAFLD, i.e., determining hepatic fat deposition, is desired during ultrasound inspection. It is known that the deposition of fat droplets within hepatocytes increases the attenuation in ultrasound images. Because the texture of the attenuated ultrasound image becomes a homogeneous speckle, the probability density function of the echo envelopes can be approximated by a Rayleigh distribution. The diagnosis methods of NAFLD using echo-envelope statistics of ultrasound images have been reported. In this study, we focus on the distribution and variance of echo-envelope statistics and propose a new diagnosis method to analysis the characteristics of echo-envelope statistics using a deep learning algorithm (DLA). In the proposed method, first- and third-order moments as echo-envelope statistics are calculated at each pixel in the

ultrasound image. Then, the heatmap is formed from the distributions of both moments in each region of interest within the liver. The heatmaps are classified into the stages of fatty liver using DLA such as a convolutional neural network or a vision transformer. In this presentation, a comparative study on the accuracies by various DLA are reported.

9:40

5aBA5. Speed of sound of ultrasonic pulses in fluids relevant to biomedical research. Grace I. Nehring (Dept. of Phys., Rhodes College, 307 Forest Bay Ct, Belmont, NC 28012, nehgi-25@rhodes.edu), Emily E. Bingham, Brent Hoffmeister, and Ann Viano (Dept. of Phys., Rhodes College, Memphis, TN)

Tissue specimens are commonly stored in phosphate buffered saline (PBS) solution and formalin solution. Ultrasonic measurements of tissues may be performed in these same solutions. The goal of the present study was to measure the speed of sound in these fluids over a typical range of room temperature. An aluminum block with a 0.500-in. precision machined step was placed in a 1 L specimen tank containing the fluid of interest. The specimen tank was positioned in an 8 L tank of water with a thermostatically controlled heater and circulating pump, to which cold packs could also be added. A 7.5 MHz transducer was mechanically moved and scanned over the block to acquire echoes from either side of the step. The time difference between echoes received from either side of the step was measured to determine the speed of sound in the fluid. Measurements were performed in 0.5°C intervals from 15.0 to 25.0°C. The speed of sound was found to increase with temperature for both fluids, with ranges of 1484–1512 m/s for PBS and 1509–1532 m/s for formalin.

10:00–10:20 Break

10:20

5aBA6. Ultrasonic properties of fresh and formalin fixed brain tissue. Amalia M. Bay (Dept. of Phys., Rhodes College, Rhodes College, 2000 North Parkway, Memphis, TN 38112, bayam-25@rhodes.edu), Grant R. Jenson, Brent Hoffmeister (Dept. of Phys., Rhodes College, Memphis, TN), Cecille Labuda (Dept. of Phys. and Astronomy/National Ctr. for Physical Acoust., Univ. of MS, University, MS), Ann Viano, Kate E. Hazelwood, Phyu Sin M. Myat, and Blake Lawler (Dept. of Phys., Rhodes College, Memphis, TN)

Recent advances in transcranial ultrasound have generated increased interest in the ultrasonic properties of brain. For *in vitro* laboratory studies of brain tissue, it is convenient to use specimens preserved by formalin fixation. The goal of the present study was to investigate the effect of formalin fixation on the ultrasonic properties of brain. 1-cm thick slices of tissue were prepared from 9 bovine brains. A total of 28 slices from two anatomic planes (16 sagittal, 12 coronal) were studied. Tissue specimens were scanned in phosphate buffered saline at room temperature with a 5 MHz transducer. Scans were performed on fresh tissue and repeated one month after formalin fixation. The speed of sound (SOS) and frequency slope of attenuation (FSA) of ultrasonic pulses propagated through the tissue were determined at every measurement site. The measured values were averaged over all sites on all specimens to obtain the following results expressed as mean $\pm$ standard deviation. SOS was (1522 ± 3) m/s before fixation and (1538 ± 12) m/s after fixation. FSA was (0.438 ± 0.064) dB/cm/MHz before fixation and (0.493 ± 0.113) dB/cm/MHz after fixation. Thus, formalin fixation produced approximately a 1% increase in SOS and a 13% increase in FSA.

10:40

5aBA7. Comparing *in vivo* wave-based elastography with *ex vivo* mechanical testing in porcine vessels. Charles Capron (Biomedical Eng. and Physiol., Mayo Clinic, 200 1st St. SW, Rochester, MN 55901, capron.charles@mayo.edu), Hyoung-Ki Lee, Prabh Singh, Sreenivasulu Kilari, Gang Liu (Radiology, Mayo Clinic, Rochester, MN), Allison Tanner (Biomechanics Core Facility, Mayo Clinic, Rochester, MN), Tuhin Roy (Biomedical Eng., Columbia Univ., New York, NY), Murthy N. Guddati (Civil, Construction, and Environ. Eng., North Carolina State Univ., Raleigh, NC), Sanjay Misra, and Matthew W. Urban (Radiology, Mayo Clinic, Rochester, MN)

Arterial stiffness predicts cardiovascular disease, the leading cause of death worldwide. Ultrasound elastography methods that measure wave velocity have promise for evaluating the stiffness of blood vessels *in vivo*. However, vessel geometry modulates wave velocity by inducing geometric dispersion. Here, we compare Young's modulus E estimates obtained using *in vivo* wave velocity measurements and techniques that either do or do not consider geometry to *ex vivo* reference E values from mechanical testing. An acoustic radiation force was applied to generate propagating waves in the right common carotid artery ($N = 9$) and external jugular vein ($N = 10$) of anesthetized pigs. Vessels were imaged with ultrafast ultrasound to measure wave motion. Group velocity c_g was estimated from wave motion using a Radon transform. E was estimated from c_g using a bulk media assumption which neglects geometry, as well as the Moens–Korteweg equation and a semi-analytical finite element model, which consider geometry. Pigs were euthanized and vessels were excised. Ring-shaped samples were extracted and tested in a uniaxial tension setup to obtain reference E values for comparison with E values obtained from wave-based techniques. Techniques considering geometry exhibited higher correlation with reference E values, underscoring the need to incorporate geometry for accurate vascular elastography assessments.

11:00

5aBA8. Evaluation backscattering coefficients of heterogeneous media containing multispecies scatterers and lipids. Hayato Kutsuzawa (Graduate School Sci. and Eng., Chiba Univ., 1-33, Yayoi-cho,Inage-ku, Chiba-shi, Chiba 263-8522, Japan, h_kutsuzawa@chiba-u.jp), Shinnosuke Hirata, Kenji Yoshida (Ctr. for Frontier Medical Eng., Chiba Univ., Chiba, Japan), Emilie Franceschini (Lab. of Mech. and Acoust., CNRS/Aix-Marseille Univ., Marseille, France), and Tadashi Yamaguchi (Ctr. for Frontier Medical Eng., Chiba Univ., Chiba, Japan)

Many attempts have been reported to evaluate tissue characterization from echo signals using the backscatter coefficient (BSC) as an indicator. In the evaluation of the BSC of the biological tissues, it is known that the effects of attenuation properties such as scattering attenuation and absorption attenuation, as well as interference between different scatterers, are significant. In this study, the effects of attenuations and interference were verified by evaluating a phantom with high absorption attenuation and a phantom with a mixture of two types of scatterers. Both the reflector method and the reference phantom method were applied to BSC evaluation for these phantoms, and the accuracy of the evaluation was verified. The results suggest that the effect of the acoustic impedance ratio between the scatterer and the surrounding medium tends to be reflected in the accuracy of the evaluation in media with large absorption attenuation. On the other hand, the difference between the mathematical theory and the actual evaluation results was observed due to interference effects. In terms of the accuracy of the evaluation method, the reflector method is effective in the vicinity of the focus, while the reference phantom method is effective in terms of application.

11:20

5aBA9. Verification of effect of analytical conditions of shear wave velocity on ability to evaluate microstructure. Kodai Osato (Graduate School of Sci. and Eng., Chiba Univ., 1-33 yayoi-cho inage-ku, Ctr. for Frontier Medical Eng., Chiba-shi, Chiba-ken 263-8522, Japan, osato@chiba-u.jp), Takuma Oguri, Naohisa Kamiyama (GE Healthcare Japan, Tokyo, Japan), Shinnosuke Hirata, Kenji Yoshida, and Tadashi Yamaguchi (Chiba Univ., Chiba, Japan)

In this study, the relationship between the analytical conditions of shear wave velocity (SWV) and the evaluation ability of microstructure were verified by FDTD simulation. By adding acoustic radiation force (ARF) to the left edge of the simulation space, which mimicked various fatty livers with different fat mass, shear waves propagating in the lateral direction were simulated in each liver. SWV was calculated using the time difference calculated by the cross-correlation method from two timewaves laterally located in the simulation. By varying the resolution of the SWV evaluation, the effect of scattering and refraction of shear waves in fat droplets, etc. on the macroscopic shear propagation was evaluated. Although shear waves could be evaluated stably when tracking shear waves at the spatial step (time width) of the clinical device level, the effects of individual fatty substances were strongly indicated when tracking in the microscopic sense of 10 μm (50 ns). When there was a thick fat layer or a sparse muscle layer near the body surface, the irradiation of ARFI did not work as expected and the evaluation accuracy of SWV decreased.

11:40

5aBA10. Ultrasonic spectroscopy at 20–80 MHz to evaluate lymph node status during breast conserving surgery. Timothy E. Doyle (Calliope Biophys. LLC, 47 N 970 W, Orem, UT 84057, timothydoyle256@gmail.com), Dolly A. Sanjinez (Calliope Biophys. LLC, Orem, UT), Audrey P. Butler (Audrey Butler Consulting LLC, Fairview, UT), Andrea F. Yzaguirre (Intermountain Medical Ctr., Murray, UT), Garrett M. Wagner (Calliope Biophys. LLC, Orem, UT), and Huda A. Al-Ghaib (Deloitte LLP, Arlington, VA)

High-frequency (20–80 MHz) ultrasound was used to evaluate 77 lymph nodes from 38 patients to identify metastatic breast cancer. Lymph nodes were resected during routine breast conserving surgery. Through-transmission point measurements were collected from the resected nodes immediately following surgery using two single-element transducers (50-MHz center frequency, 6.35-mm diameter), a high-frequency ultrasonic pulser-receiver, and a 1-GHz digital oscilloscope. Attenuation and two spectral parameters—peak density and tortuosity—were calculated from the ultrasonic waveforms and power spectra, respectively. Fisher's exact test was used to optimize the sensitivity and specificity for each parameter separately and for multivariate analyses. A multivariate analysis of peak density versus tortuosity with a cube-root decision boundary produced the most significant results, with an 89.6% accuracy, 100% sensitivity, 88.4% specificity, and p-value of 6.12×10^{-7}. Computer simulations and phantom experiments showed that the sensitivity of the spectral parameters to tissue malignancy arises from ultrasonic Mie scattering from cell nuclei and nuclear pleomorphism. The results demonstrate that ultrasonic spectroscopy at 20–80 MHz provides high sensitivity and specificity for malignant lymph nodes in the breast, and has promise as a rapid, intraoperative, and potentially *in vivo* diagnostic tool for surgeons and a wide range of soft tissue cancers.

Session 5aCA

Computational Acoustics: Topics in Computational Acoustics

Alex Higgins, Chair
Electrical & Computer Eng., Portland State Univ., Portland, OR 97201

Contributed Papers

8:00

5aCA1. An acoustic cloaking design considering flow effects by using topology optimization. Jingjing Zhu (College of Eng., Peking Univ., Beijing 100871, China, 2101112014@stu.pku.edu.cn) and Xun Huang (Peking Univ., Beijing, China)

A topology optimization method is utilized to design an acoustic cloak in the presence of background mean flows, highlighting both the scientific significance and practical importance of acoustic invisibility in fluids. This study focuses on understanding the inherent physical mechanism and considering realistic constraints for actual manufacturing, ultimately inspiring the development of an optimized cloak that is available for practical fabrication in the presence of flows. Here, we use the density-based topology optimization to assign specified materials and impose practical constraints on the material allocation to achieve the desired cloaking performance. The optimization problem is then efficiently solved using the gradient-based globally convergent method of moving asymptotes, leveraging the derivative information from the finite element simulation studies of the linearized acoustic potential equation. To validate the proposed method, several numerical simulations are conducted, exploring different frequencies, incident flow speeds, and incident angles of flow. The results confirm the effectiveness and efficiency of the optimization method, expanding the range of applications of the acoustic cloak and contributing to a deepened physical understanding of how it manipulates sound waves in flows.

8:20

5aCA2. Ventilated silencer based on layered Helmholtz resonators. Alexandru Crivoi (School of Mech. and Aerosp. Eng., Nanyang Technolog. Univ., Nanyang Ave. 50, Singapore 639798, Singapore, acrivoi@ntu.edu.sg), Liangfen Du, and Zheng Fan (School of Mech. and Aerosp. Eng., Nanyang Technolog. Univ., Singapore, Singapore)

This study investigates the performance of a multilayer sound-attenuating metamaterial with ventilation capabilities. The design incorporates arrays of Helmholtz resonators embedded within the wall structure. Each cell of the barrier's front wall is a Euclidean polygon, triangular, square, or hexagonal, housing a parallel array of resonators interconnected via an axial ventilation duct. The sound attenuation performance of these barriers is meticulously examined using the lumped parameter theory and the transfer matrix method (TMM), complemented by finite element (FE) simulations. The results indicate that sound attenuation in the audible frequency range (300–2000 Hz) can be significantly enhanced by employing multiple layers of resonators, each layer assigned a unique peak resonance frequency. Despite the design's inherent trade-offs between barrier thickness, ventilation capacity, operational frequency range, and overall sound attenuation, its structural simplicity and straightforward theoretical performance estimation methods make it a promising candidate for applications requiring a specific attenuation spectrum. The sound-blocking performance of the design is further validated through impedance tube experiments, which show a reasonable agreement with the numerical predictions, thereby affirming the efficacy of the proposed design. The performance of multilayer design is also compared with the similar ventilated barriers proposed in the recent studies.

8:40

5aCA3. Wave injection in the proximity of a target and propagation of the backscattering to the far-field. Alessandro Monti (CMRE, STONATO, La Spezia, Italy, Alessandro.Monti@cmre.nato.int), Roberto Sabatini (Embry-Riddle Aeronautical Univ., Daytona Beach, FL), Yan Pailhas, and Angeliki Xenaki (CMRE, STO-NATO, La Spezia, Italy)

Direct numerical simulations of the three-dimensional acoustic far field scattered by a target prone or buried under the seafloor are still computationally demanding. They become prohibitive when the distances source-target and target-receiver are of the order of hundreds-to-thousands of wavelengths. At the same time, the interaction between the target and the wave induced by the source occurs in a narrow region around the object. In light of this, we propose to reduce the computational burden by splitting the task into three major steps. First, the source field is assumed known analytically and is injected into a relatively small computational box surrounding the target. Second, the linearized equations of continuum mechanics are solved within the box to calculate the acoustic and elastic near fields. Finally, the acoustic backscattered far field is computed via the Helmholtz–Kirchhoff theorem. Idealized validation tests will be examined during the talk. The presentation will be enriched with the discussion of results for more realistic target scattering scenarios.

9:00

5aCA4. Computational optimization of acoustofluidic devices for advanced particle micromanipulation. Kirill Kolesnik (Biomedical Eng., Univ. of Melbourne, 203 Bouverie St., Melbourne, Victoria 3053, Australia, kkolesnik@student.unimelb.edu.au), Vijay Rajagopal, and David J. Collins (Biomedical Eng., Univ. of Melbourne, Melbourne, Victoria, Australia)

Acoustofluidic devices, which combine principles of acoustics and microfluidics, have emerged as a promising platform for biological micro-object micromanipulation due to their non-invasive, accurate, rapid, and label-free qualities. Acoustofluidic devices have found utility in various biomedical applications including single-cell studies, point-of-care testing, lab-on-a-chip studies, and tissue engineering. In this work, we present novel device configurations which enable complex and high-resolution control of suspended micro-objects. The studies presented here utilize computational analysis to optimize (1) traveling surface acoustic wave device dimensions, (2) the configuration of a planar acoustic resonator that integrates a structured surface, (3) the thickness of the coupling layer and superstrate materials for bulk-wave transmission, and (4) the shape of acoustically actuated 3-D microstructures. These devices were verified experimentally to generate highly localized acoustic fields and acoustic streaming effects enabling rapid and spatially controllable particle capture, transport, and patterning. In doing so, this work demonstrates that compu-tational analysis is an integral part of the development of acoustofluidic devices for advanced micromanipulation, which have extensive potential in biomedical applications.

9:20

5aCA5. High performance calculation of nonlinear wave propagation on GPU for focused ultrasound treatment planning. Maxim Solovchuk (Inst. of Biomedical Eng. and Nanomedicine, National Health Res. Insti- tutes, No. 35, Keyan Rd., Zhunan, Miaoli County 35053, Taiwan, solovchuk@gmail.com)

High intensity focused ultrasound (HIFU) is very promising new tech- nology that has many therapeutic application, and among them are the treat- ment of cancer in different organs without major side effects. Some of the limitations of HIFU are the long treatment time and incomplete ablation. High performance computing can help in this regard. Simulation of nonlinear wave propagation in three dimensional geometry is very time consuming process especially when shocks are presented. In the current work, we are going to present three dimensional parallel solver on GPU. Some examples for the simulation in a patient specific geometry will be also presented. The presented results can be used for the optimization of the treatment. For the validation of the results, ex-vivo and in-vivo experiments have been per- formed. It has been shown that large volume can be ablated within a short period of time.

9:40

5aCA6. An analysis of the efficiency and accuracy of applying Huygen's principle to acoustic scattering models. Aubrey Espana (Acoust., Appl. Phys. Lab, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, aespana@apl.washington.edu), Steven Kargl (Acoust., Appl. Phys. Lab, Univ. of Washington, Seattle, WA), and Ahmad Abawi (Heat, Light & Sound Res., Inc., La Jolla, CA)

Over the last decade, the Target-in-the-Environment-Response (TIER) model has proven to be an efficient simulation tool for computing the acoustic scattering from axisymmetric objects in an underwater environment. The model functions by first computing the free-field scattered pressure via a high-fidelity Finite Element (FE) computation. The scattered pressure is converted into a scattering amplitude and stored in a look-up table. These scattering amplitudes are used in a fast ray model to compute the acoustic scattering from an object on or embedded in a seafloor, for any vehicle path imaginable. Computing the scattered pressure amplitude for the look-up tables via the FE method is a bottleneck of the computation. For monostatic scattering problems, the look-up table historically sampled the azimuthal angle domain with 1-deg resolution. However, this approach is not efficient in time nor disk space when moving to bistatic scattering situations. Here, we present a more efficient alternative, which is based on application of Huygen's principle. The results from this method are assessed against the original sampling method and validated against additional high-fidelity models including a hybrid 2-D/3-D FE-Propagation model and a fully 3-D Impedance Matrix method, the latter of which is numerically exact and offers a benchmark solution.

10:00

5aCA7. A novel and computationally efficient subgrid technique for modeling scattering and Doppler effects of rough dynamic sea surfaces using the finite-difference time-domain method. Alex Higgins (Elec. & Comput. Eng., Portland State Univ., 1900 SW 14th Ave., Ste. 25-01, Portland, OR 97201, higginja@ece.pdx.edu) and Martin Siderius (Elec. & Comput. Eng., Portland State Univ., Portland, OR)

Over the last decade, the improvements in computational performance that multiprocessing, solid-state hard drives, and fast memories have brought to desktop workstations have made modeling and simulation methods that were previously inaccessible into the office workspace. These methods were previously restricted to large servers or computer clusters because of their demanding computation and large memory requirements when applied to underwater acoustic simulation scenarios. One such method is the Finite-difference Time-domain (FDTD) technique. The FDTD method is a discrete numeric model that provides the full pressure wave solution solved directly in the time domain. This makes it an ideal model for investigating scattering and Doppler effects of boundaries with complex geometries and motion. A fully developed sea surface generated using the Pierson–Moskowitz wavenumber spectra meets these criteria. A novel subgrid technique has been developed to improve the accuracy of signals scattered from fully developed sea surfaces. This method remains computationally efficient because the subgrid is applied only along the rough sea surface as it varies over time. The FDTD subgrid method has been applied to simulations that investigate the "frozen" sea surface assumption that is utilized by other traditional models that incorporate boundary motion.

Session 5aMU

Musical Acoustics: Player–Instrument Interaction I

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

Andre Almeida, Cochair
Physics, UNSW, 8/133 Boundary Street, Clovelly 2031, Australia

John Smith, Cochair
Physics, UNSW, Sydney 2052, Australia

Chair's Introduction—7:55

Invited Papers

8:00

5aMU1. Towards data-driven physical modeling synthesis. Champ C. Darabundit (Music Technol., McGill Univ., 550 Rue Sherbrooke O, Montréal, QC H3A 1B9, Canada, champ.darabundit@mail.mcgill.ca) and Gary Scavone (Music Technol., McGill Univ., Montreal, QC, Canada)

A current research trend is the combination of physical models with machine learning. These hybrid models leverage the strength of both fields and require smaller datasets in training. Trained models are interpretable and well suited towards solving inverse problems. We present recent advancements on differentiable digital waveguide and reed modeling. The proposed models are implemented in the PyTorch framework and can extract physical parameters from audio data. The proposed models allows us to examine player interactions by retrieving player parameters from audio data.

8:20

5aMU2. Developing methodologies to correlate perceived sound qualities of violins with controlled construction parameters. Claudia Fritz (Institut Jean le Rond d'Alembert, UPMC, 4 Pl. Jussieu, Paris 75005, France, fritz@lam.jussieu.fr)

The holy grail for violin makers is to find correlations between construction parameters and sound qualities. This is challenging for two main reasons: it is difficult to build violins reliably enough to ensure that the change in the sound is indeed a result of the change of construction parameters; when listening to the violins being played, differences seem to be smoothed out by the players who adapt very quickly. Therefore, while players had so far been preferred in our experiments to maximize the ecological validity and take into account the complexity of the interaction between the player and the instrument, we have decided to test whether other methods, which reduce the influence of the player but are quite artificial, may be useful to explore the influence of some construction parameters on the tone. In the context of two sets of violins built with controlled thickness variations of their plates, we will compare the results of listening tests based on real recordings with a player and with a bowing machine as well as synthesized recordings (from the convolution of an excerpt recorded with piezo sensors and radiation measurements in an anechoic chamber) and discuss them in the light of audio descriptors on the recordings and vibroacoustical measurements on the violins.

8:40

5aMU3. Models of the player's interactions with strings. Gianpaolo Evangelista (IKE, Univ. of Music and Performing Arts Vienna, Anton-von-Webern-Platz 1, Vienna, Wien 1030, Austria, gianpaevan@gmail.com)

In the context of physically inspired models for the synthesis, the development of accurate models for the interaction of the player with the instrument plays a fundamental role. Depending on the musical instrument, the palette of possible interactions is generally very broad and includes the coupling of body parts, mechanical objects, and/or devices with various components of the instrument. In this work, we focus on models of the interaction of the player with strings, which include elastoplastic models of the fingers, dynamic models of the bow, the plectrum and the friction of objects such as bottle necks. We consider interactions both at the stimulation side (e.g., plucking or bowing) and at the fingerboard side, where collisions and/or imperfect pressures introduce nonlinear effects. Other possible interactions in less conventional play modes such as those based on rotating motors and electromagnetic stimulation are taken into account in this work. The proposed models do not depend on the specific implementation of the physical model in, e.g., Finite-Difference Time-Domain (FDTD)-based or waveguide-based schemes. While the former class of implementations may produce more accurate results, the latter allows for simplifications and lower computational costs provided that stability can be guaranteed.

9:00

5aMU4. Exploring player's feelings on clarinet mouthpiece geometry using artificial blowing machines and airflow simulations. Tsukasa Yoshinaga (Graduate School of Eng. Sci., Osaka Univ., 1-1 Tempaku, Hibarigaoka, Toyohashi 441-8580, Japan, yoshinaga@me.tut.ac.jp), Hiroshi Yokoyama (Toyohashi Univ. of Technol., Toyohashi, Japan), Tetsuro Shoji, Akira Miki (Yamaha Corp., Hamamatsu, Japan), and Akiyoshi Iida (Toyohashi Univ. of Technol., Toyohashi, Japan)

The mouthpiece geometry of single-reed instruments has been known to have a great impact on sound quality and playability. Although many researchers have investigated the effects of the geometry using artificial blowing machines or user case studies, few attention has been paid to the essential cause of physical changes in the instrument. In this study, we propose a method combining the artificial blowing machine and numerical flow simulations to explore the player's feelings about the airflow and sound generation in the clarinet mouthpiece. With the artificial blowing machine, the mouth pressure, as well as lip force, were systematically changed using a pressure regulator and strain gauge. At particular conditions, the numerical flow simulation was conducted by solving fluid–structure interactions with the Navier–Stokes equations and the one-dimensional beam equation. The playability of the mouthpiece was shown by the changes in oscillation threshold pressures and sound spectral characteristics, whereas the cause of the changes could be described by the airflow pressures inside the mouthpieces. The results suggest that the combination of the artificial blower and the flow simulations can be effectively used for clarifying the cause of changes in players' feelings.

Contributed Papers

9:20

5aMU5. Physical aspects of live clarinet performances. Andre Almeida (Phys., UNSW, 8/133 Boundary St., Clovelly, New South Wales 2031, Australia, a.almeida@unsw.edu.au), Weicong Li (MARCS, Western Sydney Univ., Sydney, New South Wales, Australia), Joseph Wolfe (Phys., UNSW, Sydney, New South Wales, Australia), John Smith, and Emery Schubert (UNSW, Sydney, New South Wales, Australia)

The clarinet is arguably the wind instrument that has been most extensively studied and described from the point of view of its physics. It has a relatively simple geometry, its embouchure can be fairly well approximated by a set of two control variables, and the air column can be accessed relatively easily, allowing for the measurement of control variables and acoustic variables close to the mouthpiece. As a result, extensive datasets have been collected, allowing understanding of how the sound produced by the instrument changes as a function of its control parameters. In this presentation, we explore a new dataset of control variables (mean reed displacement, blowing pressure, bite position, and others) and acoustic variables (sound level, pitch, spectral centroid, and others) recorded in performances on the clarinet by expert players. These data are compared with measurements and modeling of the instrument blown in mechanical conditions.

9:40

5aMU6. Two-dimensional playability maps for single-reed woodwind instruments. Vasileios Chatziioannou (Dept. of Music Acoust., Univ. of Music and Performing Arts Vienna, Anton-von-Webern-Platz 1, Vienna 1030, Austria, chatziioannou@mdw.ac.at) and Alex Hofmann (Dept. of Music Acoust., Univ. of Music and Performing Arts Vienna, Vienna, Austria)

One method to analyze musical instrument playability is by visualizing a two-dimensional subspace of the musician's control parameters. This has been widely used in the form of Schelleng diagrams for bowed string instruments. There, it is possible to identify regions within the bow force—bowing position subspace where Helmholtz motion is achieved. Such diagrams may be populated either on the basis of experimental measurements, or via physical modeling. In fact, physical modeling is particularly suited to this task, since playing parameters can be directly controlled. It has been recently suggested (Woodhouse, in Proc. SMAC 2023) that similar diagrams may be used for analyzing wind instrument playability. This study aims at exploring this direction using a physical model which has been previously validated against experimental measurements. Initially, an informed decision is made on which parameters should be chosen for plotting an equivalent playability diagram. Subsequently, various diagram versions are generated, focusing on different aspects of the generated tones. As already pointed out by Woodhouse, similarities appear between the diagrams for bowed-string and woodwind instruments. It remains to examine how valuable information regarding woodwind playability may be extracted from such studies.

Session 5aNSa

Noise and Physical Acoustics: Jet Noise

Daniel Edgington-Mitchell, Cochair
Mechanical and Aerospace Engineering, Monash University, Department of Mechanical and Aerospace Engineering, 14 Alliance Lane, Clayton 3800, Australia

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

Chair's Introduction—7:55

Contributed Papers

8:00

5aNSa1. Recent findings regarding noise from tactical jet engines. Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., N281 ESC, Provo, UT 84602, kentgee@byu.edu), Matthew A. Christian (Dept. of Mech. Eng., Brigham Young Univ., Provo, UT), Hunter J. Pratt, and Michèle L. Eggleston (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

This paper reviews the outcomes of several recent investigations into full-scale supersonic jet noise. First, concordance has been shown between a T-7A-installed GE F404 engine's sound power and classical jet noise theory in both the subsonic and supersonic regimes [M. A. Christian *et al.*, *JASA Express Lett.* **3**, 073601 (2023)]. The results indicate approximate agreement between the "eighth-power" law for subsonic engine conditions and the "third-power" law for supersonic engine conditions, with a radiation efficiency of ~0.5–0.6% at afterburner. However, a sudden jump in radiation efficiency in the jet's transonic region is not presently explained. Second, normalized far field-derived sound power spectra for the F404 do not agree well with a curve found in NASA SP-8072 (Eldred, 1971), likely indicating differences between the importance of different Strouhal number regimes in military jet and rocket noise and, possibly, different radiation phenomena. Third, the maximum directivity angle for overall level appears to be primarily driven by the convective Mach number associated with Kelvin–Helmholtz instability waves (M. A. Christian *et al.*, AIAA Paper 2023-3351). Together, these recent findings represent improved understanding of full-scale tactical jet noise phenomena. [Work supported by ONR.]

8:20

5aNSa2. Acoustic characterization of an installed elliptical jet. Jayson R. Beekman (Mech. and Aerosp. Eng., Monash Univ., Clayton, Victoria 3800, Australia, jayson.beekman@monash.edu), Joel Weightman, Petronio Nogueira (Mech. and Aerosp. Eng., Monash Univ., Clayton, Victoria, Australia), Peter Jordan (Fluid Sci., Thermal Sci. and Combustion, Institut Pprime - CNRS- Université de Poitiers-ENSMA, Poitiers, France), and Daniel Edgington-Mitchell (Mech. and Aerosp. Eng., Monash Univ., Clayton, Victoria, Australia)

This work demonstrates a preliminary acoustic characterization of an installed elliptical jet with an aspect ratio of 2. This configuration is compared against the nominal axisymmetric installed jet case of equivalent diameter. A baseline case for the elliptical and axisymmetric nozzle geometries are obtained in the uninstalled configuration. For all cases, the acoustic Mach number is varied between 0.4 and 1.0, the azimuthal observer angle between 0° and 90°, and the axial distance from the jet exit to the observer between 0 and 30 nozzle equivalent diameters. The latter of these is an analogue for polar angle variation by keeping the radial distance from the jet to the observer constant. A flat metal plate is used for the installed case, creating a jet–plate interaction mimicking that of an aircraft engine's exhaust and its wing. Using the same variations as the free jet, parameters such as plate height, trailing-edge distance and, for the elliptical jet, nozzle exit rotation angle are investigated to give a broad characterization of the installed elliptical jet and enable direct comparison to the axisymmetric jet case. The key comparison metric for this work is the overall sound pressure level and polar directivity of the acoustic fields.

8:40

5aNSa3. Abstract withdrawn.

9:00

5aNSa4. Two distinct mechanisms of tonal sound production in supersonic jet impingement. Daniel Edgington-Mitchell (Dept. of Mech. and Aerosp. Eng., Monash Univ., 14 Alliance Ln., Clayton, Victoria 3800, Australia, daniel.m.mitchell@gmail.com), Joel Weightman, and Petronio Nogueira (Mech. and Aerosp. Eng., Monash Univ., Clayton, Victoria, Australia)

The interaction of high-speed jets with a perpendicular surface produces not only broadband sound, but "impingement tones," that can be more than 30 dB above the broadband noise. It is well recognized that these tones are the result of an aeroacoustic resonance, driven by a feedback loop between the jet nozzle and the plate. However, the exact mechanism by which the tones are produced remains a topic of some dispute; some researchers locate the source within the core of the impinging jet, others in the wall jet generated after impingement with the plate. In this work, we demonstrate that both camps have correctly identified sources; there are two distinct mechanisms, with two distinct locations. The relative strength of these sources depends on the geometry of the problem, but we show that they can actually coexist at a single condition. Decomposition of high-speed schlieren data directly visualizes the location of these sources as a function of jet operating condition.

9:20–9:40 Break

9:40

5aNSa5. Convolutional neural networks for tracking coalescing waveforms in supersonic jet noise. William A. Willis (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX 78713-8029, william.willis@utexas.edu), John A. Valdez, Charles E. Tinney, and Mark F. Hamilton (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

High-speed schlieren images of the field surrounding a supersonic jet provide a rich database for testing convolutional neural network (CNN) classification methods due to the presence of many waveform structures of interest. One such structure is waveform coalescence, where waves intersect at small angles, which can lead to increased steepening and nonlinear distortion in the jet near field [Willis *et al*., AIAA Journal (2023)]. Although waves exhibiting coalescence behavior have been identified in narrow field-of-view (FOV) schlieren images, new methods are needed for large FOV images to improve computational efficiency. This presentation will explore methods to track and classify waves observed in large FOV schlieren images as coalescing or noncoalescing using CNNs. Using transfer learning, pretrained networks can be retrained for this problem, with training data obtained from narrow FOV schlieren or from two-dimensional simulated waveforms using the Khokhlov–Zabolotskaya–Kuznetzov (KZK) equation. The impact of model hyperparameters, choice of pretrained network, image scaling, and other factors on the results of waveform classification by the CNN will be explored. [WAW is supported by the ARL:UT Chester M. McKinney Graduate Fellowship in Acoustics.]

Invited Paper

10:00

5aNSa6. Development and evaluation of laser acoustic measurement system for supersonic jet noise. Kazuki Matsumoto (The Univ. of Tokyo, 5-1-5, Kashiwanoha, Kashiwa, Chiba 277-8561, Japan, 2590122433@edu.k.u-tokyo.ac.jp), Youngha Kwon (The Univ. of Tokyo, Kashiwa, Japan), Koji Okamoto (The Univ. of Tokyo, Chiba, Chiba, Japan), Masahito Akamine, and Susumu Teramoto (The Univ. of Tokyo, Bunkyo, Tokyo, Japan)

Acoustic measurements in the near field are significant to suppress the noise generated by a supersonic jet. The microphone measurement is commonly used to measure noise, but it is not appropriate for the jet noise measurement in the near field, because it may be broken or disturb the sound field. In this study, we propose an optical acoustic measurement method using a laser and a 2-D position sensitive detector. This system can measure the propagating direction and the spectra of acoustic waves by detecting the angle and direction of laser path deflection by acoustic wave passing. To evaluate this laser measurement method, it was applied to various supersonic jet noise phenomena, such as Mach waves and screech tones, and then the results were compared with those of microphone measurements. The spectra measured by the laser measurement show good agreement with those by microphone measurements, and the difference in OASPL was less than 0.7 dB. As for the propagating direction, it is calculated using the principal component analysis in the laser measurement, and its results also show good agreement with those of acoustic intensity vector measurement.

Contributed Papers

10:20

5aNSa7. The effect of cant angle on the sound production of mutually inclined rectangular supersonic jets. Daniel Edgington-Mitchell (Dept. of Mech. and Aerosp. Eng., Monash Univ., 14 Alliance Ln., Clayton, Victoria 3800, Australia, daniel.m.mitchell@gmail.com), Joel Weightman, and Petronio Nogueira (Mech. and Aerosp. Eng., Monash Univ., Clayton, Victoria, Australia)

Rectangular nozzle geometries offer a number of benefits over their axisymmetric counterparts, particularly for military aviation. Among these benefits are altered thermal and noise signatures, ease of airframe integration, and simpler thrust vectoring. Many next-generation tactical aircraft concepts employ two rectangular jets in close proximity. These jets can couple together, producing high-amplitude acoustic tones that are deleterious to aircraft structure. In this work, we take a combined experimental and theoretical approach to examine the effect of cant angle (i.e., mutual inclination angle) on the jets' proclivity to couple. We employ modal decomposition of flow visualization data alongside resonance predictions based on linear stability theory to explore the degree to which mutual inclination can suppress resonance.

10:40

5aNSa8. An investigation of shock cell-related noise in an installed, full-scale tactical jet engine using acoustical holography. Logan T. Mathews (Phys. and Astronomy, Brigham Young Univ., N201 ESC, Provo, UT 84602, lmathew3@byu.edu) and Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Shock cells in imperfectly expanded supersonic jet flows have been found to contribute to the noise radiated by these jets. Acoustic phenomena such as screech and broadband shock-associated noise have been linked to the shock cell structure in jets. Acoustical holography has been used to study various jet noise source and radiation characteristics. Holography-based reconstructions near the acoustic source in an installed, full-scale tactical jet engine have yielded features indicative of the known shock cell structure in the jet. This paper presents these results and investigates the characteristics of these features, including their distribution and relative amplitudes.

Session 5aNSb

Noise: Construction Noise

Irene Van Kamp, Cochair
National Institute for Public Health and the Environment, The Netherlands

Tracy Gowen, Cochair
Renzo Tonin and Assoc., Elizabeth St., Surry Hills, New South Wales 2010, Australia

Contributed Papers

8:00

5aNSb1. Is an exceedance always a non-compliance? Laurence Clark (AAS, Unit 27, 43-53 Bridge Rd., Stanmore, New South Wales 2093, Australia, larry.clark@acousticstudio.com.au)

Consider a typical scenario taking place across NSW every day: construction on a major infrastructure project. The contractor has previously assessed the noise impact in a report as required by Regulators. But the actual noise levels exceed the levels in their report. Are they "non-compliant"? Yes, for respite nights when construction noise must not exceed Noise Management Levels (NMLs). By contrast, NMLs can be exceeded on "noisy" nights, but all feasible and reasonable noise mitigation measures must be applied. What is reasonable is related to the predicted level of construction noise—there is an expectation in the community and by regulators that the higher the predicted noise level, the more reasonable it is to implement additional feasible noise mitigation measures. Performance is usually evaluated by monitoring. Monitored noise levels that exceed NMLs, or predicted levels, usually called "exceedances," can be considered to be non-compliances on "noisy" nights. This approach has some undesirable outcomes, such as encouraging overprediction. A better approach might be to require an evaluation into the reasons why a monitored level is greater than a predicted level, followed by the application of appropriate corrective action. A non-compliance would then only occur if this does not take place.

8:20

5aNSb2. What should be in a construction noise monitoring report? Laurence Clark (AAS, Unit 27, 43-53 Bridge Rd., Stanmore, New South Wales 2093, Australia, larry.clark@acousticstudio.com.au)

Consider a typical scenario taking place across NSW every day: construction on a major infrastructure project. The contractor has predicted the noise impact, monitors noise levels throughout the works, and must reports the results, as required by Regulators. But what should be included in reports of results? There is a general requirement for providing information that is meaningful to the public. But what might be meaningful information to the public is not specified. The reporter needs to use their judgment and act in good faith. Other guidance, including Standards, list records, sometimes extensive, that should be kept. Surprisingly, photographs, or even videos, do not seem to be mentioned. Nor does there seem to be mention of other useful information that can be collected efficiently with an electronic device, such as a mobile phone, which most people now carry. Screenshots of weather information, locations indicated on satellite imagery, distance measurements; videos of noise sources and meter displays; photos of instrument details, location, set up, and calibration can be efficiently recorded. Information that is meaningful to the public can be selected or derived from these records, with more detail for individual monitoring events and summaries for periodic reporting.

8:40

5aNSb3. A quantified risk-based approach to construction noise and vibration management. Dave J. Davis (Jacobs, Level 4, 12 Stewart Ave., Newcastle West, New South Wales 2302, Australia, dave.davis@jacobs.com)

The management of environmental noise and vibration impacts from construction activities can significantly influence the scheduling, program, activities, and costs of construction projects. While there are a number of mitigation and management measures available to the construction site to control their environmental impacts upon neighboring sensitive receivers, efforts to reduce impacts should be focused on taking the steps that will result in the greatest reduction to the overall project's environmental risk profile. In order to identify the countermeasures with the highest efficacy to reduce the overall risk, it is advantageous to quantify noise and vibration impacts based on ordinal metrics such as decibels and degree of annoyance. This paper describes a proposed method to quantify construction noise and vibration risk in order to improve the identification and selection of the countermeasures that should be implemented on site.

9:00

5aNSb4. Minimizing unavoidable construction noise impacts in urban environments using soundscape masking effects. Daniel Weston (EMM Consulting, 20 Chandos St., St Leonards, New South Wales 2065, Australia, dweston@emmconsulting.com.au) and Lance Jenkin (EMM Consulting, Newcastle, New South Wales, Australia)

Construction activities in urban environments often generate unavoidable noise impacts that can adversely affect the well-being of a surrounding community. Traditional noise mitigation strategies focus on reducing noise emissions, which may not be entirely feasible due to the nature of construction activities. It is often the case that construction activity needs to be undertaken during evening and night periods due to the need to occupy a road, a rail corridor, or to facilitate utility service provider requirements to work during low demand periods. This, coupled with the increased community sensitivity at night, is then exacerbated by diminishing background noise levels as daytime human activity dissipates. This paper presents an approach for minimizing the adverse effects of residual construction noise by leveraging the masking effects of artificially generated soundscapes to counteract this existing phenomenon. We identify possible sound and level combinations that effectively mask construction noise without creating additional disturbance. A case study compares the impacts of construction noise with and without the implementation of soundscapes to evaluate the effectiveness. The study explores the compatibility of integrating loudspeakers onto lighting-towers, which are already a mandatory requirement for evening and night construction works, to distribute the soundscape to surrounding noise sensitive receivers.

9:20–9:40 Break

9:40

5aNSb5. Comparison of predictions from web-based construction noise tool with operator-attended field measurements. Adrian Morris (Renzo Tonin & Assoc., 1/418 Elizabeth St., Surry Hills, New South Wales 2010, Australia, adrian.morris@renzotonin.com.au), Mattia Tabacchi, and Raihan Zhafranata (Renzo Tonin & Assoc., Surry Hills, New South Wales, Australia)

Web-based noise prediction tools are increasingly being used to assist infrastructure projects in Australia with managing construction noise impacts. Typically, these noise tools use a pre-calculated set of noise predictions from a detailed 3-D computer noise model, with the web app providing an interface for the user to select (manually or automatically) the most appropriate reference source and adjusting the result to align with the desired sound power level. Verification noise monitoring is often prescribed as a management measure, particularly when the noise predictions exceed the nominated target at one or more noise-sensitive receivers. These noise measurements are then used to determine if corrective actions are necessary. An analysis of predicted levels generated using Gatewave and subsequent verification noise measurements has been conducted, with consideration given to key parameters such as plant/equipment and measurement locations and equipment sound power levels. Identification of key causes for deviations between predicted and measured levels is also discussed.

10:00

5aNSb6. Modernization of the "Noise Database for Prediction of Noise on Construction and Open Sites." Jesse Tribby (EMM Consulting Pty Ltd, 1 Forbes St., Unit 42, Carrington, New South Wales 2294, Australia, jtribby@emmconsulting.com.au)

The "Noise Database for Prediction of Noise on Construction and Open Sites" was prepared by Hepworth Acoustics Ltd in 2004, distributed by the UK Department for Environment Food and Rural Affairs (DEFRA), and updated in 2005 and 2006. It has been adopted worldwide and provides a degree of standardization of assumptions used for certain projects. With access to thousands of results from sound power testing done in general accordance with ISO methodology, which uses multiple microphone positions to account for directional noise emissions, there is a unique opportunity in Australia to significantly improve the quality of the database. Improvements to the database would include a broader range of equipment and larger sample sizes allowing a better understanding of statistical uncertainty, options for reasonable and feasible attenuation, and low-frequency noise components that the DEFRA database is currently lacking. This could be used as a reference for both acousticians and regulators to improve the quality and consistency of assumptions used in the assessment of noise impact from a wider range of projects.

10:20

5aNSb7. Managing noise and vibration on mega projects in Australia. Tracy Gowen (Construction Team, Renzo Tonin and Assoc., Level 1, 418A Elizabeth St., Surry Hills, New South Wales 2010, Australia, Tracy.Gowen@renzotonin.com.au) and Mattia Tabacchi (Construction Team, Renzo Tonin and Assoc., Surry Hills, New South Wales, Australia)

The last decade has seen the emergence of the mega project in Australia, including Sydney Metro, WestConnex, Western Sydney Airport, Cross River Rail, Melbourne Metro, and North East Link Project. These projects provide challenges in the management of noise and vibration due to site constraints, proximity to sensitive receivers and condensed project timelines. While the environmental noise and vibration regulation of these projects can differ from state to state, the design process remains largely the same. This paper looks at noise and vibration design and management on some of Australia's biggest infrastructure projects. Design considerations include physical mitigation measures such as acoustics sheds, noise barriers, temporary noise barriers or enclosures, and ventilation mitigation. Management measures and development of ongoing management tools are described. Finally, this paper reflects upon the overall noise and vibration outcomes in terms of impacts on the affected communities compared with cost of mitigation.

10:40

5aNSb8. Comparison of predicted and measured noise levels from major infrastructure construction projects. Nicholas Henrys (Resonate Consultants, Level 4, 23 Peel St., Adelaide, South Australia 5000, Australia, nick.henrys@resonate-consultants.com), Andrew Parker (Resonate Consultants, Sydney, New South Wales, Australia), Tom Evans (Resonate Consultants, Melbourne, Victoria, Australia), and Darren Jurevicius (Resonate Consultants, Adelaide, South Australia, Australia)

Urbanization has led to major infrastructure projects increasingly being constructed within heavily populated residential areas. Prediction and management of noise impacts associated with construction works is an increasingly important consideration during both the planning and delivery phase of projects. However, accurate prediction of construction noise is often challenging, due to the dynamic nature of infrastructure construction sites, with a large number of mobile and intermittent noise sources. Particularly during the planning phase, there is typically a lack of detailed information required to support accurate noise modeling. Noise predictions are nevertheless relied upon to inform the project team and surrounding community of the expected noise impact of upcoming works and to determine the most appropriate mitigation and management measures to adopt. In this paper, we compare noise levels predicted using a variety of common methods, with attended and unattended noise monitoring results obtained on a number of recent projects.

11:00

5aNSb9. Application of audio spectrogram transformer machine learning model for audio tagging of construction activities. Ben Cooper-Woolley (SiteHive, Level 1, 2-12 Foveaux St., Surry Hills, New South Wales 2010, Australia, ben@sitehive.co) and Sipei Zhao (Botany Ctr. for Audio, Acoust. and Vib., Univ. of Technol. Sydney, New South Wales, Australia)

Major construction projects are approved based on an Environmental Impact Statement, which includes modeled predictions of noise impacts based on planned program. However, actual on site construction activities can differ significantly from planned works, resulting in modeled acoustic impacts (which have been used to mitigate impacts and inform stakeholders) out of date. A potential solution to this may be the use of machine learning models, to initially classify, and later predict, actual on site activities and commensurate impacts on nearby stakeholder and communities caused by site works. By leveraging emerging lower cost, smaller noise monitoring devices more data can be collected at receivers, and classified to determine the contributing sources of sound. SiteHive has worked with the University of Technology Sydney to design and develop a machine learning model to classify construction works in real-time on site, integrated as part of the SiteHive Hexanode multi-sensor environmental monitoring device. This presentation will showcase the design and development undertaken to date, and highlight results as tested as part of a major works program.

Session 5aPA

Physical Acoustics: General Topics in Physical Acoustics

Jun Kuroda, Cochair
Advanced Technology Research Institute, KYOCERA Corporation, 3-7-1 Minatomirai, Nishi-ku, Yokohama, 220-0012, Japan

Joel B. Lonzaga, Cochair
NASA Langley Research Center, 2 N. Dryden St. (MS 463), Hampton, VA 23681

Contributed Papers

8:00

5aPA1. Growth rates of harmonics in nonlinear vortex beams. Chirag A. Gokani (Appl. Res. Labs. and the Walker Dept. of Mech. Eng., Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, chiragokani@gmail.com), John M. Cormack (Div. of Cardiology, Dept. of Medicine, Univ. of Pittsburgh, Pittsburgh, PA), and Mark F. Hamilton (Appl. Res. Labs. and the Walker Dept. of Mech. Eng., Univ. of Texas at Austin, Austin, TX)

Nonlinear effects in acoustical vortex beams were described originally in a numerical investigation by Marchiano *et al*. [Phys. Rev. E **77**, 016605 (2008)]. The focus of the present investigation is the rate at which nonlinearly generated harmonics grow with distance in the field of a vortex beam radiated by a monofrequency or bifrequency source as a function of the topological charge. The motivation is a paper by Richard *et al*. [New J. Phys. **22**, 063021 (2020)] suggesting that the rate of nonlinear distortion in the direction of the beam axis is increased in a vortex beam due to the lengthened propagation distance along the ray paths spiraling about the beam axis. In the present work, harmonic growth rate is characterized in terms of both amplitudes along different ray paths and total power across the beam. Growth rates are calculated and compared for harmonics generated in nonlinear vortex beams radiated by sources with different topological charges and amplitude distributions. Both unfocused and focused beams are considered. The likelihood of shock formations is also explored. [CAG is supported by the ARL:UT Chester M. McKinney Graduate Fellowship in Acoustics.]

8:20

5aPA2. Phase holograms for the three-dimensional patterning of microparticles in a traveling wave field. Mohamed A. Ghanem (Appl. Phys. Lab., Univ. of Washington, Seattle, WA, mghanem@uw.edu), Adam D. Maxwell (Dept. of Urology, Univ. of Washington, Seattle, WA), Diane Dalecki (Dept. of Biomedical Eng., Univ. of Rochester, Rochester, NY), Oleg A. Sapozhnikov, and Michael R. Bailey (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Acoustic fields can generate radiation forces that can align particles in patterns. Forces from standing waves pattern particles in three dimensions (3-D) at either nodal or anti-nodal regions. These patterns can be utilized to form 3-D microstructures for applications in tissue engineering or fabrication of layered materials. However, standing waves require more than one transducer or a reflector, which can complicate implementation. Here, we developed a method to suspend and align microspheres using a traveling wave from a single transducer. Acoustic fields were shaped to align polyethylene microspheres mimicking tissue cells in parallel planes along the axis of the transducer. A Bessel-like beam was developed using diffraction theory and an iterative angular spectrum approach were used to design phase holograms to shape pressure fields. Gor'kov potential was used to calculate radiation forces while minimizing the axial forces to create a stable trap. The resulting pressure fields and particle patterns matched predictions with a similarity index >0.92, where 1 is a perfect match. The transverse radiation force was ten times each microsphere's weight and comparable to the standing wave radiation forces. Next steps are *in vivo* implementation of cell patterning for tissue engineering. [Work supported by NIH K25-DK132416 and P01-DK043881.]

8:40

5aPA3. Characterizing bulk-driven acoustic streaming in air. Christopher Stone (Dept. of Mech. Eng., Univ. of Bristol, Queens Bldg., University Walk, Bristol, City of Bristol BS81TR, United Kingdom, c.stone@bristol.ac.uk), Anthony Croxford, Mahdi Azarpeyvand, and Bruce Drinkwater (Dept. of Mech. Eng., Univ. of Bristol, Bristol, City of Bristol, United Kingdom)

The time-independent fluid flows induced by the attenuation of acoustic energy known as acoustic streaming have been experimentally measured in air using particle image velocimetry for two high powered ultrasonic transducers. The literature declares that these bulk-driven "Eckart" streaming flows are observed as turbulent jets in the direction of the acoustic wave propagation. The aim of this work is to increase the understanding of the coupling between the acoustic and fluid domains and to validate these well-established theoretical relationships. Langevin horns and focused arrays of parking sensor transducers, which operate at ∼26 and 40 kHz, respectively, and are both capable of producing sound pressure levels of up to 170 dB, were used to assess the relationship between the first order acoustic field and the second order streaming velocity field. Streaming velocities of the order of 0.2 and 0.3 m/s were measured for the Langevin horn and the focused array, respectively. Two-step COMSOL Multiphysics models were created for both transducers, first solving the acoustic fields in the frequency domain, then using the results to drive a streaming volume force in a stationary fluid mechanics study. Both laminar and turbulent k-ε fluid mechanics models were compared to the experimental results.

9:00

5aPA4. Focus control of a concave ultrasonic gel lens. Kokichi Tagashira (Doshisha Univ., 1-3 Tatara Miyakodani, Kyotanabe 610-0394, Japan, ctwh0362@mail4.doshisha.ac.jp), Yuki Harada, Kosuke Nakamura, and Daisuke Koyama (Doshisha Univ., Kyoto, Japan)

Optical image stabilization is a powerful tool for camera devices. This paper investigates a tunable lens using ultrasound and a transparent viscoelastic gel in which the focal point can be controlled in the radial direction. The lens has a simple structure and consists of four piezoelectric ultrasonic transducers, a glass disc, and a transparent silicone gel film. A four-phase drive of the lens at the resonant frequency generates the flexural traveling-wave mode, resulting in changes of the surface profile of the gel by acoustic radiation force. At a resonant frequency of 60.8 kHz, a concave shape was observed on the gel surface, indicating that a concave lens can be fabricated by controlling the driving frequency and the phase difference between the

transducers. Optical characteristics of the lens were evaluated by ray tracing. The deformation displacement of the lens surface and the change in focal length increased with the voltage amplitude. The voltage ratio between the four transducers was changed, allowing the center position of the gel deformation and its focal position to be shifted in the radial direction, thus realizing image stabilization functions.

9:20

5aPA5. Effect of ultrasonic vibration on the orientation of C2C12 cells. Ryohei Hashiguchi (Doshisha Univ., 1-3 Tatara Miyakodani, Kyotanabe 610-0394, Japan, ctwh0311@mail4.doshisha.ac.jp), Masahiro Kumeta (Kyoto Univ., Kyoto, Japan), and Daisuke Koyama (Doshisha Univ., Kyoto, Japan)

Control of orientation direction of tissues is important for regenerative medicine techniques. This paper investigates effects of ultrasound vibration on a culture dish on the orientation of myoblast C2C12 cells. A piezoelectric ultrasonic transducer (PZT) was fixed to the bottom of a culture dish. A continuous sinusoidal signal with 10 to 30 V_{pp} at the resonant frequency of 86 kHz was input to the transducer to generate a concentric flexural vibration mode on the bottom surface of the dish. C2C12 cells were cultured on the dish for 2 days under continuous ultrasonic vibration. The ultrasound excitation was then stopped, resulting in the differentiation of the cells into myotube cells through 7 days. The orientation distribution of the cells on the dish was evaluated by spatial Fourier transform in the microscopic images. Compared with the control, the orientation of C2C12 cells was changed significantly under ultrasonication; the concentric pattern of the cell orientation appeared, which corresponds to the concentric vibration mode generated on the bottom of the dish. Larger vibrational amplitude of the dish gave a larger effect on the cell orientation.

9:40

5aPA6. Noncontact rotation of a small object using an asymmetric ultrasonic field. Eimei Yamamoto (Sci. and Eng., Doshisha Univ., 1-3 Tatara Miyakodani, Kyotanabe-shi, Kyoto-fu 610-0394, Japan, ctwh0374@mail4.doshisha.ac.jp) and Daisuke Koyama (Sci. and Eng., Doshisha Univ., Kyoto, Japan)

Noncontact manipulation techniques using airborne high-intensity ultrasound can be applied in various industrial fields such as pharmaceutical industry and future space industry. This paper discusses a method to rotate a small object in the air without physical contact using ultrasound. The experimental system consists of a vibrating disc with four bolt-clamped Langevin-type ultrasound transducers and two semicircular reflectors. The flexural vibration of the disc generates an acoustic standing wave between them, and a small object can be levitated at the nodal position. By inclining one of the two reflectors, an asymmetric acoustic field and an acoustic traveling wave in the circumferential direction are generated where the object can be rotated in the clockwise or counterclockwise direction. Vector flows of the acoustic intensity in the air between the vibrator and reflectors were calculated. When the tilt angle of the reflector was 0.32°, the object was rotated without contact, and the rotation direction corresponded with the direction of the vector flows of the acoustic intensity. A greater input current gave a greater rotation speed; the maximum rotation speed of the object was 6.84 rps in the case of 1.14 A.

10:00–10:20 Break

10:20

5aPA7. Measuring excess attenuation in littoral environments: Evolution of a measurement system. Andrea Vecchiotti (Eng., East Carolina Univ., Wanchese, NC), Diego Turo, Joseph Vignola (Mech. Eng., The Catholic Univ. of America, Washington, DC), Matthew D. Stengrim, Jeff Foeller, Maranda Byrd, Kian Nguyen (Eng., East Carolina Univ., Greenville, NC), and Teresa J. Ryan (Eng., East Carolina Univ., 1000 East Tenth St., Greenville, NC 27858, ryante@ecu.edu)

Atmospheric sound propagation measurements over long distances are generally lacking in the literature. This work presents the evolution of the system deployed to collect high spatial and temporal resolution synchronized acoustic and meteorological data. Specifically, the mature configuration includes: a long-range acoustic device capable of source levels up to 140 dB; up to four 7-channel acoustic arrays; two 7-m temperature profiling masts; and a scanning Doppler LIDAR system for volumetric wind profiling at ranges up to several kilometers. This work presents lessons learned over the various iterations of the system such as the design and evolution of the excitation signal used; physical configuration of the components of the system; logistics and planning concerns; post-processing challenges; and quality control considerations.

10:40

5aPA8. High spatial resolution long-term temperature profiling to inform near-shore atmospheric sound propagation. Matthew D. Stengrim (Eng., East Carolina Univ., E 5th St., Greenville, NC 27858, stengrimm19@students.ecu.edu), Andrea Vecchiotti (Eng., East Carolina Univ., Wanchese, NC), Jeff Foeller, Kian Nguyen, Maranda Byrd (Eng., East Carolina Univ., Greenville, NC), Diego Turo, Joseph Vignola (Mech. Eng., The Catholic Univ. of America, Washington, DC), and Teresa J. Ryan (Eng., East Carolina Univ., Greenville, NC)

Atmospheric sound propagation depends on factors such as surface characteristics and meteorological parameters. Effective numerical modeling of atmospheric sound propagation requires reliable and realistic input parameters. One of the most influential factors is the air temperature profile. This work examines the application of experimentally determined air temperature profiles in acoustic modeling of littoral environments. Long term measurements of near-surface air temperature profiles over water and marsh grass are presented. Data have been recorded using temperature loggers affixed to two permanent 7 m masts. The loggers were positioned with a 1 m vertical spacing. A common approach to characterize the near-surface boundary layer uses Monin–Obukhov Similarity Theory (MOST). This work compares high spatial resolution air temperature measurements with measurement-informed profiles based on MOST. The development of a more experimentally grounded set of assumptions will ultimately contribute to the improvement of atmospheric acoustic transmission loss models in near shore environments.

11:00

5aPA9. Influence of mesoscale meteorological observations on near shore excess attenuation prediction. Andrea Vecchiotti (Eng., East Carolina Univ., Wanchese, NC), Teresa J. Ryan (Eng., East Carolina Univ., 1000 East Tenth St., Greenville, NC 27858, ryante@ecu.edu), Joseph Vignola (Mech. Eng., The Catholic Univ. of America, Washington, DC), Matthew D. Stengrim, Jeff Foeller (Eng., East Carolina Univ., Greenville, NC), and Diego Turo (Mech. Eng., The Catholic Univ. of America, Washington, DC)

This work focuses on numerical modeling of atmospheric sound propagation in a near-shore environment that is informed by concurrent meteorological observations. The atmospheric acoustic measurement layout places the source approximately 500 m from shore with one acoustic receiver array at the shoreline and a second acoustic receiver array approximately 350 m inland. The space between the shoreline and the inland receiver array consists of highly uniform marsh grass vegetation. The concurrent meteorological observations include high resolution temperature profiling at two locations (one over land and one over water) along with scanning Doppler LIDAR wind profiling measurements in the source-to-receiver direction. These meteorological measurements inform the input parameters used in a parabolic equation solver. The numerical predictions are compared to the measured transmission loss values to evaluate model sensitivity to input parameters.

11:20

5aPA10. Development of an 128 element array for demonstrating Artificial Intelligence techniques in an Acoustic Sensor System. Wayne E. Prather (NCPA, Univ. of MS, P.O. Box 1848, 145 Hill Dr., University, MS 38677-1848, wayne@olemiss.edu), William G. Frazier (NCPA, Univ. of MS, University, MS), and Charles Gilliland (Hyperion Technol. Group, Tupelo, MS)

An 128 element acoustic array utilizing digital MEMS microphones was developed for demonstrating Artificial Intelligence techniques in an

Acoustic Sensor System. The data from all 128 elements are collected and transmitted to the host computer through USB at a bit depth of 24 bit and sample rate of 48 kHz. Under the assumption of a condition of limited processing power, Artificial Intelligence techniques are used to determine the optimum choice of which set of 24 of the available 128 elements to process for determining the direction of a detected source of sound. The design and construction of the array will be discussed. A companion presentation will discuss the signal processing techniques developed and employed on the system.

11:40

5aPA11. Tandem cylinder aeroacoustic sources. Rowena Dixon (The School of Mech. and Manufacturing Eng., UNSW, High St., Kensington, New South Wales 2052, Australia, z8612897@ad.unsw.edu.au), Chaoyang Jiang, Charitha de Silva, Danielle Moreau, and Con Doolan (The School of Mech. and Manufacturing Eng., UNSW, Sydney, New South Wales, Australia)

Tandem cylinders in turbulent flow are found in aircraft landing gear, chimney stacks, power lines, and bridge piers. The aim of this paper is to present detailed unsteady surface pressure measurements for tandem cylinders. These unsteady surface pressures are the major sources of noise in this important fundamental test case. A series of experiments were conducted in the UNSW anechoic wind tunnel to investigate the sound generation and flow distortion from a downstream cylinder interacting with wake from an upstream cylinder. This wake is highly anisotropic, containing large vortex shedding scales as well as energetic broadband components. The experiments were conducted using PIV (Particle Image Velocimetry) to obtain instantaneous flow fields and remote microphones to capture unsteady surface pressure measurements. Microphones were positioned outside the flow to record far-field noise. The cylinder was rotated to obtain surface pressure around the circumference. The acoustic results for the tandem cylinder case were compared with noise generation from a single cylinder in uniform flow. An increase in both surface pressure and far-field noise was observed in the tonal and broadband components. It was concluded that interaction with the wake of the upstream cylinder significantly increased the unsteady surface pressure of the downstream cylinder.

Session 5aPP

Psychological and Physiological Acoustics: Top-Down and Bottom-Up Processing in Individuals with Normal Hearing and Hearing Difficulties

Sriram Boothalingam, Cochair
Communication Sciences and Disorders, University of Wisconsin-Madison, 1975 Willow Drive, Goodnight Hall, Rm 482, Madison, WI 53706

Aravind Parthasarat, Cochair
Communication Sciences and Disorders, Univ. of Pittsburgh, Pittsburgh, PA 15260

Viji Easwar, Cochair
National Acoustic Laboratories, 16 University Avenue, Sydney 2109, Australia

Chair's Introduction—7:55

Invited Papers

8:00

5aPP1. Subcortical measures of multi-talker speech processing. Melissa Polonenko (Speech-Language-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr SE, SLHS, Rm. 115 Shevlin Hall, Minneapolis, MN 55455, mpolonen@umn.edu), Mishaela DiNino (Dept. of Communicative Disord. and Sci., Univ. of Buffalo, Buffalo, NY), and Ross Maddox (Biomedical Eng., Neurosci., Univ. of Rochester, Rochester, NY)

Communication in everyday acoustic environments is facilitated by the ability to locate, separate, and process speech from different locations. We do this by combining information received by both ears. Typically, the auditory system does not simply relay information delivered by both ears; rather, bilateral information is combined through elaborate binaural pathways in both the brainstem and cortex to allow us to communicate in complex acoustic environments. But traditionally it has been difficult to evaluate the subcortical underpinnings of speech communication using more complex, naturalistic continuous speech stimuli. Recently, the "peaky" speech method was created that uses stories to evoke brainstem responses using EEG. We have recorded brainstem responses from adults with normal hearing who listen to one or multiple narrators, and narrators coming from one or different perceived spatial locations. This talk will highlight the brainstem work undertaken towards creating subcortical measures of speech processing and spatial hearing.

8:20

5aPP2. Electrophysiological and behavioral assessment of binaural processing: Effects of age. John Grose (Dept. Otolaryngol. - Head & Neck Surgery, Univ. of North Carolina at Chapel Hill, 170 Manning Dr. CB#7070, Chapel Hill, NC 27599, john_grose@med.unc.edu), Emily Buss, Stacey G. Kane, and Monica Folkerts (Otolaryngol. - Head & Neck Surgery, Univ. of North Carolina at Chapel Hill, Chapel Hill, NC)

Spatial hearing deteriorates with age, but the basis(es) of this age-dependency is not clear and could reflect both bottom-up and top-down factors. Since binaural processing depends in part on interaural time difference cues, this study assessed binaural temporal acuity using both objective and behavioral measures. Three tasks incorporating interaural time/phase differences were tested: (1) in-phase and out-of-phase frequency modulation (FM) detection as a function of FM rate; (2) upper frequency limit of in-phase versus out-of-phase tone discrimination; and (3) minimum audible angle for low frequency tones. Each task incorporated parallel behavioral and electrophysiological measures: For behavior, forced-choice adaptive methods were employed; for electrophysiology, the acoustic change complex (ACC) was recorded. Additional measures of peripheral auditory function, as well as cognitive status, were also collected. Young and middle-aged adults (n = 20 per age group) with normal audiometric hearing were tested. Differences in behavioral measures of binaural performance were noted between the two age groups and these were reflected to some extent in the ACC results. However, group differences were also noted in terms of cochlear function and in terms of cognitive abilities. This pattern of results suggests that age-related deterioration in spatial perception is likely to be multi-faceted.

8:40

5aPP3. Temporal dynamics of the medial olivocochlear reflex in humans. Srikanta k. Mishra (The Univ. of Texas at Austin, 2504A Whitis Ave., Austin, TX 78712, srikanta.mishra@austin.utexas.edu)

Ipsilateral and contralateral sounds can activate the medial olivocochlear reflex. The time course of medial efferent effects could contribute to the binaural unmasking of sounds. Computational modeling suggests that implementing efferent time constants improves speech-in-noise recognition. In addition, the time course could also support the protective function offered by the medial efferents. However, the current knowledge of efferent time constants is limited to a few probe frequencies (e.g., 1000 Hz) in limited subjects. The time course of medial efferent activation might be different for other probe frequencies or elicitors. In this study, the time course of the medial olivocochlear reflex was measured using stimulus frequency otoacoustic emissions recorded at spectral peaks near 1000 and 4000 Hz with broadband elicitors in 17 normal-hearing adults. Several preliminary but important findings emerged: (1) The onset response could be fitted with fast, medium, and/or slow time constants for limited subjects; (2) The magnitude (30–40%) of the efferent effect was similar to those reported in the literature; (3) The overall decay time appears to be longer for 4000 Hz relative to 1000 Hz; and (4) There was large individual variability in all response parameters across subjects. These findings will be discussed in detail. [Work supported by NIH/NIDCD R01DC018046.]

9:00

5aPP4. Serotonin excites medial olivocochlear neurons of the auditory efferent pathway. Kirupa Suthakar (Section on Neuronal Circuitry, NIH/NIDCD, 35A Convent Dr., 3D804C, Bethesda, MD 20892, kirupa.suthakar@nih.gov) and Catherine Weisz (Section on Neuronal Circuitry, NIH/NIDCD, Bethesda, MD)

Medial olivocochlear (MOC) efferent neurons are located in the superior olive of the brainstem and form a sound evoked feedback loop that inhibits cochlear amplification via suppression of outer hair cell (OHC) electromotility. MOC neurons putatively receive a diverse range of synaptic input from various auditory and non-auditory brain regions. Given the proposed role of these neurons in context dependent tasks such as selective attention, we are interested in investigating the non-auditory modulation of MOC activity by the neurotransmitter serotonin (5-hydroxytryptamine, 5-HT). We used the ChAT-IRES-Cre;tdTomato mouse model to identify cholinergic MOC neurons in the brainstem. Immunohistochemical data have validated serotonergic terminals in apposition to both retrogradely labeled and genetically identified MOC neurons in mouse. During patch-clamp recordings from MOC neurons, exogenous application of serotonin increased neuron excitability by increasing action potential (AP) firing rate and decreasing both rheobase and AP threshold. Additionally, serotonin reduced the stimulation required to evoke a given AP firing rate in MOC neurons. These data will aid in our understanding of central auditory processing and how factors such as mood and attention are involved in modulating MOC responses in complex listening situations such as in the presence of background noise.

9:20

5aPP5. Olivocochlear efferent system degeneration and plasticity with aging and noise exposure. Amanda Lauer (Johns Hopkins Univ. SOM, 521 Traylor, 720 Rutland Ave., Baltimore, MD 21205, alauer2@jhmi.edu)

Contributions of the ascending peripheral and central auditory pathways to age-related and noise-induced hearing deficits have been well-characterized, but we know comparatively little about how the descending projections from the brain to the cochlea contribute to hearing deficits. Studies have shown dynamic structural and neurochemical changes in the olivocochlear system associated with acoustic experience and age, indicating that olivocochlear efferents cannot be considered a static system when attempting to reveal effects on hearing. I will summarize our efforts to understand how these pathways are involved in hearing dysfunction measured using physiological, behavioral, and anatomical assays in mouse models. Our work suggests that loss of olivocochlear efferent neurons is most detrimental to hearing in aged and aged, noise-exposed auditory systems which are already working with reduced afferent input to the brain. Furthermore, noise-induced changes in the lateral olivocochlear efferents appear to compensate for diminished medial olivocochlear activity—an effect that may explain the variable behavioral outcomes in hearing in noise studies involving the medial olivocochlear reflex in humans and animal models.

9:40–10:00 Break

Contributed Paper

10:00

5aPP6. Perceptual and neural measures of spectral and temporal cue strength in speech perception. Jitpakorn Pichaitanaporn (Dept. of Commun. Sci. and Disord., Mahidol Univ., Bangkok, Thailand), David A. Eddins (School of Commun. Sci. and Disord., Univ. of Central Florida, Orlando, FL), Erol J. Ozmeral, Nathan C. Higgins, Carrie Secor (Commun. Sci. and Disord., Univ. of South Florida, Tampa, FL), and Ann C. Eddins (School of Commun. Sci. and Disord., Univ. of Central Florida, 4364 Scorpius St., HSII, Ste. 101, Orlando, FL 32816, ann.eddins@ucf.edu)

Many listeners experience difficulty understanding speech in noise that may result from limited ability to use important acoustic cues, such as spectral and temporal features. One way of assessing deficits related to perceptual or neural processing is to enhance or degrade specific cues and measure the impact on behavior or neural coding. To do so, we combined a target-word identification task with simultaneous EEG measurement in 20 young and 20 older normal-hearing listeners. Coordinate Response Measure sentences were manipulated in the spectral modulation (SM) and temporal modulation (TM) domains (neutral, degraded, enhanced) and presented in four-talker babble in both passive and attention conditions. Multivariate temporal response function (mTRF) analyses were used to evaluate cortical tracking of the speech envelope. Speech perception in younger listeners improved for both SM and TM enhancement, but older listeners only benefited from TM enhancement. Both groups showed stronger cortical tracking for attention versus passive listening, but only older listeners showed tracking changes across SM and TM conditions. Cortical tracking and perceptual results in older listeners are consistent with cognitive aging theories suggesting that cortical processing changes with age involve both compensation and dedifferentiation to help maintain auditory perceptual performance.

10:20

5aPP7. Effects of age-related cochlear deafferentation and central gain on auditory scene analysis. Hari Bharadwaj (Commun. Sci. and Disord., Univ. of Pittsburgh, 3600 Atwood St., Forbes Tower 5063, Pittsburgh, PA 15213, hari.bharadwaj@pitt.edu) and Varsha Mysore Athreya (Speech, Lang., and Hearing Sci., Purdue Univ., West Lafayette, IN)

Animal models show that cochlear afferent nerve endings are more vulnerable than sensory hair cells to age-related damage. Because such cochlear deafferentation is not apparent in standard audiometry, the extent to which it contributes to deficits in human hearing is debated, and the intervening neural processes are poorly characterized. This presentation will describe our efforts to address these gaps through co-ordinated experiments in at-risk humans and a chinchilla model. Our results suggest that cochlear deafferentation is widespread in middle age despite clinically normal audiometric sensitivity. Furthermore, the resulting reduction in peripheral input appears to trigger compensatory central "gain" at the cortical level, likely through altered local balance of excitation and inhibition. Consistent with the important role of inhibition in parsing temporal regularities across different frequency components of sound, central gain is also associated with reduced ability to perceptually segregate acoustic scenes with multiple sources into individual perceptual streams. Taken together, our results suggest that age-related cochlear deafferentation may affect hearing not only by reducing the fidelity of input encoding but also by interfering with the central auditory system's ability to extract targets in noisy environments.

10:40

5aPP8. Dissociated neural tracking as an index of speech understanding in background noise. David Meng (National Acoust. Labs., 16 University Ave., Macquarie Univ., NSW 2109, Sydney, Australia, david.meng@nal.gov.au), Angela Wong (National Acoust. Labs., Sydney, New South Wales, Australia), Jennifer Clemesha (Cochlear Ltd., Sydney, New South Wales, Australia), and Jorge Mejia (National Acoust. Labs., Sydney, New South Wales, Australia)

Human brain activity has been shown to track hierarchical linguistic units embedded in connected speech and these responses can be directly modulated by changes in speech intelligibility caused by spectral degradation or prior knowledge. In this study, we introduce a background noise and manipulate its level relative to the target speech to test the hypothesis that the tracking responses are modulated differently by the variations in Signal-to-Noise Ratio (SNR) and changes in speech intelligibility. Electroencephalography (EEG) responses to speech in quiet and in noise (multi-talker babble) at three different levels were measured from 19 normal hearing participants. Driven by the reduction in intelligibility, cortical coherence to "abstract" linguistic units with no accompanying acoustic cues was reduced relative to the "speech only" condition, and lateralized to single cerebral hemispheres. In contrast, brain responses coherent to words, aligned with acoustic onsets, were bilateral and reduced systematically as noise level increased. Strength of the tracking response correlated with subjective ratings from each participant on how much they can understand the speech sentences at all different linguistic levels. These results provide an objective and sensitive neural marker of speech intelligibility which can be further developed into clinical applications for objective assessment of speech-in-noise understanding.

11:00

5aPP9. Auditory cortical coding of task-relevant variables during active listening. Nathan A. Schenider (Ctr. for Neurosci., Univ. of Pittsburgh, Pittsburgh, PA), Michael Malina (Neurosci. Inst., Carnegie-Mellon Univ., Pittsburgh, PA), Rebecca F. Krall (Otolaryngol., Univ. of Pittsburgh, Pittsburgh, PA), and Ross S. Williamson (Otolaryngol., Univ. of Pittsburgh, W1448 Biomedical Sci. Tower, 200 Lothrop St., Pittsburgh, PA 15213, rsw@pitt.edu)

Auditory-guided behaviors are ubiquitous in everyday life, whenever auditory information is used to guide our decisions and actions. The primary route for auditory information to propagate from the ACtx is through intratelencephalic (IT) and extratelencephalic (ET) neurons in layer (L) 5. To investigate the behavioral role of IT and ET neurons, we engaged mice in a head-fixed auditory categorization task and longitudinally monitored populations of L5 IT and ET neurons using two-photon calcium imaging. Clustering analyses of these populations revealed heterogenous response motifs that correlated with various stimulus and task variables, but with distinct expression patterns across learning and task proficiency. Our results suggest a tradeoff of information between two distinct populations within L5, with IT projections playing a role in initial task acquisition, while ET projections are recruited and reinforced throughout the learning process. This underscores the differential roles of distinct descending systems and contribute to our understanding of how auditory information is processed and utilized to guide decision-making and action.

11:20

5aPP10. Modulation of auditory processing by non-auditory brain areas: Effects of acoustic trauma. Jack W. Zimdahl (School of Human Sci., Univ. of Western Australia, Crawley, Western Australia, Australia), Kristin M. Barry, Jennifer Rodger, Chenae de Vis, Donald Robertson (School of Human Sci., Univ. of Western Australia, Crawley, Western Australia, Australia), and Wilhelmina Mulders (School of Human Sci., Univ. of Western Australia, University of Western Australia, 35 Stirling Hwy., Crawley, Western Australia 6009, Australia, helmy.mulders@uwa.edu.au)

Acoustic trauma (AT) induced hearing loss elicits plasticity in the form of altered spontaneous firing throughout the central auditory pathway, including at the level of the medial geniculate nucleus (MGN). Non-auditory areas such as the prefrontal cortex (PFC) and amygdala are thought to modulate auditory processing in the MGN, which may play a role in sensory gating. In our laboratory, we investigated the functionality of inputs from the PFC and amygdala on MGN activity in different animal models (rat and guinea pig) both without and with AT. The effects of AT on auditory thresholds were investigated using auditory brainstem response (ABR) recordings or compound action potential recordings of the auditory nerve. To test circuitry functionality, we used single neuron recordings in the MGN combined with electrical stimulation of the PFC or amygdala. Our data showed, first, that electrical stimulation of either the PFC or amygdala resulted in a variety of effects within the MGN (facilitation, inhibition, or no effect) and, second, that AT and subsequent hearing loss significantly increased the magnitude of inhibition. This change in inhibition may represent a compensatory mechanism in response to the increased spontaneous activity in the central auditory pathway known to occur following hearing loss.

Session 5aSCa

Speech Communication: Phonetics of Emerging Varieties of English

Felicity Cox, Cochair
Linguistics, Macquarie University, Balaclava Road, North Ryde 2109, Australia

Benjamin V. Tucker, Cochair
Communication Sciences and Disorders, Northern Arizona University, 208 E. Pine Knoll Dr. PO Box: 15045, Flagstaff, AZ 86011

Invited Papers

8:40

5aSCa1. Individual profiles amidst a multiethnolect: Acoustic hetereogeneity in London English. Paul Kerswill (Dept. of Lang. and Linguistic Sci., Univ. of York, York, United Kingdom), Andy Gibson (Dept. of Linguist, Queen Mary, Univ. of London, London, United Kingdom, andy.gibson@qmul.ac.uk), Devyani Sharma, Kathleen McCarthy, Elisa Passoni (Dept. of Linguist, Queen Mary, Univ. of London, London, United Kingdom), and Sam Hellmuth (Dept. of Lang. and Linguistic Sci., Univ. of York, York, United Kingdom)

Multicultural London English (MLE; Kerswill and Torgersen 2008; Cheshire *et al.*, 2011) arose in working class areas of London around 30 years ago through intensive, multiethnic social contact (Kerswill and Torgersen, 2021). The variety's highly systematic phonological system has been seen as displacing traditional London vernacular varieties including Cockney, which have moved further East (Fox, 2015). MLE and Standard Southern British English are two strands among many, interwoven with features of the earlier London vernacular and features of non-MLE varieties (Sharma and Sankaran, 2011). Using preliminary data from a new project, *Generations of London English*, we present an acoustic analysis that shows both the stability and emerging maturity of MLE as well as strands of continuity from the past. The GOAT vowel shows a diverse range of variants indexing social class, ethnicity, age, and gender. By contrast, all Londoners participate in GOOSE-fronting, which is thus primarily an index of age, and women lead FOOT-fronting, a city-wide change showing less ethnic and social class sensitivity. A generalized use of labels such as MLE risks mischaracterizing the sociophonetic reality. We argue for analysis of individual phonetic profiles as unique intersections of historically layered features, diffusing differently through a heterogeneous population.

9:00

5aSCa2. Abstract withdrawn.

Contributed Paper

9:20

5aSCa3. Variation in pre-nasal raising of TRAP in Australian English. Joshua Penney (Linguist, Macquarie Univ., Macquarie Univetrsity, New South Wales, Australia), Felicity Cox (Linguist, Macquarie Univ.16 University Ave., Macquarie Univ., New South Wales, Australia, felicity.cox@mq.edu.au), and Sallyanne Palethorpe (Linguist, Macquarie Univ., Macquarie Univ., New South Wales, Australia)

Phonetically raised and fronted realizations of pre-nasal TRAP are increasingly common in Australian English with previous research suggesting that greater raising may be associated with speakers from monolingual English-speaking backgrounds. Here, we present the results of an acoustic examination of pre-nasal TRAP raising (which we refer to as HAND) in a corpus of speech recordings from 183 adolescent speakers (aged 15–18) from five areas of Sydney that differ according to levels of linguistic diversity and the major heritage languages spoken (English-only, Arabic, Chinese, Indian, and Vietnamese). We extracted 1971 items containing the HAND vowel, along with 1807 items of pre-obstruent TRAP and 540 items of pre-obstruent DRESS. We calculated the degree of raising in HAND relative to pre-obstruent TRAP and DRESS through Euclidean Distance. The results show increased HAND raising in female speakers, although less raising is seen in the area with the highest proportion of Vietnamese language background speakers. Among male speakers the greatest incidence of raising is seen in the area with the lowest level of linguistic diversity (i.e., highest level of English-speaking background). These results suggest that HAND raising is associated with English-speaking backgrounds, but also that this is a change led by females.

Invited Paper

9:40

5aSCa4. Creaky voice prevalence across Sydney. Hannah White (Dept. of Linguist, Macquarie Univ., Balaclava Rd., Sydney, New South Wales 2109, Australia, hannah.white@mq.edu.au), Joshua Penney (Dept. of Linguist, Macquarie Univ., Macquarie Univ., New South Wales, Australia), Andy Gibson (Queen Mary Univ., London, United Kingdom), Anita Szakay, and Felicity Cox (Dept. of Linguist, Macquarie Univ., Sydney, New South Wales, Australia)

Creaky voice is a voice quality that has been associated with various social categories such as gender, age, and socioeconomic status in previous literature. However, few studies have investigated creaky voice in relation to linguistic or ethnic heritage. Using a corpus of conversational speech from 131 Australian English-speaking teenagers who live in various areas across Sydney characterized by differences in linguistic and ethnic diversity, we explore how creaky voice use is influenced by gender and ethnic heritage. Creaky voice is automatically detected using the optimized Union method (White *et al*., 2022, JASA), which employs a combination of acoustic cues to identify the various phonetic realizations of creaky voice. Findings show that creaky voice use is highly variable across Sydney indicating that a complex relationship exists between creaky voice prevalence, speaker gender and ethnic heritage. This study shows that while variability may correlate with social groupings, other factors, such as a speaker's orientation towards their community, may contribute to levels of creaky voice prevalence.

5a FRI. AM

Session 5aSCb

Speech Communication: Voice Therapy: Science and Clinical Efficacy I

Zhaoyan Zhang, Cochair
University of California, Los Angeles, 1000 Veteran Avenue, Suite 31-11, Los Angeles, CA 90095

Michael Dollinger, Cochair
ENT, University Hospital Erlangen, Waldstrasse 1, Erlangen, 91054, Germany

Catherine Madill, Cochair
Speech Pathology, University of Sydney, Susan Wakil Health Building, D18 Western Avenue, Camperdown 2006, Australia

Invited Paper

10:20

5aSCb1. Objective tools in pediatric voice therapy: A path to improved outcomes. Rita R. Patel (Dept. of Speech, Lang. and Hearing Sci., Indiana Univ., 2631 E Discovery Parkway, Bloomington, IN 47408, patelrir@iu.edu)

This is an invited presentation in the Special Session titled, "Voice therapy: objective measurement science and clinical efficacy." Pediatric dysphonia (hoarseness) is a common condition with prevalence estimates ranging from 1.4% to 23.9%. Dysphonia can be detrimental to children both psychologically and academically; hence, early identification and restoration of optimal vocal health is critical. Direct visualization of vocal fold vibratory patterns is fundamental for appropriate diagnosis and treatment of vocal fold pathology. Thus, evaluation of vocal fold structure and cycle-to-cycle vibratory motion through techniques of direct visualization of the vocal fold is essential for clinical assessment and measurement of treatment outcomes. This session will discuss the most recent advances in objective voice assessment and measurement of voice therapy outcomes in children. The focus will be on discussing an innovative approach that leverages high-speed videoendoscopy and a cutting-edge custom developed laser projection system for quantitative evaluation of lesion size and comprehensive evaluation of opening and closing phase dynamics of the glottal cycle before and after voice therapy in children with vocal fold nodules.

Contributed Papers

10:40

5aSCb2. Formant change and individual variation in male-to-female transgender speakers during voice therapy. Tuende Szalay (Speech Pathol., Univ. of Sydney, Susan Wakil Health Bldg., D18 Western Ave., Camperdown, New South Wales 2006, Australia, tuende.szalay@sydney.edu.au), Duy D. Nguyen (Speech Pathol., Univ. of Sydney, Sydney, New South Wales, Australia), Antonia Chacon, and Catherine Madill (Speech Pathol., Univ. of Sydney, Camperdown, New South Wales, Australia)

Increased fundamental frequency (F0), resonance frequency (F2), and vowel formant frequency (F1, F2, F3) are linked to be perceived as feminine, making them desired outcomes of successful gender-affirmative voice therapy for male-to-female transwomen (MTF). However, treatment often only targets pitch, reporting F0 as an acoustic outcome measure. Vowel formants are measured less often and show inconsistent results. We retrospectively explored the effect of voice therapy on formants of Australian English /i:/. 44 MTF speakers were treated with Sob Voice Therapy using the Optimal Phonation Task, Sob Voice Quality, and Sob Variant therapy components in consecutive order. Speakers were recorded at baseline and after mastering each therapy component. F1, F2, and F3 were extracted at /i:/ target in maximally three words per component. Formant values were analyzed using linear mixed models with therapy component and therapy length as fixed effects and patient as random effect. We found no statistically significant effect of therapy component nor session on formant values at $p = 0.05$. Graphic inspection of the data suggests between-speaker variation, indicating that some patients may respond to treatment despite the lack of group effect. Future research may explore other acoustic outcome measures for the efficacy of gender-affirmative voice therapy.

11:00

5aSCb3. Improving auditory-perceptual evaluation of disordered voice quality using a novel three-dimensional matching task. Yeonggwang Park (School of Commun. Sci. and Disord., Univ. of Central Florida, Orlando, FL), Supraja Anand (Dept. of Commun. Sci. & Disord., Univ. of South Florida, Tampa, FL), Shaheen N. Awan (School of Commun. Sci. and Disord., Univ. of Central Florida, Orlando, FL), Rahul Shrivastav (Office of the Provost and Executive Vice President, Indiana Univ. Bloomington, Bloomington, IN), and David A. Eddins (School of Commun. Sci. and Disord., Univ. of Central Florida, Orlando, FL, David.eddins@ucf.edu)

Current methods of auditory-perceptual evaluation of disordered voice quality using ordinal or interval scales have limited reliability and precision to quantify the magnitude of change in response to disorder progression or treatment. Matching tasks with synthetic comparison sounds have emerged as a more effective alternative, reducing biases and providing adjustable comparative values to represent perceived magnitude of change in quality. Previous studies have focused on the individual dimensions of breathiness, roughness, and strain separately and indicate covarying voice qualities may impact listener judgments. Here we investigate a three-dimensional matching (3-DMA) task where three variables, each representing a major quality

dimension, can be adjusted concurrently. The comparison sound was a low-pass filtered sawtooth waveform ($f_o = 151$ Hz) mixed with noise. Independent variables for matching were the signal-to-noise ratio for breathiness, amplitude modulation depth for roughness, and bandpass filter gain for strain. Listeners used sliders to adjust each variable, matching all three dimensions of 26 natural voice samples, selected to contain wide variation in all three dimensions. Preliminary results from five listeners demonstrated good reliability, similar to that observed in previous single-variable matching tasks for individual voice qualities. The results support the feasibility of the 3-DMA task. [Work supported by NIH R01DC009029.]

11:20

5aSCb4. Normalization of speech kinematic data for characterizing primary muscle tension dysphonia. Riya Shipurkar (School of Behavioral and Brain Sci., The Univ. of Texas at Dallas, 8227 Ranchview Dr., Irving, TX 75063, riyashipurkar14@gmail.com), Mackenzie White, and William Katz (School of Behavioral and Brain Sci., The Univ. of Texas at Dallas, Dallas, TX)

Primary muscle tension dysphonia (pMTD) is a voice disorder of unknown etiology in which people have reduced volume and become easily fatigued. The tongue has biomechanical linkage to the laryngeal motor system that affects the voice, suggesting that tongue movement variability might be a marker for this disorder. Previous studies of healthy/disordered speech have reduced individual talker differences in physical vocal tract characteristics via normalization procedures. Here, we obtained tongue movement data for diadochokinetic sequences produced by healthy adult talkers and people with pMTD. We used an electromagnetic articulography system to test three healthy adult talkers and a participant with pMTD. We recorded vertical displacement of the tongue dorsum for sustained /a/ and 30 sec of rapidly repeated /pataka/. Displacement (mm) was scaled relative to each talker's sustained vowel production, while movement time (msec) was normalized by dividing each /ka/ portion by the corresponding /pataka/ utterance length. The results suggest (1) normalization effectively reduced talker variability in tongue movement displacement and timing, and (2) the talker with pMTD showed markedly higher temporal variability, compared to the control talkers. We are currently testing this putative group difference using additional participants.

Invited Paper

11:40

5aSCb5. How does the choice of acoustic measures influence evaluation of voice therapy treatment outcomes? Catherine Madill (Speech Pathol., Univ. of Sydney, Susan Wakil Health Bldg., D18 Western Ave., Camperdown, New South Wales 2006, Australia, cate.madill@sydney.edu.au), Antonia Chacon, Tuende Szalay (Speech Pathol., Univ. of Sydney, Camperdown, New South Wales, Australia), and Duy D. Nguyen (Speech Pathol., Univ. of Sydney, Sydney, New South Wales, Australia)

There exists a large amount of predominantly low-level evidence that behavioral voice therapy is effective but there is limited evidence that one therapy is more effective than another. Variability in voice therapy treatment outcomes research leaves clinicians with little guidance in the selection of treatment. A range of acoustic outcome measures with differing sensitivity for detecting voice change are used, statistical analyses vary and experimental design presents both pros and cons regarding applicability of outcomes to real-world clinical settings. In this presentation, a review of acoustic outcome measures, statistical methods, and research design used to evaluate treatment for two clinical cohorts—muscle tension voice disorder and gender affirming voice therapy—will be presented. Two treatments investigated across the two cohorts (Resonant Voice Therapy and Sob Voice Therapy) will be used to demonstrate how acoustic analysis has been used and how implications of results differ. Method: Comparison of data available in published literature will be presented alongside new data as yet unpublished. Acoustic measures, research design, and statistical tests used will be evaluated for sensitivity to voice change, level of evidence, and translation to clinical practice.

Session 5aSP

Signal Processing in Acoustics and Physical Acoustics: Passive Acoustic Sensing of the Underwater Environment Using Gliders and Uncrewed Renewable Energy Powered Surface Vessels

Brian G. Ferguson, Cochair
DSTG, 13 Garden Street, Eveleigh 2015, Australia

Kay L. Gemba, Cochair
Physics, Naval Postgraduate School, Spanagel Hall Room 203, 833 Dyer Road, Monterey, CA 93943

Chair's Introduction—8:15

Contributed Papers

8:20

5aSP1. Acoustic signals received on buoyancy gliders in the Beaufort Sea. Lora Van Uffelen (Ocean Eng., Univ. of Rhode Island, 215 South Ferry Rd., 213 Sheets Lab., Narragansett, RI 02882, loravu@uri.edu), Jessica Desrochers (Naval Undersea Warfare Ctr., Div. Newport, Newport, RI), Luis O. Pomales Velázquez (Univ. of Rhode Island, Narragansett, RI), Isaac Salazar (Ocean Eng., Univ. of Rhode Island, Narragansett, RI), Sarah Webster (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Matthew A. Dzieciuch (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA), and Peter F. Worcester (Scripps Inst. of Oceanogr., Univ. of California, San Diego, San Diego, CA)

The buoyancy glider is a quiet and persistent underwater acoustic receiving platform. Traveling in a sawtooth pattern, buoyancy gliders equipped with acoustic recorders can sample acoustic transmissions at many ranges and depths with respect to moored acoustic sources transmitting on a timed schedule. In the Beaufort Sea, six broadband acoustic tomography sources consecutively transmitted 135-s linear frequency modulated (LFM) swept-frequency signals centered near 250 Hz every 4 h. These signals were received by two Seagliders at ranges up to 500 km and depths between the surface and 800 m in the summer of 2017. Sources were moored within the Beaufort Duct, a sound-speed minimum characteristic of the region, at a depth of approximately 180 m. Due to the presence of this duct, many acoustic paths are focused within a relatively narrow depth span, resulting in a complicated arrival structure. Pulse-compressed acoustic signals received on the gliders are interpreted in the context of broadband acoustic arrival predictions. The individual snapshots of acoustic arrival structure that make up this unique dataset offer insight into the acoustic travel-time arrival structure as it evolves with range from a transmitting source.

8:40

5aSP2. Passive acoustic glider for marine acoustic environment characterization. Yong-Min Jiang (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC V8W 2Y2, Canada, minj@uvic.ca), Stan Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada), Luigi Troiano, and Alain Maguer (NATO-STO Ctr. for Maritime Res. and Experimentation, La Spezia, La Spezia, Italy)

Acoustic payload-equipped underwater gliders have great potential for assessing the marine soundscape (oceanic acoustic environment), including characterizing properties of the sound field, of acoustic sources, and of the marine environment (water column and seabed). This talk presents observations and analysis of passive acoustic recordings carried out by gliders during the 2017 Seabed Characterization Experiment (SBCEX17). Two Teledyne Webb Research Slocum gliders equipped with omni-directional hydrophones were deployed as virtual acoustic moorings in 72 m deep water on the New England Mud Patch for time periods of $\sim$6 days for one glider and $\sim$3 days for the other. Acoustic recordings collected by the gliders captured the marine soundscape, including natural ambient noise, marine-life sounds, ship noise, and acoustic signals transmitted by the SBCEX17 participants. The acoustic environment in the water column is characterized in terms of the acoustic power spectral density as a function of time, depth, and frequency. As an example of the use of passive acoustic gliders for environmental characterization, Bayesian geoacoustic inversion for seabed properties is presented, making use of recordings of controlled sources associated with SBCEX17. [Work supported by the Office of Naval Research.]

9:00

5aSP3. Source range and depth estimation of propeller cavitation bubble collapse transients in a multipath environment. Brian G. Ferguson (DSTG, 13 Garden St., Eveleigh, New South Wales 2015, Australia, brian.ferguson@defence.gov.au) and Eric L. Ferguson (ACFR, The Univ. of Sydney, Sydney, New South Wales, Australia)

An experiment is conducted in which a wide aperture array, which is proximate to the sea floor in water 20 m deep, records transits of two motorboats. Occasionally, acoustic transients, which are attributed to collapsing propeller cavitation bubbles, are observed to be superimposed on the continuous broadband radiated noise component of each of the array's three widely spaced hydrophones. By measuring the time differences of arrival of the transient signals propagating via the direct path at adjacent pairs of hydrophones, passive ranging by wavefront curvature is used to estimate the range and bearing of the transient source. Next, the source depth is estimated by measuring the multipath delay of the surface reflected transient signal. Also, the spectrogram of a single hydrophone's output is observed to display a Lloyd's mirror interference pattern due to the direct propagation path of the broadband radiated noise combining with the indirect propagation path signal reflected from the sea floor. The reciprocal of the frequency difference between adjacent destructive interference fringe minima is equal to the multipath time delay. This multipath delay, which equates to the quefrency of the cepstrum's first rahmonic, contains source range information. The results of the two passive ranging methods are presented.

Session 5aUW

Underwater Acoustics: Computers in Underwater Acoustics

Alan Hunter, Cochair
University of Bath, Claverton Down, Bath BA2 7AY, United Kingdom

Christina Frederick, Cochair
Department of Mathematical Sciences, NJIT, 323 M. L. King Boulevard, Newark, NJ 07102

Contributed Papers

8:00

5aUW1. Deep learning and beamforming comparison of source ranging in a laboratory tank. Corey E. Dobbs (Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, cedobbs@byu.edu) and Tracianne B. Neilsen (Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Deep learning has been shown to be useful for ocean acoustics applications, such as source ranging. The variable nature of the ocean causes difficulty when attempting to apply trained models to new datasets. To explore ways to train supervised deep learning models that are robust to environmental variability, laboratory tank measurements can be used. A laboratory tank presents a system where uncertainty in the acoustic environment can be adjusted in a more controlled manner. In this work, recordings of ultrasonic chirps (50–200 kHz) are obtained at different source-receiver positions. Time-averaged spectral density levels are used to train one-dimensional convolutional neural networks that predict source-receiver range. The trained networks are tested on data obtained under different conditions. Data augmentation techniques are applied to increase the models' robustness in the presence of limited uncertainty. Metrics such as root mean-squared error and mean absolute percent error are used to quantify the model's performance. The results from deep learning with single channel recordings and multi-channel recordings are compared with traditional beamforming methods on the multi-channel data. [Work supported by the Office of Naval Research.]

8:20

5aUW2. Simulation-aided machine learning: Applications in underwater acoustic perception and explainability. Alan Hunter (Univ. of Bath, Claverton Down, Bath BA2 7AY, United Kingdom, a.j.hunter@bath.ac.uk), Oscar Bryan, Edward Clark, and Ciaran Sanford (Univ. of Bath, Bath, United Kingdom)

Underwater acoustic applications of machine learning are challenging due to sparsity, lack of diversity, and often uncertain labeling of training data. This is due to low levels of consumer relevance. One approach to the problem is to collect and label more data. However, this is expensive and unlikely to fully address the issue. A complementary approach is to use simulation for generating synthetic data and augmenting the machine learning pipeline. This also enables explainability by encoding human knowledge of the underlying physics. We present two case studies in simulation-aided machine learning. The first is an application to underwater munition dumpsite characterization using multi-view sonar surveys. Self-supervised learning has been used to pre-train a generative deep neural network to predict alternate views using unlabeled data. Subsequently, training is completed using a small volume of labelled data. Simulation was necessary for successful pre-training to continuously fill viewing angle gaps caused by limited and rigid survey directions. The second application is autonomous tactical planning for uncrewed underwater vehicles. Reinforcement learning policies have been trained to passively detect and localize underwater acoustic sources. These have been trained and evaluated in a fully synthetic environment, using a sophisticated propagation model together with empirical environmental parameters.

8:40

5aUW3. Machine learning based ship localization in shallow water using ship noise recorded by two vertical line arrays. Moon Ju Jo (Dept. of Marine Sci. & Convergence Eng., Hanyang Univ., 55, Hanyangdaehak-ro, Sangnok-gu, Ansan, Gyeonggi-do 15588, Korea (the Republic of), jmjnz@hanyang.ac.kr), Dong-Gyun Han (Res. Ctr. for Ocean Security Eng. and Technol., Hanyang Univeristy ERICA, Ansan, Gyeonggi-do, Korea (the Republic of)), Su-Uk Son (Agency for Defense Development, Changwon, Korea (the Republic of)), and Jee Woong Choi (Dept. of Marine Sci. & Convergence Eng., Hanyang Univ., Ansan, Korea (the Republic of))

For the application of machine learning to sound source localization, much train data distinguished from test data is needed to build the machine learning model. In Shallow-water Acoustic Variability Experiment (SAVEX-15) conducted in shallow water (water depth $\sim$100 m) in Northern East China Sea (ECS), ship noise of the R/V Onnuri was recorded by two vertical line arrays. Acoustic data of $\sim$80% was applied to the training dataset and the others having different trajectories were used for the test data. The recorded data is preprocessed by a sample covariance matrix and it is used as the input data of the machine learning model: Feedforward neural network (FNN) and support vector machine (SVM). The results by FNN and SVM will be discussed with conventional localization method using ray-based blind deconvolution (RBD) and array invariant (AI). [Work supported by the Agency for Defense Development, Korea (UD210004DD).]

9:00

5aUW4. Improving mode extraction with physics-informed neural network. Seunghyun Yoon (Seoul National Univ., 1, Gwanak-ro, Gwanak-gu, Seoul, Republic of Korea, 34-305, Seoul 08826, Korea (the Republic of), justin1128@snu.ac.kr), Yongsung Park (Scripps Inst. of Oceanogr., San Diego, CA), and Woojae Seong (Seoul National Univ., Seoul, Korea (the Republic of))

This study aims to enhance conventional mode extraction methods in ocean waveguides using a physics-informed neural network (PINN). Mode extraction involves estimating mode wavenumbers and corresponding mode depth functions. The approach considers a scenario with a single frequency source towed at a constant depth and measured from a vertical line array (VLA). Conventional mode extraction methods applied to experimental data face two problems. First, mode shape estimation is limited because the receivers only cover a partial waveguide. Second, the wavenumber spectrum is affected by issues such as Doppler shift and range errors. To address these challenges, we train the PINN with measured data, generating a densely sampled complex pressure field, including the unmeasured region above the VLA. We then apply the same mode extraction methods to both the raw data and the PINN-generated data for comparison. The proposed method is

validated using data from the SWellEx-96, demonstrating improved mode extraction performance compared to using raw experimental data directly.

9:20

5aUW5. Optimal sampling strategies for seabed classification and source localization with Gaussian processes and machine learning. Zoi-Heleni Michalopoulou (Mathematical Sci., New Jersey Inst. of Technol., Newark, NJ) and Christina Frederick (Dept. of Mathematical Sci., NJIT, 323 M. L. King Boulevard, Newark, NJ 07102, christin@njit.edu)

Workshop '97 data are employed for seabed classification and source range estimation. The data are acoustic fields computed at vertically separated receivers for various ranges and three different environments. Gaussian Processes are applied for denoising the data and predicting the field at virtual receivers, sampling the water column densely within the array aperture. The enhanced fields are then used in combination with machine learning in order to map the signals to one of 15 sediment-range classes (corresponding to three environments and five ranges). In prior work, the classification results after using Gaussian Processes for denoising were demonstrated to be superior to those when noisy workshop data are employed. Here, we explore optimal sampling strategies (e.g., nonuniform sampling, subsampling) for inducing sparsity in the correlation matrices that are based on hydrophone locations, and compare these with uniform sampling that was used in our prior work.

9:40

5aUW6. Effective feature fusion via analysis of quantitative similarity matrices among various acoustic features for underwater active target detection. Sungjin Shin (Dept. of Ocean Systems Eng., Sejong Univ., 209, Neungdong-ro, Seoul, Gwangjin-gu 05006, Korea (the Republic of), sungjin900998@gmail.com), Geunhwan Kim, YoungSang Hwang (Dept. of Ocean Systems Eng., Sejong Univ., Seoul, Korea (the Republic of)), Juho Kim (Sonar System PMO, Maritime Technol. Res. Inst., Agency for Defense Development, Changwon, Korea (the Republic of)), and Youngmin Choo (Dept. of Defense Systems Eng., Sejong Univ., Seoul, Korea (the Republic of))

For underwater active target detection, using conventional machine learning technique is limited and unsuitable because of small training data samples and diversity of environment. Therefore, to apply conventional machine learning for underwater acoustics target detection, three methodologies can be manipulated (1) feature, (2) architecture, and (3) learning strategy. In this paper, we implement various acoustic features in terms of feature similarity and feature fusion. From numerous studies in field of acoustics, various acoustic feature extraction methods have been proposed such as Mel-frequency cepstral coefficient, Gammatone-frequency cepstral coefficient, cepstral coefficient, short-time Fourier transform, constant Q transform, and wavelet packet decomposition. In this paper, we calculate a quantitative similarity between acoustic features by interpreting their data distributions with the corresponding probability densities in a reduced dimension. Furthermore, we fuse the acoustic features by simple concatenation. Fusion of a strongly correlated two-dimensional features tends to follow the performance of poor one, whereas the fusion of weakly correlated features improves performance remarkably. The performance improvement by the fusion of weakly correlated features is attributable to complementing acoustic information each other.

10:00–10:20 Break

10:20

5aUW7. On coherently processing multiple arrays in underwater environments with spatial coherence losses from random medium effects and sensor position uncertainties. Ivars Kirsteins (NUWCDIVNPT, 1176 Howell St., Newport, RI 02841, i.kirsteins@gmail.com) and Ahmad Abawi (HLS, La Jolla, CA)

Under ideal circumstances coherently beamforming multiple distributed arrays should provide improvements over the standard approach of processing the arrays individually and combining them incoherently. These include improved directivity and higher gain. However, two major challenges when coherently processing spatially separated arrays in a real ocean are dealing with coherence losses or wave front distortions caused by random medium effects like internal waves and uncertainties in sensor locations. Some of the consequences are degraded beampatterns and poorer array directivity. In this paper, we perform a simulation-based study utilizing numerical propagation models along with theory examining the effects of wave front distortions induced by random medium effects like internal waves and sensor position uncertainties on coherently beamforming distributed arrays in representative ocean waveguides. Realistic realizations of wave front distortions are generated from actual internal wave measurements and models. The sensitivities of coherent processing to these effects are characterized by resultant beampatterns and losses in array directivity along with a quantification of the tolerances.

10:40

5aUW8. Frequency analysis of passive sonar signals using multiple-measurement physics-informed neural network. Sunyoung Ko (Sejong Univ., 209, Neungdong-ro, Gwangjin-gu, Seoul, Republic of Korea, 1006, Kwanggaeto Bldg., Seoul 05006, Korea (the Republic of), smail7045@naver.com), Mingu Kang, Junho Bae (Sejong Univ., Seoul, Korea (the Republic of)), Woojae Seong (Seoul National Univ., Seoul, Korea (the Republic of)), Myoungin Shin (Agency for Defense Development, Seoul, Korea (the Republic of)), and Youngmin Choo (Sejong Univ., Seoul, Korea (the Republic of))

We propose a deep learning based frequency analysis for passive sonar signals by considering a linear system based on a relation between time and frequency domain signal representations. An adaptive learned iterative shrinkage thresholding algorithm (Ada-LISTA) is utilized as the deep learning architecture to solve the linear system effectively and to ensure generalization performance. To obtain more reliable solutions, we involve the linear system in loss function during training. Futhermore, in order to reduce noise effectively, we employ multiple measurements from time and space domains, which have common frequency components of passive sonar signals. Thus, the loss function is modified for solutions from the multiple measurements to have shared frequency components. We conduct experiments using synthetic and underwater *in-situ* data to examine performance of frequency analysis considering the linear system in designing the architecture and finding the optimal weights. The frequency analysis demonstrates superior performance in detecting frequency components and reducing noise, compared to fast Fourier transform and sparse Bayesian learning with low computational burden during its application to the test measurements.

11:00

5aUW9. Learning dictionary-based passive underwater source localization. Ivan I. Rodriguez-Pinto (Littoral Acoust. & Target Phys., Naval Surface Warfare Ctr. Panama City Div., 110 Vernon Ave. 2C09B, Panama City, FL 32407, ivan.i.rodriguez-pinto.civ@us.navy.mil) and Raymond Lim (Littoral Acoust. & Target Phys., Naval Surface Warfare Ctr. Panama City Div., Panama City, FL)

We demonstrate the use of dictionary learning to improve passive localization of a source in a range-independent water column monitored with a vertical linear acoustic array. A learned dictionary, generated by applying modern dictionary learning algorithms to historical sound velocity profile (SVP) data from a region of interest, acts as a generalized sparse representation for SVP datasets from this region. By minimizing differences between multipath reception intervals detected at a receive array, we show that an optimal SVP match to the unknown true SVP can be reconstructed by an efficient dictionary search. Ray traces backpropagated through the reconstructed SVP according to beamformed receive angles were then found to well estimate the source position as the centroid of a cluster of multipath signal intersections. The accuracy of this algorithm was evaluated on a randomly sampled, 30 SVP testing set obtained from an area near the training set, demonstrating mean location errors less than 5% in both distance and depth. Five representative profiles spanning the range of observed sound speed variations were used to assess localization performance. Compared to using the average sound speed profile to estimate source position, using the learned dictionary demonstrated a consistent performance increase in position accuracy.

11:20

5aUW10. Deep learning based effective feature extraction for active sonar target detection. YoungSang Hwang (Dept. of Ocean Systems Eng., Sejong Univ., 209, Neungdong-ro, Gwangjin-gu, Seoul 05006, Korea (the Republic of), ldzezl@naver.com), Geunhwan Kim, Sungjin Shin, Wooyoung Hong, and Youngmin Choo (Dept. of Ocean Systems Eng., Sejong Univ., Gwangjin-gu, Seoul, Korea (the Republic of))

We propose a deep learning (DL) model suitable for classifying target and non-target using a small amount of active sonar data. The proposed model uses two hand-crafted features (STFT and CQT) extracted from the same raw active sonar data, to complement each other and enhance the generalization of DL under insufficient data. The attention-based complementary learning module in proposed model reinforces one feature by referring to the other feature. The comprehensive feature from the complementary learning module pass through a shallow layer CNN and classify targets and non-targets. To verify the performance of the proposed model, we compared it with prevalent deep learning models including ResNet and ViT in terms of generalization performance and learning stability using two real-ocean datasets. The generalization performance of the proposed model having much smaller number of parameters was similar to or superior to the existing deep learning models depending on the training dataset, while the proposed model had the best learning stability for the two datasets.

11:40

5aUW11. Deep learning approach to generate propeller cavitation noise. Youngjoo Kim (Sejong Univ., 209, Neungdong-ro, Seoul, Gwangjin-gu 05006, Korea (the Republic of), submakim@gmail.com), Jongkwon Choi, Wooyoung Hong, and Keunhwa Lee (Sejong Univ., Seoul, Korea (the Republic of))

In underwater acoustic simulators, propeller cavitation noise has traditionally been modeled as a modulated broadband signal. This study aims to enhance the realism of these simulators by employing deep learning to generate propeller cavitation noise. The training data were collected from the modeled propeller under various pressure conditions in the Samsung Cavitation Tunnel. We have utilized a variant of Generative Adversarial Networks (GANs), wherein both the generator and discriminator are designed with a recursive structure. To assess the advantages of our data-based approach, we analyze the characteristics of the signals generated and compare them with the real signals or the modeled signal based on physics. The results demonstrate the effectiveness of our proposed model in generating realistic propeller cavitation signals. [Work supported by the Korea Research Institute for Defense Technology Planning and Advancement (20-106-B00-003).]

Session 5pAB

Animal Bioacoustics, Acoustical Oceanography, Underwater Acoustics, and Signal Processing in Acoustics: Fisheries and Marine Park Management II

Xavier Mouy, Cochair
Northeast Fisheries Science Center, National Oceanic and Atmospheric Administration, National Marine Fisheries Service, 166 Water Street, Woods Hole, MA 02543

Miles Parsons, Cochair
AIMS, 71a Aurelian St, Palmyra, Perth 6157, Australia

Chair's Introduction—12:55

Invited Paper

1:00

5pAB1. Moonlight driven biological choruses in coral reefs. Daniel Duane (Naval Undersea Warfare Ctr., 1176 Howell St., Newport, RI 02841, daniel.m.duane.civ@us.navy.mil), Simon Freeman (ARPA-E, Washington, DC, DC), and Lauren A. Freeman (Naval Undersea Warfare Ctr., Newport, RI)

Sounds from fish and invertebrates in coral reefs can create persistent cacophonies that can be recorded for ecosystem monitoring, including during nighttime hours where visual surveys are typically not feasible. Here, we use soundscape measurements in Hawaii and Bermuda to demonstrate that multiple coral reef communities are rapidly responsive to shifts in nighttime ambient light, with sustained changes in biological sound between moonrise and moonset. Feeding sounds from parrotfish become more prominent as moonlight increases nighttime visibility, and fish vocalizations become less prominent with moonlight, likely to minimize detection during times with increased predation. The response of invertebrate sounds to moonlight varies by location, increasing in Hawaii reefs and decreasing in Bermuda reefs. These discoveries suggest that the rising and setting of the moon triggers regular shifts in coral reef ecosystem interactions. Future acoustic monitoring of reef health may be improved by comparing soundscapes during moonlight and non-moonlight hours, which may provide early indicators of shifts in the relative abundance of separate reef communities.

Contributed Papers

1:20

5pAB2. Biological chorusing and fish calls in Bermuda. Lauren A. Freeman (NUWC, Naval Undersea Warfare Ctr., Newport, RI, lauren.a.freeman3.civ@us.navy.mil), Daniel Duane (Naval Undersea Warfare Ctr., Newport, RI), and Simon Freeman (ARPAE, Washington, DC, DC)

A series of sea tests and long term passive acoustics monitoring studies have been conducted in Bermuda from 2020–2023. Observations of biological choruses and individual fish call events from sensors in deep water off the flank of the Bermuda Atoll, and alongside shallow sub-tropical reefs, will be discussed. We find striking similarities between sub-tropical Atlantic reefs in Bermuda with documented trends in tropical coral reef soundscapes studies in Hawaii and the tropical Pacific. This observation suggests that evening chorusing and patterns in reef sound and fish calls are potentially pervasive across large portions of the world's oceans. Deep water calls that appear to be associated with large fish such as groupers were observed on the atoll flank. Evening chorusing at change of light is commonly observed on tropical reefs and also persistent on Bermuda's sub-tropical reefs. Evening biological chorusing is also evident at deep water sights, indicating that this may not be a strictly littoral biological soundscape feature.

1:40

5pAB3. Soundscapes of the central Red Sea mesophotic reef zones. Michelle-Nicole Havlik (Red Sea Res. Ctr., King Abdullah Univ. of Sci. and Technol., KAUST University, Thuwal, Makkah 23955, Saudi Arabia, michellenicole.havlik@kaust.edu.sa), Alexandra Steckbauer, Anieka Parry, Fabio Marchese, Marta Ezeta Watts, Francesca Benzoni, and Carlos Duarte (Red Sea Res. Ctr., King Abdullah Univ. of Sci. and Technol., Thuwal, Saudi Arabia)

Despite covering a large amount of seafloor & representing a significant percentage of coral reef ecosystem biomass, Mesophotic Coral Ecosystems (MCEs) globally have been largely understudied in comparison to their shallow counterparts. The Red Sea is a unique and biodiverse tropical ecosystem, where few mesophotic coral reefs have been studied. To define spatiotemporal activity of marine life and characterize the soundscapes of the MCE in the Central Red Sea, two locations were surveyed at 70–80 m, as well as at two adjacent shallow reef sites at $\sim$10 m depth, respectively, during the winter (Jan 2022) and summer (Jul–Aug 2022) seasons. The results of this study, the first of its kind in the Red Sea, show a clear shift in the soundscape between seasons, as well as between the shallow and mesophotic zones, driven mainly by a change in fish and invertebrate chorusing. During the study periods, temperature and oxygen levels in the mesophotic zone remained relatively stable, in comparison to the shallow reef zone which saw a steep temperature increase and large diel fluctuations in oxygen in the summer. Chronic exposure to shipping noise is evident across Red Sea MCE soundscapes and the resulting potential effects are discussed.

Invited Papers

2:00

5pAB4. What does protection sound like? A modern approach to understanding New Zealand's underwater soundscapes and acoustic anthropogenic pressures. Jenni A. Stanley (School of Sci., Univ. of Waikato, Hamilton, New Zealand, jstanley@waikato.ac.nz) and Leah M. Crowe (Marine Sci., Univ. of Otago, St. Clair/Dunedin, Otago, New Zealand)

Global marine ecosystems have experienced degradation and loss of biodiversity as a result of human impacts and climate change. Monitoring of ecologically significant areas is vital in understanding these effects and their subsequent management. Soundscapes offer a unique opportunity for examining organisms and habitats in a way that eliminate many potential weaknesses of traditional monitoring techniques. This fundamental property of every environment is being increasingly influenced by the presence of human activities but is not yet recognized for the breadth of information it holds, and as a key indicator of change in an ecosystem. A nation-wide program, focusing on several high-priority protection sites, utilize passive acoustic monitoring (PAM) creating baseline datasets. With the emphasis on characterizing the ambient soundscapes and acoustic signatures of significant species, identifying key acoustic parameters that reflect ecosystem health and biodiversity, and in locations where human use will potentially contribute to negative acoustic exposures. Here, spatiotemporal trends in the underwater soundscapes at sites within Fiordland National Park are reported, illustrating diel, lunar, and seasonal patterns in sound pressure levels, identifying biotic, abiotic, and anthropogenetic sound sources where possible. PAM is an increasingly useful tool in understanding anthropogenic presence and ecosystem status, and offers an effective, minimally invasive, and less labor-intensive way of monitoring marine ecosystems.

2:20–2:40 Break

Contributed Papers

2:40

5pAB5. Behavioral responses of *Chinook salmon* to shipping noise in Cowichan Bay, British Columbia. Kelsie A. Murchy (Biology, Univ. of Victoria, 3800 Finnerty Rd., Victoria, BC V8W 3N5, Canada, kmurchy@uvic.ca), Will Duguid (Biology, Univ. of Victoria, Victoria, BC, Canada), Jamieson Atkinson (BC Conservation Foundation, Nanaimo, BC, Canada), Jessica Qualley, Katie G. Innes, Hailey L. Davies, Bridget Maher, and Francis Juanes (Biology, Univ. of Victoria, Victoria, BC, Canada)

The increase in human-generated noise over the last 60 years has led to concerns regarding the impacts of shipping noise on marine species. Two important species in the northeast Pacific are declining, southern resident killer whales and their main prey, *Chinook salmon*. We know that killer whales change their behavior in the presence of ships, but no work has been done on salmon. Acoustic receivers and underwater hydrophones were deployed in Cowichan Bay, British Columbia, Canada, to understand potential changes in behavior as salmon encounter shipping noise. Depth and acceleration of adult *Chinook salmon* were monitored using acoustic tags and were modeled against environmental conditions (e.g., currents) to understand general movement and behavior. Underwater sound pressure levels were then added to the model to assess potential responses to ship noise. Over 3 years, 53 *Chinook salmon* were tagged, resulting in 167,628 individual detections. Detections revealed spatial and temporal patterns in habitat use and behavior prior to river entry and provided the first data on *Chinook salmon* responses to anthropogenic noise. These data provide novel insights into the behavior of adult *Chinook salmon* and crucial information on the impacts of anthropogenic noise on this key species.

3:00

5pAB6. An automated approach for detection and classification of toothed whales in Hawai'i marine protected areas. Brijonnay Madrigal (Marine Mammal Res. Program, Univ. of Hawai'i at Mānoa, 46-007 Lilipuna Rd., Kaneohe, HI 96744, bcm2@hawaii.edu), Jennifer McCullough (NOAA Pacific Islands Fisheries Sci. Ctr., Honolulu, HI), Marc Lammers (Hawaiian Islands Humpback Whale National Marine Sanctuary, Kihei, HI), Erin Oleson (NOAA Pacific Islands Fisheries Sci. Ctr., Honolulu, HI), Lars Bejder, and Aude Pacini (Marine Mammal Res. Program, Univ. of Hawai'i at Mānoa, Kaneohe, HI)

Passive acoustic monitoring is an effective technique for studying cetacean presence within marine protected areas (MPAs). The Hawaiian archipelago is home to 18 species of resident toothed whales, but little is known regarding the spatio-temporal variability of false killer whales (*Pseudorca crassidens—FKW*) and short-finned pilot whales (*Globicephala macrorhynchus*) across Hawaiʻi MPAs. Bottom-moored recorders were deployed inside and outside the Hawaiian Islands Humpback Whale National Marine Sanctuary and Papahānaumokuākea Marine National Monument through SanctSound and Pacific Islands Fisheries Science Center. High Frequency Acoustic Recording Packages (HARPs) and SoundTraps (200 kHz/48 kHz SR, respectively) deployed between 2018 and 2022, allowed leveraging of long-term datasets. Data used in this study include 27 deployments across 11 sites, encompassing >187 months of data. Here, we describe an automated approach for species classification which is challenging given the high spectral overlap in whistle features and acoustic masking during humpback whale season. PAMGuard detectors were used to detect whistles, clicks, and burst pulses. Features were extracted using PAMpal and species classified using BANTER models and confirmed from a Hawaiian Islands Ecosystem Assessment Survey annotated dataset. Acoustic monitoring provides vital information about the value of MPAs and a better understanding of endangered insular FKW habitat use.

3:20

5pAB7. Unsupervised beluga whistle end-point detection under burst pulse interferences. Yibo Zhao (College of Underwater Acoust. Eng., Harbin Eng. Univ., No.145 Nantong St., Nangang District, Harbin, 150001, P.R.China, zhaoyibo1998@hrbeu.edu.cn), Songzuo Liu, Gang Qiao, and Xin Qing (College of Underwater Acoust. Eng., Harbin Eng. Univ., Harbin, China)

Passive acoustic monitoring is an increasingly important tool for whale research. Accurately detecting the end-point of whale whistle is an essential task in the study of marine mammal calls. However, the detection accuracy of the beluga whistle is reduced due to the non-stationary burst pulse interference in the marine environment. In this paper, an unsupervised two-stage beluga whistle end-point detection method is proposed to solve the above problem. Based on the high Q-factor wavelet decomposition, a rough whistle detection method is proposed to remove the silent time and the low intensity pulse. On this basis, local density adaptive spectral clustering is designed to further distinguish whistle and strong pulse interference based on the sparsity difference in time-frequency domain. The performance of the detector is tested with real signals of beluga whales, and its F_1-score is calculated. The result shows that the detector is obviously better than the traditional whistle detectors under the background of burst pulse interferences, and has higher robustness. In the future, the presented method is supposed to be applied for detecting whistles of some other whale species.

Session 5pMU

Musical Acoustics: Player-Instrument Interaction II

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

Andre Almeida, Cochair
Physics, UNSW, 8/133 Boundary St., Clovelly 2031, Australia

John Smith, Cochair
Physics, UNSW, Sydney 2052, Australia

Contributed Papers

1:00

5pMU1. Lip-valve interactions with the resonant bore of a trombone. Henri Boutin (IRCAM, Sorbonne, Paris, France), John Smith (Phys., UNSW, Phys., UNSW, Sydney, New South Wales 2052, Australia, john.smith@unsw.edu.au), and Joseph Wolfe (Phys., UNSW, Sydney, New South Wales, Australia)

Players of brass instruments usually play at a pitch close to one of the strong resonances of the instrument bore, but can easily play either above or below a resonance. Simple one-degree-of freedom models having aperture in phase with longitudinal lip displacement do not readily explain auto-oscillation with realistic values of lip losses and cannot explain oscillation at frequencies below resonance. Experimental measurements of lip motion, pressures and flows show that the lips' longitudinal motion leads the transverse: the lips move downstream closed, then open, then retract while open. Because of the forward sweeping motion, flow starts before the lips open. The sweeping motion has two important effects: first, it allows a range of phases between pressure and flow. Second, it means that the DC pressure acts on a larger area in the forward motion than during retraction. Both have consequences for oscillation energetics. A simple calculation of the PdV work done on the longitudinal component of motion, using measured quantities, indicates that it could provide a substantial fraction of the energy necessary for auto-oscillation with lossy lips. These mechanisms might have implications for other auto-oscillatory systems such as the vocal folds.

1:20

5pMU2. A mathematical study about the sustaining phenomenon of overtone in Flageolet harmonics on bowed string instruments. Shodai Tanaka (Sapporo Kaisei Secondary School, 21 Chome-1-1 Kita 22 Johigashi, Higashi Ward, Sapporo, Hokkaido 065-0022, Japan, shodai.harmonics@gmail.com), Hiroshi KORI (Complexity Sci. and Eng., The Univ. of Tokyo, Kashiwa, Chiba, Japan), and Ayumi Ozawa (The Univ. of Tokyo, Kashiwa, Japan)

Flageolet harmonics is a playing technique, in which a player lightly touches a nodal point on a string with their finger. Previous studies have reported that the harmonic sound sustains for a short time after the finger is removed from the string while flageolet harmonics is performed on bowed string instruments. However, the mechanism of this harmonics-sustaining phenomenon and the parameter dependency of its sustaining time remains unclear. The purpose of this study is to mathematically investigate the dependence of sustaining time on parameters related to violin playing and string characteristics, and thereby, to elucidate the mechanism of the harmonics-sustaining phenomenon. To this end, a mathematical model was devised by incorporating the effects of bowing and touching into a one-dimensional wave equation. Subsequently, numerical simulation were performed to analyze the behavior of the model. The devised model successfully reproduced the harmonics-sustaining phenomenon in which the parameter dependence of sustaining time was qualitatively consistent with the author's empirical observations. It was found that the parameter dependence of sustaining time follows the power law. Furthermore, dimensional analysis was performed, yielding a formula that expresses the relationship between the sustaining time and the maximum and minimum bow force required to generate Helmholtz motion.

1:40

5pMU3. Needs of musicians during performance, without electronic amplification, in Concert Halls and Opera Houses. Edward L. Harkness (PO BOX 335, Pyrmont, New South Wales 2009, Australia, tedharkness@edwardleoharkness.com)

Drawing on experience of having listened to music in 60 concert halls and opera houses around the World, conducted experimental stage and orchestra pit modifications for live performance with audiences, analyzed the onset characteristics of musical instruments and their attack sequences; manipulated the distribution of acoustic energy using stage furniture with performer responses; and taught acoustics for 40 years, the author offers insights into the design of stages and orchestra pits for the performance of music without electronic amplification, in the context of his experience as a musician, architect, and engineer. In the orchestra pit of the Sydney Opera House: A second desk first violinist said he could not hear his instrument before the furniture was installed; and said after the furniture had been installed, "I am like thunder!" The tympani player asked, "What have you done? My instruments are alive!" In the Macquarie Auditorium, a violinist playing her Stradivarius violin said, "I used to dread playing in that auditorium now I am soaring." The Australian Chamber orchestra refused to make a recording in the Gore Hill ABC TV Studio, without the stage furniture. What was done in several auditoria and responses from musicians is described in the paper.

2:00

5pMU4. Voice matching in sung duos: Is it related to spectrum envelopes? Kajornsak Kittimathaveenan (Div. of Speech, Music and Hearing (TMH), KTH Royal Inst. of Technol., Lindstedtsvägen 24, Stockholm SE-114 28, Sweden, kkit@kth.se), Sten Ternström (Div. of Speech, Music and Hearing (TMH), KTH Royal Inst. of Technol., Stockholm, Sweden), and Sara D'Amario (Dept. of Music Acoust. – Wiener Klangstil, mdw – Univ. of Music and Performing Arts Vienna, Vienna, Austria)

Choir singers report anecdotally that two voices can be perceived as a good match for each other, or not. Could "matching voices" be explained by

the spectrum envelopes? Thirteen singers sang a duo in unison or canon with an adjacent prerecorded reference singer, in a moderately reverberant room. Singers controlled the stimulus timbre, using variable filters in medium and high frequency bands. They were asked to adjust the filters, while singing, for "best" and "worst" perceived matching. The singers then performed the song again, but with the filters automatically set to their chosen (dis-)preferences. The Self-to-Other ratio as a function of frequency [SOR(f)] at the ipsilateral ear of the participant was estimated from multiple microphone signals to predict separately the long-time average spectra of Self and Other. Most participants rated the sound with extreme filter settings at $\pm$15 dB as the "worst" match, while "best" matches were fairly evenly distributed. Some but not all participants preferred the spectra to be complementary. However, at low frequencies, SOR(f) was about +10 dB, and very irregular but rarely negative at medium and high frequencies; so how an adjacent singer can be heard at all will require further investigation.

2:20

5pMU5. Emergence of metrically structured rhythms and inter-partner coordination in joint drum improvization. Alexander T. Han (Music, Stanford Univ., 660 Lomita Ct, Stanford, CA 94305, tae1han@stanford.edu) and Takako Fujioka (Music, Stanford Univ., Stanford, CA)

Improvization is a common aspect of music across cultures and eras. Due to its spontaneous nature, it often proves difficult to examine empirically how collective improvization evolves, unless genre-specific constraints are available and followed by experts (e.g., jazz). Here, we investigate the rhythmic dimension of free improvization by non-experts. We focus on two widely used approaches in joint musical improvization: call-and-response (trading) and simultaneous playing (tandem). We hypothesized that non-experts could engage in meaningful joint music-making with a model partner, and that they produce rhythmic patterns with different emergent structures depending on the task. The first author served as a confederate playing with each participant for five blocks each of trading and tandem tasks. To preserve the open-ended nature of improvization, neither metronome nor background music were provided. Preliminary analysis of the rhythmic content indicates that within-player inter-onset-intervals (IOI) reflected a clear hierarchical metrical structure. Additional analysis suggests bi-directional influence between partners in terms of IOI distribution, note density, and timbre choice. During tandem improvization, each partner favored timbres the other did not, showing a complementary pattern. During trading, timbre choice appears more imitative. Subjects also reported an increase in their self-assessed competence and enjoyment of the task over successive blocks.

Session 5pNSa

Noise and Physical Acoustics: Rocket Noise

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

Daniel Edgington-Mitchell, Cochair
Mechanical and Aerospace Engineering, Monash University, Department of Mechanical and Aerospace Engineering, 14 Alliance Lane, Clayton 3800, Australia

Chair's Introduction—12:55

Invited Papers

1:00

5pNSa1. Rocket noise: What does it mean for Australian spaceports? Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., N281 ESC, Provo, UT 84602, kentgee@byu.edu), Logan T. Mathews, Bradley McLaughlin, Mark C. Anderson (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), Daniel Edgington-Mitchell (Dept. of Mech. and Aerosp. Eng., Monash Univ., Clayton, Victoria, Australia), and Grant W. Hart (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

As the global space industry expands, rockets are being launched from an increasing number of spaceports, including in Australia. As launch cadence increases to meet demand for space access and as vehicle optimization for weight and cost reduction becomes more pressing, noise has the potential to create harmful impacts—from vehicle vibroacoustic loading to expanded environmental footprint. Using data from recently measured launches, this presentation reviews rocket noise generation and propagation fundamentals and discusses some of these impacts. As small- and medium-payload orbital rocket launches are currently planned for Australian spaceports, preliminary noise predictions around these spaceports are described for representative vehicle launches. Eventual refinement and validation of these predictions will aid in assessing potential noise impacts on threatened and endangered species, such as the southern emu wren.

1:20

5pNSa2. Ongoing analysis of far-field acoustical measurements during the Artemis-I launch. Makayle S. Kellison (Dept. of Phys., Rollins College, Box 2743, Winter Park, FL 32789, mkellison@rollins.edu), Kent L. Gee, Carson F. Cunningham, Mark C. Anderson, Grant W. Hart (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), and Whitney L. Coyle (Dept. of Phys., Rollins College, Winter Park, FL)

As the global space industry expands, impacts from super heavy-lift launch vehicle noise on payloads, communities, and natural habitats are better understood with improved source models. To support model development, this paper discusses ongoing analyses of far-field acoustical measurements made during the NASA Space Launch System (SLS) Artemis-I mission. Fifteen acoustical measurement stations were deployed prior to the launch, including ten autonomous stations within Kennedy Space Center and five manned stations off-Center, up to 50 km from the pad. This paper presents a brief summary of data analyzed from all 15 stations, including A-weighted and unweighted sound pressure levels and spectra. Additionally, the skewness of the pressure time derivative is shown as a quantifier of the noise's crackling component. Finally, source directivity and sound power analyses are presented. [Work supported in part by the NSF REU Program and the Utah Space Grant Consortium.]

Contributed Papers

1:40

5pNSa3. A look into far-field rocket noise modeling: Approaches and potential sources of error. Reese Rasband (Ball Aerosp., 2875 Presidential Dr., Fairborn, OH 45324, r.rasband18@gmail.com) and Alan T. Wall (Air Force Res. Lab., WPAFB, OH)

In recent years, rocket launches have elevated from novelty to being seen as means of transportation, exploration, and delivery. Defense, commercial, and public sectors are looking to increase their cadence in missions and ability to perform them. Like airports before them, spaceports are being developed and regulated. Much research has been done in rockets as a noise source, and how they may affect communities, structures, and wildlife. The most common approach for modeling the far-field rocket noise relies heavily on NASA SP-8072 and work done by Dr. Sally McInerny, a NASA researcher. The approach takes known mechanical properties of the rocket, such as rocket nozzle diameter, exit flow velocity, and number of engines to calculate the sound power of the system, and then propagate the noise to the receiver with an applied directivity factor. The level decay is based on spherical spreading, and additional attenuation factors such as atmospheric conditions and ground impedance. This paper attempts to show an overview of these different components and their effects on model predictions. Their relevance and limitations are also discussed.

2:00–2:20 Break

2:20

5pNSa4. Directivity and sound power of rockets: A comparative analysis of source characteristics and their implementation into a predictive model. Logan T. Mathews (Dept. of Phys. and Astronomy, Brigham Young Univ., N201 ESC, Provo, UT 84602, lmathew3@byu.edu), Hunter J. Pratt, Mark C. Anderson, and Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Fundamental jet and rocket noise modeling includes quantifying the acoustic power and directivity characteristics of the source. While many attempts have been made to define a model for the sound power and directivity of jets and rockets, many of these models have been based on flawed measurements or methodologies. This paper presents the directivity and sound power characteristics as obtained from launched rockets. The results are compared with those in the literature. The directivity and sound power properties are used in a standard source model for rocket noise, and the resulting predictions are evaluated.

2:40

5pNSa5. Frequency-dependent directivity and sound power measurements of two launched United Launch Alliance Atlas V rockets. Mark C. Anderson (Dept. of Phys. and Astronomy, Brigham Young Univ., ESC N201, Provo, UT 84602, mark.anderson@byu.net), Logan Mathews (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), Reese Rasband (Ball Aerosp., Fairborn, OH), and Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

The noise generated during rocket launches is both intense and directional. Historically, rocket noise directivity has been studied using static firing data. Although such tests result in greater averaging times at all angles, it is also possible to estimate directivity curves and sound power for a launched rocket so long as the trajectory data and microphone locations are known. Acoustical measurements were made during two United Launch Alliance Atlas V launches out of Vandenberg Space Force Base, CA. Multiple microphones were placed around the base ranging from 300–7000 m from the launch pad at different azimuthal angles around the pad. Frequency-dependent directivity curve estimates and sound power spectra are calculated and compared between measurement sites and launches. Directivity indices are compared with those found in the literature. The overall sound power levels agree well between the two launches, though a general trend persists where the sound power estimate decreases as a function of measurement distance. The sound power spectra are also nondimensionalized using Strouhal number scaling and compared with historical literature. [Work supported in part by the Oak Ridge Institute for Science and Education and the Air Force Research Laboratory.]

3:00

5pNSa6. Worldwide detection of rocket launches for space missions using International Monitoring System infrasound arrays. Christoph Pilger (BGR Hannover, Stilleweg 2, Hannover 30655, Germany, christoph.pilger@bgr.de), Patrick Hupe, and Peter Gaebler (BGR Hannover, Hannover, Germany)

The signals of rocket launches can be detected at infrasound arrays in thousands of kilometers distance. We use microbarometer arrays, which are part of the International Monitoring System (IMS) for the Comprehensive Nuclear-Test-Ban Treaty (CTBT), to identify and characterize rocket launches all over the world. We studied more than 1000 launch events for space missions and their infrasonic signatures. Even small-lift launch vehicles such as Electron rockets starting from New Zealand with payloads of only a few hundreds of kilograms, e.g., for small satellites, can regularly be detected by one or more remote infrasound stations. We present selected cases of interest, including the latest NASA Artemis 1 Space Launch System and SpaceX Starship launches as well as airborne rocket starts and small-lift launches by different companies. Furthermore, we present a systematic analysis of infrasound recorded from multiple and regularly launched vehicles like Ariane 5, Falcon 9, and various Soyuz and Long March rocket types. Our main focus is on the connection between the observed signatures and propagation-corrected acoustic amplitudes of rocket launches as well as rocket parameters like payload, size and thrust, rocket stage, and engine characteristics of different rocket types.

3:20

5pNSa7. Comparing acoustical measurements from Falcon 9 launches. Makayle S. Kellison (Dept. of Phys., Rollins College - Box 2743, Winter Park, FL 32789, mkellison@rollins.edu), Noah L. Pulsipher, Levi T. Moats, Mark C. Anderson, Logan T. Mathews, Carson F. Cunningham, Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), Megan R. McCullah-Boozer, Lucas K. Hall (Dept. of Biology, California State Univ., Bakersfield, CA), and Grant W. Hart (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

The Falcon 9 rocket has successfully launched nearly 300 orbital missions, providing the opportunity to study noise radiation and propagation variability. Acoustical measurements of several Falcon 9 launches have been made on and near Vandenberg Space Force Base, ranging 0.5 to 14 km from the launch pad. This paper's purpose is to compare collocated measurements from different Falcon 9 launches to begin to understand data variability as a function of launch and environmental conditions. One far-field location, at a distance of 8.4 km, has been measured across all launches, whereas several other locations spanned subsets of launches. This comparative analysis includes time-varying levels, spectra, and waveform statistics such as the pressure derivative skewness. Time periods of particular interest are liftoff, peak noise, and late into the launch when the vehicle is significantly downrange. [Work supported in part by the NSF REU program and by USACE.]

Session 5pNSb

Noise and Physical Acoustics: Benefits and Drawbacks of Non-Typical Hearing Protectors

Hilary Gallagher, Cochair
Air Force Research Laboratory, United States Air Force, 2610 Seventh Street, Bldg 441, Wright-Patterson AFB, OH 45433

Heather Rowsell, Cochair

Chair's Introduction—12:55

Invited Papers

1:00

5pNSb1. Enhancing hearing protection in complex noise environments with extended-wear hearing aids. Douglas S. Brungart (Walter Reed, 4401 Holly Ridge Rd., Rockville, MD 20853, dsbrungart@gmail.com), LaGuinn Sherlock (Walter Reed, Bethesda, MD), Nathaniel J. Spencer (Henry M Jackson Foundation, WPAFB, OH), and Hilary Gallagher (Air Force Res. Lab., U.S. Air Force, Wright-Patterson AFB, OH)

Conventional hearing protection devices (HPDs) work well in predictable noise environments where it is practical to put them on prior to exposure and remove them after. However, some noise environments are characterized by long periods of relative quiet that are unpredictably interrupted by brief periods of hazardous noise. These environments make it difficult to ensure that HPDs are worn when exposure to hazardous levels occurs. One possible solution is the extended-wear hearing aid (EWHA), a hearing aid that completely blocks the ear canal and can remain in the ear for months at a time. Previous research has shown that the EWHA can provide protection from impulse noise without significantly interfering with detection and localization of environmental sounds. At that time, the major drawback of the EWHA was that once it was removed it could only be re-inserted by an audiologist. This limitation has been addressed by the development of a self-insertion tool that allows users to remove and re-insert the EWHA as desired. We will discuss the potential advantages of using the EWHA as an HPD, with an emphasis on situations where the EWHA could be used to reduce noise exposure without modifying the current commercially available device.

1:20

5pNSb2. Protecting beyond the bone-conduction limit: Lessons learned developing and fielding a passive Hearing Protection Helmet. Jed C. Wilbur (Creare LLC, 16 Great Hollow Rd., Hanover, NH 03755, jcw@creare.com), Lindsay Allen, William Audette, Christian Passow, Jacob VanMalden, James Arsenault, Jonathan Hamilton (Creare LLC, Hanover, NH), Kimberly Gould, Tiffany Lei, John Farnese, Sienna Pollock (PMA-202, NAVAIR, Patuxent River, MD), and Robert Kline Schoder (Edare LLC, Lebanon, NH)

Traditional hearing protection devices focus on attenuating air-conducted sound and are limited by the bone conduction threshold (Berger, 1983). In extreme noise fields, such as aircraft maintainers operating near jet engines on aircraft carriers, the bone conducted pathway carries sufficient energy to damage the cochlea. Here, we describe the development, qualification, and field-testing of a passive Hearing Protection Helmet (HPH) designed to surpass the bone conduction limit. The HPH features a noise-isolating shell sealed to the head via a compliant edge seal. When worn with foam earplugs and tested to ANSI S12.6-2016 (Experimenter Fit), the Noise Reduction Rating (NRR) of the HPH is 38 dB and the Noise Reduction Statistic (NRS_A) (ANSI S12.68) (80% to 20% of users) is 45 to 52 dB. We also describe challenges associated with user acceptance of the HPH. User feedback led to the integration of an electronic hear-through system to restore auditory situational awareness. Users also rejected wired communication-enabled ear plugs, resulting in the integration of ear cup speakers and adding challenges associated with setting the hear-through sound pressure level limits. We conclude with anecdotes from trial deployments aboard U.S. Navy Aircraft Carriers in 2020, 2022, and 2023.

1:40

5pNSb3. Custom hearing protection devices: Are they better? JR Stefanson (USAARL, U.S. Army, Fort Novosel, AL, earl.w.stefanson.civ@health.mil), Jennifer Noetzel (USAARL, U.S. Army, Fort Novosel, AL), and Heath Jones (APPD, USAARL, Fort Rucker, AL)

When hazardous noise is unavoidable, hearing protection devices (HPDs) are used to reduce the risk of noise-induced hearing injuries. The question of which HPD is appropriate or "the best" is often a subject of debate. Appropriateness may be dictated by environmental factors, such as noise exposure type and level, mission requirements such as communication needs, compatibility with other personal protective equipment, or even personal preference. Regardless of the type used, properly fitting hearing protection is paramount to prevent or reduce hazardous noise entering the auditory system. Custom HPDs are designed to fit the specific anatomy of an individual and, when made and fit properly, may be appropriate for many applications and missions (e.g., band members, certain aviators). In addition to being tailor-made to each individual, custom HPDs may be designed to couple with communication devices, frequency or level-specific filters, and/or may incorporate electronics to provide capabilities such as amplification or reduction of ambient noise. Attenuation performance of custom HPDs coupled to a communication earplug (CEP) was measured and compared to traditional foam HPDs coupled to the CEP. This discussion will include custom HPD fabrication options, performance characteristics, benefits, drawbacks, and finally their potential real-world application.

2:00

5pNSb4. Design, development, and testing of an inflatable earplug. Anthony Dietz (Paxauris, 4101 E Fanfol Dr., Phoenix, AZ 85028, tony.dietz@paxauris.com)

Current earplugs can be categorized into the following types: roll-down foam, push-to-fit foam, premolded, custom-molded, and formable. Here, we present a new type of earplug that has the potential to be easier to insert and more comfortable than other types of earplugs. The earplug consists of a slender stem that is inserted into the ear canal, and a fluid-filled bulb that rests in the concha. The stem comprises a central core enclosed in an inflatable sheath while the bulb is formed from a bi-stable hemisphere. The insertion method for this earplug is unlike that of current earplug types. The wearer slips the stem into their ear canal, then pushes the bulb, squeezing fluid to inflate the sheath at its tip until the hemisphere pops into its stable inverted form. The inflated sheath forms a comfortable acoustic seal deep in the ear canal. To remove, the wearer deflates the sheath by pulling a tab attached to the hemisphere to pop it back to its normal state, which deflates the sheath and refills the bulb. The design of the earplug is presented, along with the results of attenuation and life tests.

2:20

5pNSb5. Evaluation of sound localization and speech recognition with a sound enhancing connection tube for non-linear earplugs. Nir Fink (Commun. Disord. Dept., Ariel Univ., 65 Ramat HaGolan st., Ariel 407763, Israel, nirfi@ariel.ac.il), Aaron Benn (Commun. Disord. Dept., Ariel Univ., Yavne, Israel), and Liron Grushko (Commun. Disord. Dept., Ariel Univ., Rehovot, Israel)

Noise-induced hearing-loss (NIHL) can occur when an individual is exposed to high-intensity acoustic stimuli. To prevent NIHL, individuals may use non-linear-earplugs (NLEPs) that provide some attenuation, substantially within the frequency range of 1–4 kHz. Wearing NLEPs can reduce speech recognition as well as sound localization. As weapon noise is mostly present in the 1–4 kHz range and is therefore partly attenuated by NLEPs, there is a downside to the reduced ability of a soldier using them to localize the source of enemy gunfire. A perforated sound-enhancing connection tube (SECT) was designed to connect to NLEPs for improving the user's ability to localize sound and speech recognition in noise while retaining the NLEPs's noise attenuation benefits of protecting one's hearing. A previous experiment examined the effect of the SECT with NLEPs on the Head Related Transfer Function (HRTF) of a head-and-torso-simulator (HATS) (manikin). The obtained HRTF demonstrated a 3–10 dB difference in the 1–4 kHz frequency range relative to the HRTF with the NLEPs alone, implying that sound stimuli in that frequency range might be more audible to the user. The presented work will demonstrate the speech recognition in noise and sound localization of participants when using the SECT with NLEPs.

2:40–3:00 Break

3:00

5pNSb6. Improving hearing protection compliance by optimizing user Mission Effectiveness. Allan Schrader (Design and Development, Lightspeed Aviation, 6135 SW Jean Rd., Lake Oswego, OR 97035, aks@lightspeedaviation.com), Brian Frost, and Kathy Zhang (Design and Development, Lightspeed Aviation, Lake Oswego, OR)

Traditionally intra-aural (ear plugs) and Circum-aural (muff) style HPD's are provided in elevated noise environments to provide additional noise protection. With proper training around insertion and wearing, these devices are used to provide compliance with safety standards for hearing exposure. It is well understood that the wearer's ACTUAL protection, in their work/mission roles, are regularly eroded by misuse due to user isolation from "wanted" or "needed" sounds: Situational awareness for safety or mission needs, and Ambient and/or needed communications Alternative technologies can be applied to *enhance* the users Mission Effectiveness. With that, they "want to" wear devices. These can be customized/personalized to fit their mission while provide appropriate protection. While the overall protection might not reach the levels "possible" with a clinical HPD application, users wear the devices properly, gaining significant and consistent overall protection. This paper will explore additional technologies available, to incorporate into Muff-type HPD's, that will *enhance user productivity* while also improving hearing protection: Active noise reduction (FB and FF), Active/personalized audio compensation and HPD effectiveness testing, Noise dosing information, and Customized alerts for preset thresholds covering hearing and environmental hazards.

3:20

5pNSb7. Method to measure the effect of acoustic leaks on active noise reduction device performance. Hilary Gallagher (Air Force Res. Lab., U.S. Air Force, 2610 Seventh St., Bldg 441, Wright-Patterson AFB, OH 45433, hilary.gallagher.1@us.af.mil) and Zachariah Ennis (Ball Aerosp. and Technologies Corp, Wright-Patterson AFB, OH)

Passive hearing protection (HP) systems are designed to reduce the level of noise at the ear. Passive HP typically excel in attenuating mid- to high-frequency noise. Active noise reduction (ANR) has the potential to provide additional attenuation of low-frequency noise. There are documented reports describing unwanted sound being generated by ANR systems if an acoustic leak is present. The objective of this effort was to develop a method to measure the performance of ANR devices with and without the introduction of an acoustic leak. A number of ANR earcups were selected for this study. Data were collected with an artificial ear and a flat base plate in a noise environment that ranged from 65–115 dB SPL. Wedges, designed in-house, created a consistent acoustic leak that ranged in size from 1–2 mm in radius. The results revealed that some ANR earcups provided high levels of protection when the earcup was well-sealed; however, when an acoustic leak was introduced, the earcups became unstable and generated noise in the earcup. This instability was not seen in all ANR devices. The outcome of these measurements solidified the need for a method to measure ANR device performance when acoustic leaks are introduced.

3:40

5pNSb8. The impact of phase on auditory situational awareness. Brian Fowler (Gentex Corp., 645 Harvey Rd., Ste. 102, Manchester, NH 03103, bfowler@gentexcorp.com)

A person's situational awareness is the ability to understand and correctly interpret the world as it exists around them. One component of auditory situational awareness is sound localization, or the ability to identify where a sound is in space relative to the listener. Different aural cues contribute to a listener's sound localization: the difference in the time it takes for a sound to reach one ear compared to the other (interaural time difference, or ITD) and the difference in the level of the sound at each ear (interaural intensity difference, or IID). The relative contribution of these interaural differences changes with frequency regime, with ITD/IPD being more dominant at low frequencies and IID being more dominant at high frequencies. These cues, as well as a spectral coloration dependent upon the shape of the listener's pinna and concha bowl, contribute to what is called the head-related transfer function (HRTF). This study examines the impact of a phase inversion in one ear on a person's auditory situational awareness. Testing was conducted in accordance with ASA/ANSI S3.71-2019, Method 3 to measure the impact on response time, using full spectrum stimuli as well as band-limited stimuli to focus testing on each aural cue.

Contributed Paper

4:00

5pNSb9. Measuring the effect of electronic hearing protection devices on behavioral detection thresholds. Eric R. Thompson (Air Force Res. Lab, 2610 7th St., B441, Wright-Patterson AFB, OH 45433, eric.thompson.28@us.af.mil), Hilary Gallagher (Air Force Res. Lab, Wright-Patterson AFB, OH), and Nina Pryor (Air Force Res. Lab, Dayton, OH)

Electronic hearing protection devices can provide amplification for quiet ambient sounds, while still providing protection against loud sounds. These can be important in environments with sporadic and unpredictable hazardous noise, but also with a need for maintaining awareness of ambient sounds, or for conducting face-to-face communication. This study's aim was to understand what impact electronic hearing protection devices have on auditory detection thresholds. Human subjects were used to measure thresholds for narrowband noise bursts without hearing protection, with electronic hearing protectors powered off, and with electronic hearing protectors powered on with several gain settings. In addition, electroacoustic measurements were made of the input/output functions and electronic noise floors of the hearing protectors at the same gain settings. The data showed that thresholds in quiet with electronic hearing protectors were higher than without, regardless of the device amplification. This could be attributed to the hearing protector electronic noise limiting performance. Thresholds measured with ambient masking noise were similar across open ear and device conditions, reflecting that the signal-to-noise ratio for ambient sounds does not change with amplification. Listeners with a simulated conductive hearing loss had lower thresholds with an electronic hearing protector than without, suggesting a benefit of the device amplification.

Invited Paper

4:20

5pNSb10. A new method for measurement of impulsive peak level attenuation of hearing protectors. Cameron Fackler (Personal Safety Div., 3M, Indianapolis, IN), Dan Gauger (Bose Corp. (retired), Framingham, MA), Jeff G. Schmitt (VIacoustics, 2512 Star Grass Circle, Austin, TX 78745, jeffs@viacoustics.com), William Murphy (Stephenson and Stephenson Res. and Consulting, Batavia, OH), and Elliott Berger (Berger Acoust. Consulting, Indianapolis, IN)

ASA/ANSI S12.42-2010 contains the first standardized test method for measuring the performance of hearing protection devices (HPDs) with high-level impulsive noise. This standard defines the impulsive peak insertion loss (IPIL) as a time-domain metric that quantifies the reduction in peak sound pressure level provided by an HPD for impulsive noises. However, IPIL as defined in S12.42-2010 is dependent on the spectrum of the impulsive noise source used for measurements of HPDs. Recent studies of HPDs with impulsive noise have led to the investigation of frequency-domain metrics and calculation methods. Using frequency-domain calculations allows the impulsive peak level attenuation to be computed for a wide range of impulsive noises, not only those used for the measurement. Consequently, significant revisions to S12.42 are under consideration and a new standard, ISO 4869-7, is being developed in parallel with the ANSI revision. This presentation highlights key proposals and substantive changes including: measurement and calculation of the frequency-dependent impulsive insertion loss (IIL), incorporation of frequency-domain aspects such as real-ear attenuation at threshold (REAT) limits to attenuation, and a new impulsive peak level attenuation metric (IPLA) to be calculated from the REAT-bounded IIL.

4:40

5pNSb11. High level sound transmission through cadaver human ears—On the influence of bone conduction and hearing protective devices. Nathaniel T. Greene (Otolaryngol., Univ. of Colorado School of Medicine, Aurora, CO, nathaniel.t.greene@gmail.com), Juanantonio Ruiz (Otolaryngol., Univ. of Colorado School of Medicine, Aurora, CO), Ted Argo (Appl. Res. Assoc., Inc., Littleton, CO), Andrew D. Brown (Speech and Hearing Sci., Univ. of Washington, Seattle, WA), and David A. Anderson (Univ. of Minnesota Duluth, Duluth, MN)

High level sound exposure can cause substantial injury to the auditory system, motivating efforts to predict and prevent this injury. Measurement techniques using acoustic manikins are effective for low and moderate sound levels, but nonlinear effects in the middle ear and alternate sound transmission pathways to the inner ear limit their utility at higher sound pressure levels. Here, we describe results from a series of measurements made in cadaveric human ears conducted in our laboratory over the last several years. We quantified sound transmission to the inner ear by measuring the difference in sound pressure level across the cochlear partition near the base of the cochlea, which drives basilar membrane motion. We describe measurements quantifying sound transmission through the human middle ear, which is limited by suspensory ligaments for sounds above approximately 130 dB SPL, resulting in nonlinear (harmonic) distortion of sounds in the cochlea at higher sound pressure levels. Similarly, we describe observations of sound transmission to the cochlea via bone conduction from vibratory transducers, as well as high level sound and impulse noise/blast exposures. The implications of these measurements for hearing loss predictions, effectiveness of hearing protective devices, and injury mechanisms are discussed.

5:00

5pNSb12. Absorptive noise barrier development. Harvey Law (Mech. Eng. & Mater. Eng., Monash Univ., Bldg. 3 (Rear), 621 Whitehorse Rd., Mitcham, Victoria 3132, Australia, harvey.law@megasorber.com), Jenny Law (Mater. Eng., Monash Univ., Mitcham, Victoria, Australia), and Marek Kierzkowski (Member of Australian Acoust. Society, Mitcham, Victoria, Australia)

This paper reveals the development of an absorptive, self-supporting noise barrier panel for external applications. Designed for use as a temporary external noise barrier for construction sites and roadside noise barriers. The absorptive noise barrier panel is a module base design that is easy to assemble, disassemble, and re-deploy. The barrier panel utilizes recycled plastic materials, such as milk bottles and other soft plastics, making it one of the most environmentally friendly products. The barrier is manufactured in one process via a unique rotation molding method. It is weather resistant, aesthetically appealing, and offers great design flexibility. Standard size panels are 100 mm thick, by 2000 m wide x 500 mm high. The acoustic design is modeled to achieve NRC 0.85 at 100 mm thick based on an advanced air-flow resistive layer technique combined with an air cavity. The actual sound absorption measurement results are well in line with the predicted modeling. Such a case study potentially uncovers a whole new range of innovative absorptive noise barriers. Further studies and investigations are also recommended.

Session 5pSC

Speech Communication: Voice Therapy: Science and Clinical Efficacy II

Zhaoyan Zhang, Cochair
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Catherine Madill, Cochair
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Contributed Papers

1:00

5pSC1. Disordered voice recognition benchmark. Rijul Gupta (Computing and Audio Res. Lab., School of Elec. and Information Eng., The Univ. of Sydney, New South Wales 2006, Sydney, Australia, rijul.gupta@sydney.edu.au), Dhanshree R. Gunjawate, Duy D. Nguyen, Catherine Madill (Voice Res. Lab., Sydney School of Health Sci., Faculty of Medicine and Health, Univ. of Sydney, Camperdown, New South Wales, Australia), and Craig Jin (Computing and Audio Res. Lab., School of Elec. and Information Eng., The Univ. of Sydney, New South Wales, Sydney, Australia)

Over the past decade, the application of machine learning to voice disorder recognition has shown promising results. However, several areas of the discipline that impact recognition accuracy remain understudied. These areas include the impact of different vocal tasks, patient demographics, and symptom details. Additionally, the hyperparameters associated with voice features cannot always be easily explored in terms of recognition accuracy. Furthermore, preliminary research from an ongoing scoping review reveals differences in diagnostic terms used within the field. To address the issues raised above, we introduce the Disordered Voice Recognition (DiVR) Benchmark that includes a variety of vocal tasks, some patient history data, and a hierarchical organization of diagnostic terms derived from a consensus of multiple clinical experts. The DiVR Benchmark includes detailed spreadsheets outlining the variety of datasets, features, and machine learning algorithms described in the literature. In this work, we present an ensemble of baseline models and compare recognition results with some of the most commonly employed algorithms. We also describe how the classification accuracy varies with data availability, the granularity of the disorder classification label, and the vocal task employed.

1:20

5pSC2. Voice quality alteration and its potential implications for voice therapy. Isabel S. Schiller (Work and Eng. Psych., Inst. of Psych., RWTH Aachen Univ., Jaegerstrasse 17-19, Aachen 52066, Germany, isabel.schiller@psych.rwth-aachen.de), Karolin Krüger, Firdous Bin Ismail (Digital Signal Processing and System Theory, Dept. of Elec. and Information Eng., Christian-Albrechts-Univ., Kiel, Germany), Sabine J Schlittmeier (Work and Eng. Psych., Inst. of Psych., RWTH Aachen Univ., Aachen, Germany), and Gerhard Schmidt (Digital Signal Processing and System Theory, Dept. of Elec. and Information Eng., Christian-Albrechts-Univ., Schmidt, Germany)

Modulating a speaker's auditory feedback is a valuable technique for investigating vocal motor control. In auditory feedback alteration (AFA) experiments, participants receive real-time perturbed feedback of their voice through earphones while producing vocalizations into a microphone. Previous studies primarily focused on pitch and amplitude alterations, which typically result in compensatory vocal responses. The purpose of this study is to present a voice resynthesis system, called *VQ-Synth*, designed for real-time auditory feedback alteration of the speaker's voice quality in terms of hoarseness. While an initial version of *VQ-Synth* was implemented using Matlab, the system has now been enhanced in a real-time framework written in ANSI-C to achieve effective voice quality resynthesis with minimal delay. Additionally, a graphical user interface (GUI) was implemented to guide participants through the experiment. This work presents the technical architecture of *VQ-Synth* and its intended application in AFA experiments involving both healthy and dysphonic speakers. Furthermore, we discuss the potential implications of the system in the context of auditory cognition and voice therapy. We hope that the novel capabilities of *VQ-Synth* open up exciting possibilities for investigating how alterations in voice quality affect vocal motor control and how this knowledge can be applied in therapeutic settings.

Invited Papers

1:40

5pSC3. The Estill Voice Model: A paradigm for voice training and treatment. Kimberly Steinhauer (Estill Voice Int., 245 Merion Dr., Pittsburgh, PA 15228, ksteinhauer@estillvoice.com)

For over 40 years, the Estill Voice Model (EVM) has defined voice quality according to movement of anatomy and physiology. EVM addresses the daunting degrees of freedom issue in voice motor control by isolating *Craft* of voice production from *Artistry* and *Performance Metaphysics*. The EVM proposes an integrated implicit-explicit approach for voice motor learning that flows through all

training and therapy protocols. Implicit instructions include auditory-perceptual prompts (e.g., quack like a duck to produce "twang") and explicit prompts train physiologic conditions of the vocal anatomy correlated with the voice quality (e.g., narrow your aryepiglottic sphincter to produce "twang"). Estill exercises address power, source, and filter properties of voice production, and include narrowing the aryepiglottic sphincter for "ring" in opera and belt and for increased power in hypofunctional voices, and varying vocal fold mass for register shifts and optimizing contact for hyperfunctional voices. Patients learn to feel, see, and hear the voice via multiple feedback modes including hand gestures, magnitude estimation of bodily kinesthetic effort, visual acoustic cues in real-time spectral analysis programs. This presentation will highlight objective measurement science and clinical evidence for using Estill exercises in treatment for all voices, from the novice speaker to the expert performer.

2:00

5pSC4. Duration of vocal effects due to water resistance therapy. Matthias Echternach (Phoniatrics and Pediatric Audiol., LMU Univ. Hospital Munich, Marchioninistr. 15, 2, Munich 81377, Germany, matthias.echternach@med.uni-muenchen.de), Marie Köberlein (Phoniatrics and Pediatric Audiol., LMU Univ. Hospital Munich, Munich, Germany), Marco Guzman (Universidad de los Andes, Santiago, Chile), Anne Maria Laukkanen (Tampere Univ., Tampere, Finland), Bernhard Richter (Freiburg Univ., Freiburg, Germany), and Marie Köberlein (Phoniatrics and Pediatric Audiol., LMU Univ. Hospital Munich, Munich, Germany)

There are only few data about effects of semi-occluded vocal tract exercises on vocal fold oscillation patterns *in vivo*. Furthermore, it has not yet been clarified for how long these effects last. Material: 16 subjects (8 non-pathological and 8 with vocal fold mass lesion) were asked to sustain a phonation on the vowel /i/. During phonation, the subjects were simultaneously recorded with transnasal high speed videoendoscopy (HSV, 20.000 fps), electroglottography, and audio signals. After that, these subjects performed a Water Resistance Therapy (WRT) for 10 min (tube 30 cm, 9 mm diameter, 5 cm below surface). Repeated measurements of sustained phonation were performed 0, 10, and 30 min after exercising. The data showed strong inter-individual differences concerning the courses of the different measures after WRT. For non-pathological voices, directly after WRT there was a lowering of the Glottal-Area-Waveform derived Period-Perturbation -Quotient, a lowering of the closing quotient. For pathological voices, however, such a decrease was not observable. Five minutes after the exercise such effect was not observable any more. WRT showed strong inter-individual effects in the courses of the evaluated measures. General tendencies of some measures directly after the intervention showed a brief effect of only a few minutes.

Contributed Papers

2:20

5pSC5. The effect of vocal tract semi-occlusion on the voice source and implications for voice therapy. Zhaoyan Zhang (Dept. of Head and Neck Surgery, Univ. of California, Los Angeles, 1000 Veteran Ave., Ste. 31-11, Los Angeles, CA 90095, zyzhang@ucla.edu)

Semi-occluded vocal tract exercises are often used in voice therapy to improve voice production. It is generally assumed that by enhancing source-filter interaction, such vocal exercises improve vocal efficiency while minimizing vocal fold collision. The goal of this computational study is to quantify such effects of source–filter interaction on the voice source. Voice simulations were performed with vocal tract constriction at various locations (false vocal folds, aryepiglottic folds, pharynx, oral cavity, and lips). The results showed that effects of vocal tract constriction on the voice source were generally small except for conditions of extreme constriction, at which constrictions at any location along the vocal tract decreased the mean and peak-to-peak amplitude of the glottal flow waveform. In general, source-filter interaction that reduced vocal fold contact pressure also decreased sound pressure level, thus making it impractical when targeting a desired output vocal intensity. While source–filter interaction is often assumed to improve vocal efficiency and economy, our results indicate that such improvements are mainly due to improvement in the filter rather than the voice source at the glottis. It is possible that speakers make additional laryngeal adjustments during vocal exercises, which may also contribute to the therapeutic benefits of such exercises.

2:40–3:00 Break

3:00

5pSC6. Vocal intensity control challenges in bilateral cochlear implant users. Simin Soleimanifar (Speech and Hearing Sci., Univ. of Illinois at Urbana Champaign, Champaign, IL) and Justin Aronoff (Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, 901 S 6th St., Champaign, IL 61820, jaronoff@illinois.edu)

Bilateral cochlear implantation offers benefits but poses challenges for vocal intensity control. This study explored how differences in loudness growth between ears contribute to poor vocal intensity control in BiCI users. Experiment 1 tested 13 BiCI users with sustained vowel vocalization using both cochlear implant devices together and each one separately to measure their ability to control long-term vocal intensity variation (vAm). To determine if deficits in vAm when using both ears reflected mismatched loudness growth across ears, Experiment 2 examined loudness growth from each ear for a subset of participants and Experiment 3 manipulated the loudness growth function of the processor to change the shape of loudness growth function while measuring vAm for a subset of participants. Experiment 1 showed decreased vocal intensity control with BiCI devices compared to unilateral CIs. Experiment 2 revealed different loudness growth perceptions between ears for most BiCI users, possibly explaining their poor vocal performance with both implants. Experiment 3 demonstrated that changing loudness growth functions of the processor altered the ability to control vAm, indicating that changes in loudness growth affect vAm. The results suggest unmatched loudness growth perception between ears likely contributes to poor vocal intensity control in BiCI users.

Invited Paper

3:20

5pSC7. Functional Electrical Stimulation (FES)—Therapeutical potentials for presbyphonia. Stefan Kniesburges (Phoniatrics & Pediatric Audiol., Univ. Hospital Erlangen, Waldstrasse 1, Erlangen, Bavaria 91058, Germany, stefan.kniesburges@uk-erlangen.de), Bernhard Jakubass (Phoniatrics & Pediatric Audiol., Univ. Hospital Erlangen, Erlangen, Bavaria, Germany), Claus Gerstenberger, Andrijana Kirsch (ENT University Hospital, Medical University of Graz, Div. of Phoniatrics, Graz, Austria), Gregor Peters, Marion Semmler (Phoniatrics & Pediatric Audiol., Univ. Hospital Erlangen, Erlangen, Bavaria, Germany), Markus Gugatschka (ENT University Hospital, Medical University of Graz, Div. of Phoniatrics, Graz, Austria), and Michael Dollinger (ENT, Univ. Hospital Erlangen, Erlangen, Germany)

Presbyphonia, an age-related decrease in voice quality is a consequence of vocal muscle atrophy. Functional Electrical Stimulation (FES) was introduced as a potential treatment to increase the muscle volume. In this study, we analyzed the effects of FES using *ex vivo* sheep larynx models. To stimulate the thyroarytenoid muscle, electrodes were implanted at the recurrent laryngeal nerve of 12 sheep (10 y). Stimulation was applied to six sheep for 9 weeks. Afterwards, the excised larynges were fixated in a mechanical setup and analyzed. For different elongations of the VFs, the acoustic signal, the subglottal pressure, and the VF motion were recorded simultaneously. The postprocessing was performed with the Glottis Analysis Tools (GAT) software and SPSS. The results show that the vibration characteristics and sound quality improved for the stimulated larynges. The vibration showed a higher degree of glottis closure (GGI) and symmetry (PAI and ASI), as well as a faster glottis closure (MADR) achieved by FES. Consequently, the sound signal exhibited a larger number of tonal sound components (CPP). Furthermore, a higher correlation between acoustic and subglottal pressure was found indicating an enhanced coupling between sub- and supraglottal regions. The results suggest that FES could be a possible therapy for presbyphonia.

Contributed Papers

3:40

5pSC8. Anterior-posterior characterization of intraglottal flow fields in excised canine larynges. Jacob Michaud-Dorko (Dept. of Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH), Charles Farbos de Luzan (Dept. of Otolaryngology-Head and Neck Surgery, Univ. of Cincinnati, Cincinnati, OH), Ephraim Gutmark (Dept. of Aerosp. Eng., Univ. of Cincinnati, Cincinnati, OH), and Liran Oren (Dept. of Otolaryngology-Head and Neck Surgery, Univ. of Cincinnati, 231 Albert Sabin Way, Cincinnati, OH 45229, orenl@ucmail.uc.edu)

It has been espoused that intraglottal flow separation vortices (FSVs) play a critical role in the rapid closure of the vocal fold vibrations by acting as an additional pulling force that affects the maximum flow declination rate (MFDR). Clinically, MFDR is important because studies showed it highly correlates with loudness, intelligibility, and vocal efficiency. Previous studies that aimed to characterize the influence of FSVs were focused on their role in the mid-coronal plane. In the current study, particle imaging velocimetry was used to measure the intraglottal flow, specifically the FSVs, in the anterior–posterior aspects of the glottis during the closing phase of the vocal fold vibrations. The results show that the FSVs are extending along the length of the vocal fold in the anterior–posterior direction and that there is a direct relation between the size of FSVs and the magnitudes of the glottal divergence angles. The results also show that the centroid of FSVs occurs towards the anterior direction, where larger divergence angles are observed. These findings suggest that the FSVs in the anterior aspect have a more significant effect on the aerodynamic force acting on the glottal wall than the FSVs in the posterior aspect.

4:00

5pSC9. A comparative analysis of intraglottal geometry and velocity flow fields: Excised human, canine, and synthetic vocal fold models. Jacob Michaud-Dorko (Dept. of Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH), Charles Farbos de Luzan (Head & Neck Surgery - Otolaryngol., Univ. of Cincinnati, 231 Albert Sabin Way, MSB, Cincinnati, OH 45229, farboscs@ucmail.uc.edu), Greg Dion (Head & Neck Surgery - Otolaryngol., Univ. of Cincinnati, Cincinnati, OH), Ephraim Gutmark (Aerosp. Eng., Univ. of Cincinnati, Cincinnati, OH), and Liran Oren (Head & Neck Surgery - Otolaryngol., Univ. of Cincinnati, Cincinnati, OH)

Synthetic vocal fold models imitate the characteristics of human vocal folds and offer advantages over tissue models, such as accessibility and prolonged lifespan, all while producing comparable vibration frequencies to those in humans. Nonetheless, due to their simplified design, they may not fully capture the intricate behavior observed in tissue models like the canine larynx model. Canine larynges, like human's, exhibit a vertical mucosal wave, which yields a phase delay between the inferior and superior edges of the vocal folds free margin. Consequently, a divergent glottis forms during closing, producing intraglottal flow separation vortices (FSVs) that increase voice loudness and intelligibility. To bridge the knowledge gap between silicone and tissue vocal fold models, we compared the intraglottal geometry and velocity flow fields during phonation of these three models. At low subglottal pressures, the synthetic model displayed small divergence angles and no flow separation, where both tissues models exhibited FSVs. At higher pressures, FSVs observed in tissue models were correlated with increased glottal flow waveform skewing and higher MFDR values. These findings highlight why the excised canine larynx model should be the preferred choice over synthetic vocal fold models for studying the aerodynamics and fluid-structure interaction during phonation.

4:20

5pSC10. Comparing direct and indirect glottal flow waveform measurements in excised canine larynges and synthetic vocal fold models. Jacob Michaud-Dorko (Dept. of Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH), Charles Farbos de Luzan (Head & Neck Surgery - Otolaryngol., Univ. of Cincinnati, Cincinnati, OH), Ephraim Gutmark (Dept. of Aerosp. Eng., Univ. of Cincinnati, Cincinnati, OH), and Liran Oren (Otolaryngol., Univ. of Cincinnati, 231 Albert Sabin Way, Cincinnati, OH 45229, orenl@ucmail.uc.edu)

Despite being considered the gold standard technique for delineating the flow glottogram during phonation, the inverse filtering technique has never been validated against actual volume flow measurements. This signal-processing technique uses the speech signal with vocal tract estimates to calculate the glottal flow waveform. The lack of validation stems from the difficulty of obtaining *in-vivo* flow measurements at the glottal exit. The current study aims to assess the accuracy of the inverse filtering technique by comparing its flow glottograms with direct, simultaneous volume flow measurement. A vocal tract with varied constrictions that mimic its shape according to specific vowels was connected to either an excised canine larynx or a synthetic vocal fold model. Direct volume flow measurements at the glottal exit were done using tomographic particle image velocimetry. Indirect measurements were obtained by fitting the known Rothenberg mask to the vocal tract and estimating the glottal flow waveform using the Glottal Enterprise system. Preliminary results indicate that the accuracy of the inverse filtering technique relies on the degree of vocal tract constriction. The flow glottogram aligns well with front, lax, and unrounded lip vowels, but its accuracy diminishes with tense and round vowels, particularly noticeable with the excised canine larynx.

Session 5pUW

Underwater Acoustics and Underwater Acoustics: Topics in Underwater Sound

Nicholas C. Durofchalk, Chair
Physics, Naval Postgraduate School, 1 University Circle, Monterey, CA 93943, Spanagel Hall, Room 141, Monterey, CA 93943

Contributed Papers

1:00

5pUW1. Experiments with coated surfaces to generate sound waves in water. Kevin Hostombe (Mechatronics, Helmut-Schmidt-Univ., Hamburg, Holstenhofweg 85, Hamburg 22043, Germany, hostombk@hsu-hh.de), Tom Avsic (thyssenkrupp MarineSystems GmbH, Kiel, Germany), and Delf Sachau (Mechatronics, Helmut-Schmidt-Univ., Hamburg, Hamburg, Germany)

Sources for underwater sound have a wide range of applications, such as active sonar, navigation, and underwater communication. Particularly, sources with large dimensions l, such as a coating on a boat structure, can be suitable for generating low-frequency underwater sound. In this context, low-frequency refers to wavelength λ in water that fall within the range of $40\ l > \lambda > 1{,}5\ l$. In this study, a sample of an active surface for generating underwater sound is created by attaching an array of 18 circular piezoelectric actuators to a glass-fiber reinforced plastic (GRP) plate. The array and the plate are coated with a potting compound that possesses the same acoustic properties as water. This coating physically separates the actuators from the water. The actuators can be individually controlled to analyze different configurations. The radiation characteristics of the active surface are investigated in an underwater test range with free field conditions, specifically for low-frequencies. Additionally, the interaction among the actuators and between the actuators and the GRP-plate are analyzed.

1:20

5pUW2. Modeling sea surface noise in the presence of seamounts and ocean fronts using Nx2D and 3-D methods. Nicholas C. Durofchalk (Phys., Naval Postgrad. School, 1 University Circle, Spanagel Hall, Rm. 141, Monterey, CA 93943, nicholas.durofchalk@nps.edu), Adrian K. Doran, and Jahn R. Torres (US Naval Undersea Warface Ctr., Newport, RI)

This study focuses on modeling the ambient noise observed along a vertical line array in the presence of distributed surface sources and extreme bathymetric and oceanographic features. The distributed surface sources considered in this study include wind-induced noise, while the extreme bathymetric and oceanographic features comprise seamounts such as Atlantis II and the ocean front caused by the Gulf Stream. Directional vertical noise levels are modeled by computing the Green's Function between the array elements and the distributed surface sources, estimating the average cross spectral density matrix, and calculating the plane wave response [1]. Green's functions are computed using the Bellhop3-D ray tracing model [2] in both an Nx2D and full 3-D environment, and the resulting directional noise levels are compared. The results of this study will provide insights into the accuracy and effectiveness of the modeling methods used to predict ambient noise levels in such challenging environments, with particular emphasis on the contribution of seamounts to ambient noise. The implications of these results can be significant for the design and operation of underwater acoustic systems in similar environments.

1:40

5pUW3. Abstract withdrawn.

2:00

5pUW4. A practical method for assessing and managing the risk of noise induced hearing loss for divers in high-noise underwater environments. Benjamin C. Lawrence (Marshall Day Acoust., 84 Symonds St., Grafton, Auckland 1010, New Zealand, ben.lawrence@marshallday.co.nz) and Reuben J. Jelleyman (Marshall Day Acoust., Grafton, New Zealand)

The purpose of this study was to develop a method to assessing and managing the risk of noise induced hearing loss (NIHL) to commercial divers operating within or nearby sites with high underwater sound levels. Our method involves using a transfer function to convert underwater sound levels to airborne sound levels at the diver's ear inside a helmet, which can then be assessed against the 85 dB $L_{Aeq\ (8\ hour)}$ exposure limit for NIHL. We measured this transfer function for a standard commercial diving helmet (Kirby Morgan 37) during nearby impact piling using a Soundtrap hydrophone in the water and Brüel and Kjær sound level meter with a type-1 microphone at the ear position. We then used the results to calculate a time limit for a diver to be in the water before the in-ear sound levels exceed the NIHL exposure limit.

2:20

5pUW5. Effect of sound speed profile on the structure of acoustic pulse and the convergence zone in deep water. Shuanglin Wu (State Key Lab. of Acoust., The Inst. of Acoust. of the Chinese Acad. of Sci., No. 21 North 4th Ring Rd., Haidian District, 100190 Beijing, China, wushuanglin@mail.ioa.ac.cn), Jixing Qin (State Key Lab. of Acoust., Inst. of Acoust., The Inst. of Acoust. of the Chinese Acad. of Sci., Beijing, China), and Zhenglin Li (School of Ocean Eng. and Technol., Sun Yat-sen Univ., Zhuhai, Guangdong, China)

Through the experimental data collected in different seasons with the propagation conditions change in the East Indian Ocean (EIO) and the South China Sea (SCS), we observe that the structure of the sound speed profiles (SSPs) in these two seas area is very different and has a significant impact on the CZs and time arrival structure of acoustic pulses. In the EIO environment, sound energy transmitting along the sound channel axis (SCA) is relatively large, and the corresponding signals arrive first, whereas signals propagating off the SCA arrive late, which is totally different with the characteristics of the waveform in the SCS. At the same time, the range of the first CZ in the EIO is 7–8 km farther than that in the SCS, and the width of the CZs in the EIO is about 2–3 km narrower when the water depths in the two experimental areas are similar. Combined with the theoretical model, the influence mechanism of SSP on the time arrival structure and CZ propagation are theoretically analyzed, which well explains the phenomenon observed in the two experiments. The results can provide a reference for the application of communication sonar in the deep-sea remote environment.

2:40–3:00 Break

3:00

5pUW6. Design of performance-weighted blended mode filters for underwater experiments. Kathleen E. Wage (George Mason Univ., 4400 University Dr., MSN 1G5, Fairfax, VA 22030, kwage@gmu.edu) and Bhargabi Chakrabarti (George Mason Univ., Fairfax, VA)

Normal mode signals are valuable in ocean acoustic tomography and source localization, assuming that they can be separated using spatial or temporal filtering. Common spatial mode filters include the matched filter (MF) [Ferris, JASA, 1972] and the pseudo-inverse (PI) filter [Tindle *et al.*, JASA, 1978]. Choosing a filter and its parameters, e.g., the PI filter rank, requires knowledge that is not readily available. For instance, it is unlikely that the levels of ambient noise or interference from other modes is known a priori. To address this problem, Chakrabarti and Wage [IEEE Oceans, 2021, 2022] developed a Performance-Weighted Blended mode filtering algorithm that adapts quickly to changing conditions and achieves better performance than a single fixed mode filter. The PWB filter blends a set of PI filters of varying ranks, including a rank-1 PI filter (MF). A PI filter can suppress interference from neighboring modes but is more vulnerable to noise than the MF, which has the maximum white noise gain. This talk explores practical aspects of the PWB implementation, including how to select the set of filters to blend given realistic underwater noise conditions. We will show experimental and simulated data to illustrate performance in different environments. [Work supported by ONR.]

3:20

5pUW7. Understanding and applications of striation-based beamforming in underwater acoustics. Changpeng Liu (State Key Lab. of Acoust., Inst. of Acoust., Chinese Acad. of Sci., No. 21 North Fourth Ring West Rd., Haidian District, Beijing, Beijing 100190, China, liuchangpeng@mail.ioa.ac.cn), Shihong Zhou, and Yubo Qi (State Key Lab. of Acoust., Inst. of Acoust., Chinese Acad. of Sci., Beijing, China)

Utilizing the shallow water acoustic interference phenomenon in the range-frequency domain, striation-based beamforming (SBF) was proposed to estimate waveguide invariant and enhance active sonar output. In the past decade, SBF has also been applied to enhance LOFARgrams to improve tracking performance. In this paper, the correlation theory is used to understand SBF, and some further applications of SBF in underwater acoustic problems are also discussed. It shows that SBF essentially selects the optimal signal components with the best correlation. During the process, the correlation of array signals is improved and the ability to converge signals arriving via different paths is achieved to ensure array gain. The level of gain improvement is related to the direction of striation-based processing. Hence, a cost function can be constructed to facilitate the implementations of SBF in tasks such as striation slope estimation and source range estimation. Additionally, the SBF technique can be further applied to estimate the Green's function by introducing the cross-correlation preprocessing. Utilizing the mathematical connection between SBF and CBF, SBF can also be naturally extended to other beamforming algorithms and improve their performance, such as the subspace algorithms. The above is explained using simulation and experimental results.

3:40

5pUW8. Investigations of the acoustic-field autoproducts in three dimensions. Leslie V. Arciniega (Naval Architecture and Marine Eng., Univ. of Michigan, Ann Arbor, MI) and David R. Dowling (Naval Architecture and Marine Eng., Univ. of Michigan, 2600 Draper Dr., 215 NAME Bldg., Ann Arbor, MI 48109, drd@umich.edu)

Acoustic remote sensing tasks, such as remote unknown source localization, are commonly completed via signal processing schemes that are implemented in the bandwidth of the acoustic recordings. Interestingly, many such schemes can also be implemented at frequencies outside of the bandwidth of the recorded field—despite their absence from the recordings—by using the frequency-difference and frequency-sum autoproducts [Worthmann and Dowling, J. Acoust. Soc. Am., **141** (2017) 4579–4590]. Recent acoustic-field autoproduct studies have involved beamforming and source localization in shallow- and deep-ocean environments, and have addressed the impacts of refraction, shadow zones, rough surface scattering, and noise. These previous efforts primarily explored the properties of the autoproducts when the sound field primarily varies in two spatial dimensions. This presentation provides the results of a study of the autoproducts formed from the simple but fully three-dimensional acoustic fields arising from an isolated source placed in a uniform-sound-speed environment bounded by reflecting surfaces that are not parallel. This study indicates the extent to which the autoproduct properties associated with two dimensional acoustic-field variations extend to three dimensional acoustic-field variations. The results from theory and simulations are presented, and from laboratory water-tank measurements if time allows.

4:00

5pUW9. Method of acoustic separation on the surface of cylindrical baffle. Lihong Zhang (CSSC System of Eng. Res. Inst., No.1 Fengxian Eastern Rd., Haidian District, Beijing 100094, China, mszhanglihong@126.com)

This paper suggests a statistically optimal near-field acoustic holography method based on the combination of plane waves and cylindrical waves, which is used for underwater cylindrical baffles. The method addresses the issue that the extraction of scattered acoustic waves from the surface of underwater cylindrical baffles requires too many measurement points in practical applications: separation of incident and dispersed sound fields and near-field measurements of sound fields at baffle surfaces.The underwater detection sound source is situated in the scatterer's far-field region, so in the actual application scenario for the surface acoustic scattering separation of the cylindrical baffle, the plane wave expansion is used to represent the surface incident acoustic wave and the cylindrical wave expansion is used to describe the scattered acoustic wave on the baffle surface. As a result, fewer expansion orders are required to describe the entire sound field on the baffle's surface. It is verified by simulation that the method is valid and efficient for near-field total sound field measurement and dispersed sound field separation of roughly cylindrical targets under underwater far-field incidence situations because the measurement holographic surface of the surface permits unlimited geometries.

4:20

5pUW10. Enhancement of Green's functions based on striation pattern. Daehwan Kim (Korea Maritime and Ocean Univ., Busan, Korea (the Republic of), oceankim823@gmail.com), Donghyeon Kim, Gihoon Byun, J. S. Kim (Korea Maritime and Ocean Univ., Busan, Korea (the Republic of)), and Heechun Song (Scripps Inst. of Oceanogr., Univ. of California, San Diego, San Diego, CA)

Enhancement of Green's function is accomplished from the striation pattern obtained from data. The waveguide invariant theory allows a Green's function observed at one location to be extrapolated into adjacent ranges [Song and Byun, J. Acoust. Soc. Am. **147**, 2150–2158 (2020)]. Conversely, Green's functions observed at adjacent ranges can converge towards a central location for coherent combination to improve the signal-to-noise ratio (SNR). This can be accomplished from the frequency shift in the striation pattern without prior range information of each Green's functions. We demonstrate a significant improvement of SNR in the broadband Green's function from a moving ship in shallow water.

4:40

5pUW11. A method to improve the update rate of underwater acoustic positioning system. Youngchol Choi (Korea Res. Inst. of Ships and Ocean Eng., 32 1312 Beon-gil, Yuseong-daero, Yuseong-gu, Daejeon 34103, Korea (the Republic of), ycchoi@kriso.re.kr), Sea-Moon Kim, Ara Cho, and Jong-Won Park (Korea Res. Inst. of Ships and Ocean Eng. Daejeon, Korea (the Republic of))

Underwater positioning system obtains relative position between a transponder and a transceiver by measuring round trip time (RTT) and angle of arrival. The position update rate (PUR) cannot be greater than 1/RTT, and when one transceiver locates multiple transponders, the PUR becomes smaller. This paper proposes a method to increase the PUR. The proposed method minimizes "idle time" by allowing the transceiver to send additional pings without packet transmission-reception collisions, rather than waiting for the response to the previous ping sent to the transponder. It is essential that the reception (transmission) of the ping and the transmission (reception) of the response should not be overlapped in the transponder (transceiver). In order to avoid these collisions, the ping transmission time is adaptively determined by considering the bound of packet reception times drawn by ping duration (PD), response signal duration (RD), and the maximum relative speed. In addition, the transceiver assigns a delay before sending the response signal upon receiving each ping. These adaptive features effectively adapt to changes due to mobility. We derive a theoretical upper bound of the PUR and verify this bound by computer simulations. [Work supported by KIMST funded by the Agency of Korea Coast Guard (KIMST-20210547).]

5:00

5pUW12. Accurate prediction of signal waveform in the convergence zone of the South China Sea based on frequency shift compensation.
Fujin Yang (Key Lab. of Underwater Acoust. Environment, Inst. of Acoust., Chinese Acad. of Sci., No.1921 North Fourth Ring West Rd., Haidian District, Beijing, Beijing 100190, China, fujin.yang@foxmail.com), Yuhuai Ni (Office of Military Marine Environment Construction of Navy General Staff, Beijing, China), Tao Hu, and Licheng Lu (Key Lab. of Underwater Acoust. Environment, Inst. of Acoust., Chinese Acad. of Sci., Beijing, China)

In a deep-sea environment in the South China Sea, a towed sound source is near the surface, and the acoustic signal is received by the vertical array of submersible buoys. The correlation values between the predicted and the measured signal waveform of the first two convergence zones (CZs) are poor, both below 0.5. The sea condition of the test area is good, the hydrological environment changes little in the horizontal direction. The analysis shows that when the underwater acoustic waveform is predicted, due to the relative motion of the transmitting and receiving elements, the channel dispersion caused by the Doppler effect leads to signal distortion. Therefore, the sampling frequency is broadened by 0.125 times to both ends, and the frequency shift corresponding to the maximum value of the waveform correlation peak is used as the optimal compensation for Doppler frequency offset. The results calculated by the RAM model show that the predicted and measured signal waveforms are in good agreement, and the correlation degree is significantly improved. Among them, the average normalized correlation values of the measured and predicted waveforms at different receiving depths in the first and second CZs are 0.84 and 0.82, respectively.