TECHNICAL PROGRAM CALENDAR
186th Meeting of the Acoustical Society of America/Acoustics Week in Canada
13-17 May 2024

MONDAY MORNING

8:05 1aAA **Architectural Acoustics, Noise, and Structural Acoustics and Vibration**: Sound Transmission and Impact Noise in Buildings I. 207

8:00 1aBAa **Biomedical Acoustics and Physical Acoustics**: Sonobiopsy for Noninvasive Molecular Diagnosis. 210

8:55 1aBAb **Biomedical Acoustics, Physical Acoustics, Engineering Acoustics, and Signal Processing in Acoustics**: Ultrasound Brain and Super-Resolution Imaging I. 212

9:30 1aCA **Computational Acoustics, Biomedical Acoustics, Structural Acoustics and Vibration, Engineering Acoustics, and Physical Acoustics**: Computational Methods for Acoustic Absorption in Materials. 211

8:00 1aID **Interdisciplinary and Student Council**: Introduction to Technical Committees. 201

8:00 1aPA **Physical Acoustics, Computational Acoustics, Engineering Acoustics, and Structural Acoustics and Vibration**: Developments and Applications in Phononic Crystals. 202

8:00 1aPP **Psychological and Physiological Acoustics**: Psychological and Physiological Acoustics Poster Session. 214

10:00 1aSA **Structural Acoustics and Vibration, Engineering Acoustics, and Physical Acoustics**: Eric Ungar's Contributions to Structural Acoustics Research and Applications. 203

9:00 1aSP **Signal Processing in Acoustics**: Signal Processing in Acoustics Poster Potpourri. 214

8:00 1aUW **Underwater Acoustics, Acoustical Oceanography, Computational Acoustics, and Signal Processing in Acoustics**: Data Science in Ocean Acoustics I. 215

MONDAY AFTERNOON

1:00 1pAA **Architectural Acoustics, Noise, and Structural Acoustics and Vibration**: Sound Transmission and Impact Noise in Buildings II. 207

1:00 1pBAa **Biomedical Acoustics**: General Topics in Biomedical Acoustics: Imaging. 210

1:00 1pBAb **Biomedical Acoustics, Physical Acoustics, Engineering Acoustics, and Signal Processing in Acoustics**: Ultrasound Brain and Super-Resolution Imaging II. 212

1:00 1pCA **Computational Acoustics, Physical Acoustics, and Underwater Acoustics**: Diffusion Equation and Energy Flux Methods Across Acoustics. 211

1:00 1pEA **Engineering Acoustics, Physical Acoustics, and Structural Acoustics and Vibration**: Applications of Acoustic Metamaterials. 204

1:30 1pMU **Musical Acoustics**: General Topics in Musical Acoustics I. 209

1:00 1pNSa **Noise**: Jet & Rocket Noise. 205

1:00 1pNSb **Noise, Psychological and Physiological Acoustics, and ASA Committee on Standards**: Evaluation of Hearing Protection Devices with Impulse Noise and Acoustic Test Fixtures. 201

1:00 1pPAa **Physical Acoustics, Structural Acoustics and Vibration, Engineering Acoustics, and Signal Processing in Acoustics**: Cyber Threats and Acoustical Systems. 202

1:00 1pPAb **Physical Acoustics**: Infrasound. 206

1:00 1pPP **Psychological and Physiological Acoustics**: Toward More Inclusive Research Practices in P&P I. 208

1:00 1pSC **Speech Communication**: Speech Perception Poster Session I. 214

1:00 1pSA **Structural Acoustics and Vibration, Education in Acoustics, and Physical Acoustics**: Mistakes and Lessons Learned in Structural Acoustics and Vibrations. 203

1:00 1pSP **Signal Processing in Acoustics**: Signal Processing in Acoustics Potpourri I. 213

1:00 1pUW **Underwater Acoustics, Acoustical Oceanography, Computational Acoustics, and Signal Processing in Acoustics**: Data Science in Ocean Acoustics II. 215

TUESDAY MORNING

7:55 2aAAa **Architectural Acoustics and ASA Committee on Standards**: Show Your Data: Architectural Acoustics Metrics I. 207

8:00 2aAAb **Architectural Acoustics**: Student Design Competition (Poster Session). 214

8:00 2aAB **Animal Bioacoustics, Computational Acoustics, Signal Processing in Acoustics, Underwater Acoustics, and Acoustical Oceanography**: Data to Information, Navigating the Application of Acoustic Data for Conservation I. 204

8:30 2aAO **Acoustical Oceanography**: Topics in Acoustical Oceanography I. 215

7:55 2aBAa **Biomedical Acoustics, Physical Acoustics, Engineering Acoustics, and Signal Processing in Acoustics**: Ultrasound Beamforming and its Applications I. 212

8:00 2aBAb **Biomedical Acoustics, Education in Acoustics, and Physical Acoustics**: Return of the Writer. 210

8:00 2aCA **Computational Acoustics**: Computational Acoustics Methods Evaluation. 211

10:15 2aEA **Engineering Acoustics**: Multidomain Modeling of Acoustical Systems. 211

8:40 2aMU **Musical Acoustics**: Winds Instruments I. 209

8:00 2aNSa **Noise, Computational Acoustics, Psychological and Physiological Acoustics, Structural Acoustics and Vibration, and Engineering Acoustics**: Advanced Air Mobility Noise: Noise from New Air Transportation in Urban and Underserved Communities I. 205

8:30 2aNSb **Noise**: Soundscape - Focus on Applications I. 201

8:00 2aPA **Physical Acoustics**: Wind Noise. 202

9:30 2aPP **Psychological and Physiological Acoustics**: Toward More Inclusive Research Practices in P&P II. 208

9:00 2aSA **Structural Acoustics and Vibration**: Acoustic Metamaterials I. 206

8:00 2aSC **Speech Communication**: VowelFest: Honoring the Past and Celebrating the Present I. 203

8:00 2aSP **Signal Processing in Acoustics**: Data Augmentation in Signal Processing: Advancing Performance through Artificial Data Generation. 213

TUESDAY AFTERNOON

1:00 2pAAa **Architectural Acoustics**: Show Your Data: Architectural Acoustics Metrics II. 207

3:20 2pAAb **Architectural Acoustics**: Building Envelope Sound Isolation I. 201

2:45 2pAB **Animal Bioacoustics**: Data to Information, Navigating the Application of Acoustic Data for Conservation II. 203

1:00 2pAO **Acoustical Oceanography**: Acoustical Oceanography Prize Lecture. 215

1:00 2pBAa **Biomedical Acoustics**: General Topics in Biomedical Acoustics: Microbubbles. 210

1:00 2pBAb **Biomedical Acoustics**: Ultrasound Beamforming and its Applications II. 212

1:00 2pCA **Computational Acoustics**: Application of Model Reduction Across Acoustics. 211

1:00 2pEA **Engineering Acoustics**: Things That Go Boom: High Amplitude Acoustic Sources. 204

1:00 2pED **Education in Acoustics**: Acoustics Education: A Potpourri of Classical and Unusual Materials and Demonstrations. 213

1:00 2pMUa **Musical Acoustics**: Winds Instruments II. 209
4:30 2pMUb **Musical Acoustics**: Native American Flute Concert. 209
1:00 2pNSa **Noise**: Advanced Air Mobility Noise: Noise from New Air Transportation in Urban and Underserved Communities II. 205
1:00 2pNSb **Noise and Architectural Acoustics**: Soundscape - Focus on Applications II. 201
3:05 2pNSc **Noise**: Honoring the Life and Legacy of Tony F. W. Embleton. 205
1:00 2pPA **Physical Acoustics, Strucutral Acoustics and Vibration, and Engineering Acoustics**: Resonant Ultrasound Spectroscopy For Characterizing Material Property and Structure. 202
1:00 2pPP **Psychological and Physiological Acoustics**: The Importance of Binaural Listening: Speech Intelligibility, Localization, and Virtual Environments. 208
1:00 2pSA **Structural Acoustics and Vibration, Engineering Acoustics, and Physical Acoustics**: Acoustic Metamaterials II. 206
1:00 2pSCa **Speech Communication**: VowelFest: Honoring the Past and Celebrating the Present II. 203
2:30 2pSCb **Speech Communication**: VowelFest III (Poster Session). 214
3:50 2pSP **Signal Processing in Acoustics, Biomedical Acoustics, Underwater Acoustics, and Physical Acoustics**: Denoising and Enhancing Acoustic Signals. 208
2:15 2pUW **Underwater Acoustics**: Geoacoustics of Marine Sediments. 215

WEDNESDAY MORNING

8:00 3aAAa **Architectural Acoustics**: Memoriam for Jerry Christoff. 201
8:00 3aAAb **Architectural Acoustics**: Soundscape Simulation: Opportunities and Challenges I. 207
9:55 3aAAc **Architectural Acoustics**: AIA CEU Presenters Course Training Session. 201
8:00 3aAB **Animal Bioacoustics**: Acoustic Ecology, Biological Soundscapes, and Animal Vocal Communication and Physiology I. 204
7:55 3aAOa **Acoustical Oceanography**: Topics in Acoustical Oceanography II. 215
10:45 3aAOb **Acoustical Oceanography, Underwater Acoustics, and Animal Bioacoustics**: Sound at Temperate and Tropical Coral Reefs. 215
7:55 3aBAa **Biomedical Acoustics, Physical Acoustics, and Structural Acoustics and Vibration**: SonoDynamic Therapy. A New Hope. 212
10:00 3aBAb **Biomedical Acoustics**: Biomedical Acoustics Best Student Paper Award Poster Session. 214
8:30 3aCA **Computational Acoustics**: The Phononic Dispersion Relations: Calculation, Interpretation, and Applications in Phononics and Metamaterials. 102
8:30 3aED **Education in Acoustics and Musical Acoustics**: Artistic and Technical Approaches to Acoustics. 211
8:30 3aMU **Musical Acoustics**: Applications of Physical Modeling of Musical Instruments. 209
8:00 3aNSa **Noise**: Assorted Topics on Noise I. 205
8:00 3aPAa **Physical Acoustics, Underwater Acoustics, and Acoustical Oceanography**: Meteorological Acoustics. 210
9:45 3aPAb **Physical Acoustics, Engineering Acoustics, Signal Processing in Acoustics, and Structural Acoustics and Vibration**: Characterization of Electronic Materials, Components, Devices, and Batteries. 202
8:00 3aPP **Psychological and Physiological Acoustics**: Temporal Processing, Aging, and Hearing Loss: Research Inspired by Christian Fullgrabe. 206/208
10:15 3aSA **Structural Acoustics and Vibration, Education in Acoustics, and Physical Acoustics**: My Favorite Homework Problems in Structural Acoustics in Vibrations. 212
8:00 3aSC **Speech Communication, Architectural Acoustics, Education in Acoustics, and Psychological and Physiological Acoustics**: Classroom Acoustics and Speech Communication. 203
8:00 3aSP **Signal Processing in Acoustics, Acoustical Oceanography, Underwater Acoustics, Computational Acoustics, and Physical Acoustics**: Bayesian and Machine Learning in Acoustics I. 213

WEDNESDAY AFTERNOON

1:00 3pAA **Architectural Acoustics**: Soundscape Simulation: Opportunities and Challenges II. 201
12:55 3pAO **Acoustical Oceanography**: Munk Award Lecture. 215
1:00 3pED **Education in Acoustics**: Acoustics Education Prize Lecture. 212
2:15 3pID **Interdisciplinary**: Keynote Lecture. 206/208
1:00 3pNS **Noise**: Assorted Topics on Noise II. 205
1:00 3pPP **Psychological and Physiological Acoustics**: Auditory Neuroscience Prize Lecture. 207

THURSDAY MORNING

8:00 4aAA **Architectural Acoustics**: Acoustics in Mass Timber. 207
8:30 4aAB **Animal Bioacoustics**: Acoustic Ecology, Biological Soundscapes, and Animal Vocal Communication and Physiology II. 204
7:55 4aBAa **Biomedical Acoustics, Structural Acoustics and Vibration, Physical Acoustics, Acoustical Oceanography, Computational Acoustics, and Engineering Acoustics**: Wave Propagation in Complex Media: From Theory to Applications I. 212
9:00 4aBAb **Biomedical Acoustics and Physical Acoustics**: Droplets Strike Back I. 210
9:00 4aCA **Computational Acoustics, Structural Acoustics and Vibration, and Engineering Acoustics**: Validation and Verification. 211
10:00 4aID **Interdisciplinary, Student Council, Education in Acoustics, and Physical Acoustics**: Graduate Studies in Acoustics Poster Session. 214
9:00 4aMU **Musical Acoustics**: General Topics in Musical Acoustics II. 209
8:30 4aNS **Noise, Physical Acoustics, and Engineering Acoustics**: Methods for Community Noise Testing and Analysis I. 205
8:00 4aPA **Physical Acoustics and Signal Processing in Acoustics**: Acousto-Optic Sensing. 206
8:00 4aPP **Psychological and Physiological Acoustics and Speech Communication**: Interactions Between Voice and Speech Perception. 208
9:00 4aSA **Structural Acoustics and Vibration, Biomedical Acoustics, Engineering Acoustics, Physical Acoustics, and Musical Acoustics**: Additive Manufacturing and Acoustics. 201
8:00 4aSCa **Speech Communication**: Language and Sports. 203
10:00 4aSCb **Speech Communication**: Speech Perception Poster Session II. 214
8:00 4aSP **Signal Processing in Acoustics, Acoustical Oceanography, Underwater Acoustics, Computational Acoustics, and Physical Acoustics**: Bayesian and Machine Learning in Acoustics II. 213
8:00 4aUW **Underwater Acoustics and Acoustical Oceanography**: Scintillation in Acoustic Propagation. 215

THURSDAY AFTERNOON

1:00 4pAAa **Architectural Acoustics, Noise, Structural Acoustics and Vibration, Signal Processing in Acoustics, and Computational Acoustics**: Artificial Intelligence and Machine Learning for Built Environments. 207
3:25 4pAAb **Architectural Acoustics and Structural Acoustics and Vibration**: Building Envelope Sound Isolation II. 201
1:00 4pBAa **Biomedical Acoustics and Physical Acoustics**: Droplets Strike Back II. 210
1:00 4pBAb **Biomedical Acoustics, Structural Acoustics and Vibration, Physical Acoustics, Acoustical**

FRIDAY MORNING

FRIDAY AFTERNOON

SCHEDULE OF COMMITTEEE MEETINGS AND OTHER EVENTS

ASA COUNCIL AND ADMINISTRATIVE COMMITTEES

Sun, 12 May, 6:30 p.m.	Executive Council Dinner	Westin Ontario Rm., Level 3
Mon, 13 May, 9:00 a.m.	Executive Council	102
Mon, 13 May, 1:00 p.m.	Technical Council	102
Tue, 14 May, 7:00 a.m.	Member Engagement	101
Tue, 14 May, 7:30 a.m.	Editorial Board	102
Tue, 14 May, 7:30 a.m.	Panel on Public Policy	103
Tue, 14 May, 8:00 a.m.	CIRDI	107
Tue, 14 May, 5:00 p.m.	Newman Fund	101
Wed, 15 May, 7:00 a.m.	Regional and Student Chapters	104
Wed, 15 May, 7:30 a.m.	Finance	101
Wed, 15 May, 9:30 a.m	Acoustical Society Foundation Fund	103
Wed, 15 May, 11:00 a.m.	Public Relations	104
Wed, 15 May, 11:45 a.m.	Medals and Awards	101
Thu, 16 May, 7:00 a.m.	Investments	101
Thu, 16 May, 7:30 a.m.	Practitioners and Industry	104
Fri, 17 May, 8:00 a.m.	Technical Council	102
Fri, 17 May, 12:00 noon	Executive Council	102

TECHNICAL COMMITTEEE OPEN MEETINGS

Tue, 14 May, 5:30 p.m.	Acoustical Oceanography	215
Tue, 14 May, 5:30 p.m.	Animal Bioacoustics	203
Tue, 14 May, 5:30 p.m.	Architectural Acoustics	201
Tue, 14 May, 5:30 p.m.	Engineering Acoustics	204
Tue, 14 May, 5:30 p.m.	Physical Acoustics	202
Tue, 14 May, 5:30 p.m.	Psychological and Physiological Acoustics	207
Tue, 14 May, 5:30 p.m.	Signal Processing in Acoustics	208
Wed, 15 May, 7:30 p.m.	Biomedical Acoustics	212
Wed, 15 May, 7:30 p.m.	Structural Acoustics and Vibration	207
Thu, 16 May, 5:30 p.m.	Computational Acoustics	211
Thu, 16 May, 5:30 p.m.	Musical Acoustics	209
Thu, 16 May, 5:30 p.m.	Noise	205
Thu, 16 May, 5:30 p.m.	Speech Communication	207
Thu, 16 May, 5:30 p.m.	Underwater Acoustics	215

STANDARDS COMMITTEES AND WORKING GROUPS

Tue, 14 May, 8:00 a.m.	Standards Plenary including TAGs	104
Tue, 14 May, 9:00 a.m.	ASC S12 Noise	104
Tue, 14 May, 11:00 a.m.	ASC S2 Mechanical Vibration and Shock	104
Tue, 14 May, 12:45 p.m.	ASC S3 Bioacoustics	104
Tue, 14 May, 2:15 p.m.	ASC S3/SC1 Animal Bioacoustics	104
Tue, 14 May, 3:45 p.m.	ASC S1 Acoustics	104
Wed, 15 May, 7:00 a.m.	ISO TC 43/SC 1/WG45	106
Thu, 15 May, 8:00 a.m.	ISO TC 43/SC 1/WG45	106
Thu, 16 May, 7:00 a.m.	Standards How To Guidelines Workshop	105

MEEETING SERVICES, SPECIAL EVENTS, SOCIAL EVENTS

Mon, 13 May 7:00 a.m.–5:00 p.m. Tue-Thu, 14-16 May 7:30 a.m.–5:00 Fri, 17 May 7:30 a.m.–12:00 noon	Registration	Colonel By Foyer South, 1st Fl
Mon–Thu, 13–16 May Fri, 17 May 7:00 a.m.–12:00 noon	A/V Preview	Office 2C (Rideau Canal North Foyer)
Mon–Thu, 13–16 May 8:00 a.m.–5:00 p.m. Fri, 17 May 8:00 a.m.–12:00 p.m.	Mothers' Room	Office 2A (Rideau Canal North Foyer)
Mon–Fri, 13–17 May, 9:30 a.m.–11:00 a.m.	Morning Coffee Breaks	Rideau Canal Foyer 2nd Floor
Tue, 14 May 2:00 p.m.–3:00 p.m.	Afternoon Coffee Break	Rideau Canal Foyer 2nd Floor
Mon, 13 May 8:00 a.m.–10:00 a.m.	Accompanying Persons	Room 101
Mon, 13 May 5:30 p.m.–7:00 p.m.	Exhibit Opening Reception	Rideau Canal Foyer 2nd Floor
Mon, 13 May 5:30 p.m.–6:30 p.m.	NSF Grant Workshop	Room 213
Mon, 13 May 5:15 p.m.–5:45 p.m.	First-Time Orientation Session	Room 203
Mon, 13 May 5:45 p.m. 7:30 p.m.	Student Meet and Greet	Trillium Foyer 4th Floor
Tue, 14 May 8:15 a.m.	Tour-National Research Council Building Acoustics	Daly St. Exit Colonel By Foyer
Tue, 14 May 9:00 a.m.–5:00 p.m.	Exhibit	Rideau Canal Foyer 2nd Floor
Wed, 15 May 9:00 a.m.-2:00 p.m.		
Tue, 14 May 6:00 p.m.–8:00 p.m.	Student Reception	Westin, Governor General, 4th Floor
Wed, 15 May 8:00 a.m.–9:30	Standards How To Guidelines Workshop	Room 105
Wed, 15 May 11:45 a.m.–1:30 p.m.	Women in Acoustics Luncheon	Westin, Twenty Two, 22nd floor
Wed, 15 May 2:15 p.m.–3:15 p.m.	Keynote Lecture	Room 206/208
Wed, 15 May 3:30 p.m.–6:00 p.m.	ASA and CAA Plenary Session/Awards Ceremony	Canada 3 3rd Floor
Wed, 15 May 6:00 p.m.–7:30 p.m.	Social Hour	Westin, Ballroom, 4th Level
Wed, 15 May 8:00 p.m.–12:00 midnight	ASA Jam	Canada 3 3rd Floor
Thu, 16 May 9:15 a.m.–11:30 a.m.	Tour - National Arts Centre	1 Eldridge St.
Thu, 16 May 12:45	Tour-National Research Council Aeroacoustics	Daly St. Exit Colonel By Foyer

186th Meeting of the Acoustical Society of America and Acoustics Week in Canada

The 186th meeting of the Acoustical Society of America and Acoustics Week in Canada will be held Monday through Friday, 13-17 May 2024 at the Shaw Centre and Westin Ottawa Hotel, Ottawa, Canada

SECTION HEADINGS

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

1. REGISTRATION

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

Registration will open on Monday, 13 May, at 7:00 a.m. in Colonel By Foyer South on the First Level of the Shaw Centre.

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

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

2. TECHNICAL SESSIONS

The technical program includes over 1200 abstracts.

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, 13 May
2-Tuesday, 14 May
3-Wednesday, 15 May
4-Thursday, 16 May
5-Friday, 17 May

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

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

In sessions where the same group is the primary organizer of more than one session scheduled in the same morning or afternoon, a fifth character, either lower-case “a,” or “b,” is used to distinguish the sessions. Each paper within a session is identified by a paper number following the session-designating characters, in conventional manner. As hypothetical examples:

paper 2pEA3 would be the third paper in a session on Tuesday afternoon organized by the Engineering Acoustics Technical Committee; 3pSAb5 would be the fifth paper in the second of two sessions on Wednesday afternoon sponsored by the Structural Acoustics and Vibration Technical Committee.

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

4. EXHIBIT AND EXHIBIT OPENING RECEPTION

An instrument and equipment exhibition will be located in the Second Floor Atrium and will open on Monday, 13 May, with an evening reception serving a complimentary drink. Exhibit hours are Monday, 13 May, 5:30 p.m. to 7:00 p.m., Tuesday, 14 May, 9:00 a.m. to 5:00 p.m., and Wednesday, 15 May, 9:00 a.m. to 2:00 p.m.

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

5. PRIZES AND PRIZE LECTURES

The 2024 Hartmann Prize in Auditory Neuroscience and the Medwin Prize in Acoustical Oceanography will be presented at the Plenary session on Wednesday, 15 May, at 3:30 p.m. in Canada Hall 3.

The Auditory Neuroscience Prize Lecture will be presented by Christopher Shera on Wednesday, 15 May, in session 3pPP at 1:00 p.m. in Room 207. The Acoustical Oceanography Prize Lecture will be presented by Julien Bonnel on Tuesday, 14 May, in session 2pAO at 1:00 p.m. in Room 215. The 2023 Acoustics Education Prize Lecture will be presented by Scott D. Sommerfeldt on Wednesday, 15 May, in session 3pED at 1:00 p.m. in Room 212.

The Munk Award Lecture will be presented by Ross Chapman on Wednesday, 15 May, at 12:55 a.m. in session 3pAO in Room 215.

6. TECHNICAL COMMITTEE OPEN MEETINGS

Technical Committees will hold open meetings on Tuesday, Wednesday, and Thursday.

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

7. TECHNICAL TOURS

Three Technical Tours are scheduled. Both tours to the National Research Council will be by bus. The third tour is a walking tour.

Buses for both National Research Council tours will load at the Shaw Centre at 8:15 a.m. for the 20-minute trip to the facility. Please remember that all tour participants must be able to show a government issued id. If your id is not written in English, please provide a translation to show at the time of the tour.

Those who wish to walk to the National Arts Centre with other tour members should gather near the ASA registration desk on the 1st floor in Colonel By Foyer South.

Tuesday, 14 May, 8:15 a.m. to 11:30 a.m.
National Research Council of Canada Building Acoustics Facilities.
Thursday, 16 May, 1:00 p.m. to 4:30 p.m.
National Research Council of Canada Aerospace Research Facilities.
Thursday, 16 May, 9:00 a.m. to 11:00 a.m.
National Arts Centre – refer to Google maps for directions
Tour starts promptly at 9:15 a.m.

8. GRANT WORKSHOP: NSF FUNDING OPPORTUNITIES

Rachel Theodore, program officer at the National Science Foundation (NSF) and ASA member, will present information about new opportunities for funding within the NSF. In particular she will describe programs for interdisciplinary research appropriate for ASA members. The workshop will be held on Monday, 13 May, 5:30 to 6:30 p.m. in Room 213.

9. STANDARDS WORKSHOP

Volunteers are sought to be part of a special Workshop developing a new ASA Standards product. This workshop—led by Dr. Donald Peterson, Dean, College of Engineering and Engineering Technology, Northern Illinois University—is tasked with promoting the involvement of stakeholders to encourage collaboration across various sectors to capture expectations for new ASA Standards “how to” guides that will accompany and complement select ASA standards—Guides designed to assist standards users (e.g., university students, industry practitioners and government officials) on how to apply the standards’ technical information in a practical, efficient way for their work or research. Consider them “quick start manuals” to putting standards into play right away! The workshop will be held on Wednesday, 15 May, from 8:00 a.m. to 9:00 a.m. in Room 202.

10. PLENARY SESSION AND AWARDS CEREMONY

A plenary session will be held Wednesday, 15 May, at 3:30 p.m. in Canda Hall 3. The Hartmann Prize in Auditory Neuroscience, the Medwin Prize in Acoustical Oceanography, the Silver Medal in Acoustical Oceanography, the Wallace Clement Sabine Medal, the R. Bruce Lindsay Award, the Helmholtz-Rayleigh Interdisciplinary Silver Medal, and the Gold Medal will be presented. Certificates will be presented to Fellows elected at the Sydney meeting and to the recipients of 2023 Science Communication Awards.

The Canadian Acoustical Association will hold its Plenary Session and Awards Ceremony immediately following the ASA Plenary.

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

11. ANSI STANDARDS COMMITTEES

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

People interested in attending and in becoming involved in working group activities should contact the ASA Standards Manager for further information about these groups, or about the ASA Standards Program in general, at the following address: Nancy Blair-DeLeon, ASA Standards Manager, Standards Secretariat, Acoustical Society of America, 1305 Walt Whitman Road, Suite 110, Melville, NY 11747-4300; T: 631-390-0215; E: asastds@acousticalsociety.org

12. COFFEE BREAKS

Morning coffee breaks will be held daily from 9:30 a.m. to 11:00 a.m. and an afternoon break will be held on Tuesday from 2:30 p.m. to 3:45 p.m. in the Rideau Canal Foyer on level 2. Breaks on Monday, Thursday, and Friday will be held in Rideau Canal Foyer North.

13. A/V PREVIEW ROOM

The A/V preview room will be set up in office 2C (entrance on Rideau Canaal North foyer) and will be available Monday through Thursday, 13-16 May, from 7:00 a.m. to 5:00 p.m. and Friday, 17 May, from 7:00 a.m. to 12:00 noon.

14. MOTHERS ROOM

A Mothers Room for ASA meeting attendees will be available Monday to Friday, 13-17 May, in Room 2A Rideau Canal Foyer North. The hours are Monday to Thursday, 8:00 a.m. to 5:00 p.m. and Friday, 8:00 a.m. to 12:00 noon.

15. SOCIAL

A Social will be held on Wednesday evening, 6:00 p.m. to 7:30 p.m. at the Westin Ottawa Hotel Ballroom on the 4th Level.

The ASA hosts this social hours to provide a relaxing setting for meeting attendees to meet and mingle with their friends and colleagues as well as an opportunity for new members and first-time attendees to meet and introduce themselves to others in the field.

16. STUDENT EVENTS: MEET AND GREET, STUDENT RECEPTION

Follow the student twitter throughout the meeting @ ASAStudents.

The Student Meet and Greet will be held on Monday, 13 May from 5:45 p.m. to 7:30 p.m. in the Trillium Foyer on the 4th floor of Shaw Centre where refreshments and a cash bar will be available.

The National Council of Acoustical Consultants will present travel awards to students at the start of the event.

The Students' Reception will be held on Tuesday, 14 May, from 6:00 p.m. to 8:00 p.m. in Governor General of The Westin Ottawa Hotel. This reception will provide an opportunity for students to meet informally with fellow students and other members of the Acoustical Society. All students are encouraged to attend, especially students who are first time attendees or those from smaller universities.

17. WOMEN IN ACOUSTICS LUNCHEON

The Women in Acoustics luncheon will be held at 11:45 a.m. on Wednesday, 15 May, in Twenty Two on the 22nd floor of the Westin Ottawa Hotel.

18. JAM SESSION

You are invited to Canada 3 (3rd floor, Shaw Centre) on Wednesday night, 15 May, from 8:00 p.m. to midnight for the ASA Jam. Bring your axe, horn, sticks, voice, or anything else that makes music. Musicians and non-musicians are all welcome to attend. A full PA system, backline equipment, guitars, bass, keyboard, and drum set will be provided. All attendees will enjoy live music, a cash bar with snacks, and all-around good times. Don't miss out.

19. ACCOMPANYING PERSONS PROGRAM

Spouses and other visitors are welcome at the Ottawa meeting. A hospitality room for accompanying persons will be open in Room 101 from 8:00 a.m. to 10:00 a.m. on Monday only.

You are welcome to the accompanying persons room, the Wednesday Social, the Plenary Session on Wednesday afternoon, and the JAM on Wednesday evening.

20. PROCEEDINGS OF MEETINGS ON ACOUSTICS (POMA)

The Ottawa meeting will have a published proceedings, and submission is optional. The proceedings will be a separate volume of the online journal Proceedings of Meetings on Acoustics (POMA). This is an open access journal, so its articles are available in pdf format for downloading without charge to anyone in the world. All 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, including MS Word and LaTeX templates, can be found at the site https://pubs.aip.org/asa/poma.

Authors who are members of the Canadian Acoustical Association (CAA) are invited to submit a two-page conference proceedings paper that will be published in Canadian Acoustics in the June 2024 issue.

CAA authors who also submit a paper to POMA should change the POMA paper title to differentiate it from the CAA proceedings paper. Content of the papers is also expected to differ following naturally from the 12-page limit for POMA and the 2-page format for Canadian Acoustics. See the CAA webpage at https://jcaa.caa-aca.ca/index.php/jcaa for details.

21. TECHNICAL PROGRAM ORGANIZING COMMITTEE

Meaghan O'Reilly, Technical Program Chair; Christopher Bassett, Acoustical Oceanography; Carrie Wall-Bell, Animal Bioacoustics; Brandon Cudequest, Architectural Acoustics; John Cormack, James Kwan, Biomedical Acoustics; Amanda Hanford, Computational Acoustics; Daniel Russell, Education in Acoustics; Ahmed Allam, Michael Haberman, Engineering Acoustics; Andrew Piacsek, Gary Scavone, Musical Acoustics; Aaron Vaughn, James Phillips, Hales Swift; Noise; Ralph Herman, Joel Lonzaga, Physical Acoustics; Gregory Ellis, Chris Steckert, Psychological and Physiological Acoustics; Trevor Jerome, Signal Processing in Acoustics; Kelly Berkson, Pasquale Bottalico, Lisa Redford, Benjamin Tucker, Speech Communication; Anthony Bonomo, Stephanie Konarski, Structural Acoustics and Vibration; David Dall'Osto, Underwater Acoustics; Brijonnay Madrigal, Student Council.

22. MEETING ORGANIZING COMMITTEE

David C. Barclay, Chair; Meaghan O'Reilly, Sebastian Ghinet, Joana Rocha, Canadian Acoustical Association representatives.

23. PHOTOGRAPHING AND RECORDING

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

24. ABSTRACT ERRATA

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

25. GUIDELINES FOR ORAL PRESENTATIONS,

Preparation of Visual Aids

- See the guidelines for computer projection in section 41 below.
- Allow at least one minute of your talk for each slide (e.g., PowerPoint). No more than 12 slides for a 15-minute talk (with 3 minutes for questions and answers).
- Minimize the number of lines of text on one visual aid. 12 lines of text should be a maximum. Include no more than 2 graphs/plots/figures on a single slide. Too little information is better than too much.
- Presentations should contain simple, legible text that is readable from the back of the room.
- Characters should be at least 0.25 inches (6.5 mm) in height to be legible when projected. A good rule of thumb is that text should be 20 point or larger (including labels in inserted graphics). Anything smaller is difficult to read.
- Make symbols at least 1/3 the height of a capital letter.
- For computer presentations, use all of the available screen area using landscape orientation with very thin margins. If your institution's logo must be included, place it at the bottom of the slide.
- Sans serif fonts (e.g., Arial, Calibri, and Helvetica) are much easier to read than serif fonts (e.g., Times New Roman) especially from afar. Avoid thin fonts (e.g., the horizontal bar of an e may be lost at low resolution thereby registering as a c.)
- Do not use underlining to emphasize text. It makes the text harder to read.
- All axes on figures should be labeled.
- No more than 3–5 major points per slide.
- Consistency across slides is desirable. Use the same background, font, font size, etc. across all slides.
- Use appropriate colors. Avoid complicated backgrounds and do not exceed four colors per slide. Backgrounds that change from dark to light and back again are difficult to read. Keep it simple.
- If using a dark background (dark blue works best), use white or yellow lettering. If you are preparing slides that may be printed to paper, a dark background is not appropriate.
- If using light backgrounds (white, off-white), use dark blue, dark brown or black lettering.
- DVDs should be in standard format.

Presentation

- Organize your talk with introduction, body, and summary or conclusion. Include only ideas, results, and concepts that can be explained in the allotted time. Four elements to include are:
- Statement of research problem
- Research methodology
- Review of results
- Conclusions
- No more than 3–5 key points can be covered adequately in a 15-minute talk so keep it concise.
- Rehearse your talk so you can confidently deliver it in the allotted time. Session Chairs have been instructed to adhere to the time schedule and to stop your presentation if you run over.
- An A/V preview room will be available for viewing computer presentations before your session starts. It is advisable to preview your presentation because in most cases you will be asked to load your presentation onto a computer which may have different software or a different configuration from your own computer.
- Arrive early enough so that you can meet the session chair, load your presentation on the computer provided, and familiarize yourself with the microphone, computer slide controls, laser pointer, and other equipment that you will use during your presentation. There will be many presenters loading their materials just prior to the session so it is important that you check that all multi-media elements (e.g., sounds or videos) play accurately prior to the day of your session.
- Each time you display a visual aid the audience needs time to interpret it. Describe the abscissa, ordinate, units, and the legend for each figure. If the shape of a curve or some other feature is important, tell the audience what they should observe to grasp the point. They will not have time to figure it out for themselves. A popular myth is that a technical audience requires a lot of technical details. Less can be more.
- Turn off your cell phone prior to your talk and put it away from your body. Cell phones can interfere with the speakers and the wireless microphone.

26. SUGGESTIONS FOR EFFECTIVE POSTER PRESENTATIONS

Content

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

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

Design and layout

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

Lettering and text

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

Visuals

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

Presentation

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

Other suggestions

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

27. DATES OF FUTURE ASA MEETINGS

For further information on any ASA meeting, or to obtain instructions for the preparation and submission of meeting abstracts, contact the Acoustical Society of America, 1305 Walt Whitman Road, Suite 110, Melville, NY 11747-4300; Telephone: 516-576-2360; E-mail: asa@acousticalsociety.org

187th Meeting – Virtual Meeting, 18-22 November 2024
188th Meeting – joint with the International Congress on Acoustics, New Orleans, Louisiana 19-23 May 2025
189th Meeting – joint with the Japanese Acoustical Society,
190th Meeting – Philadelphia, Pennsylvania, 2006

Session 1aAA

Architectural Acoustics, Noise and Structural Acoustics and Vibration: Sound Transmission and Impact Noise in Buildings I

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

Invited Papers

8:05

1aAA1. The Impact Guide: Adapting ISO 12354-2 for calculating impact sound transmission to the North American context. Markus Mueller-Trapet (National Res. Council Canada, 1200 Montreal Rd., Ottawa, ON K1A 0R6, Canada, Markus.Mueller-Trapet@nrc-cnrc.gc.ca) and Jeffrey Mahn (National Res. Council Canada, Ottawa, ON, Canada)

To support a potential introduction of an impact sound requirement into one of the next editions of the National Building Code of Canada, the National Research Council of Canada has initiated several research projects. For one of these projects, an industry-sponsored consortium has been formed with the goal of providing supporting documents and online tools to help building designers incorporate impact sound requirements into their work. The main guide document, following the model of research report RR-331 for airborne sound, provides guidance and explanations on how to calculate and combine direct and flanking sound transmission, in this case for impact sound. The guide uses the method given in ISO 12354-2, adapted to the relevant ASTM standards used in North America. This contribution provides an overview of the work to create the impact sound guide document, how the implementation of ISO 12354-2 is achieved using ASTM standards and presents several example calculations highlighting the need to consider flanking sound transmission for impact sound.

8:25

1aAA2. Evaluating the perceived annoyance from impact sounds: Validation of previous experiments with an expanded set of recordings. Sabrina Skoda (National Res. Council Canada, 1200 Montreal Rd., Ottawa, ON K1A0R6, Canada, sabrina.skoda@nrc-cnrc.gc.ca), Markus Mueller-Trapet, Jeffrey Mahn, and Iara Batista da Cunha (National Res. Council Canada, Ottawa, ON, Canada)

Impact noise from neighbors in multi-unit residential buildings is commonly seen as an annoyance that may reduce the quality of life of building occupants. A project at the National Research Council of Canada has been evaluating the perceived annoyance from different types of impact sound, through the implementation of listening experiments in Canada, Korea, and Germany. A comparison among the countries revealed that test participants agreed on the relative annoyance of different impact sources, but the absolute levels of the annoyance were different between participants from the three countries. To explain the differences between the participants from the three countries, moderating factors, such as the test participants' housing situation and noise sensitivity, were taken into account. In order to validate the previous findings, the results from a further listening experiment conducted at the National Research Council using the same methodology but with a new set of impact sounds recorded on a wider variety of floor-ceiling assembly types will be presented.

8:45

1aAA3. Evaluating the uncertainty of laboratory measurements of low-frequency impact noise. Wayland Dong (Paul S. Veneklasen Res. Foundation, Santa Monica, California, CA) and John Lo Verde (Paul S. Veneklasen Res. Foundation, 1711 Sixteenth St., Santa Monica, CA 90404, jloverde@veneklasen.com)

Impact noise at low frequencies, specifically the third-octave bands below 100 Hz, is important to occupant reaction and are, therefore, routinely measured in field impact testing. For laboratory impact testing, the situation is less clear. Because the measured levels depend strongly on the specifics of the test chamber, it is expected that the reproducibility of low-frequency impact noise between laboratories to be poor and the translation to field measurements to be difficult. Despite these difficulties, the authors will demonstrate that it is possible to extract useful low frequency information from laboratory testing. The utility of laboratory testing for design of low-frequency impact noise is discussed.

9:05

1aAA4. Is there a right answer? A cross-laboratory study of concrete slabs. Evelyn Way (Maxxon Corp, 920 Hamel Rd., Hamel, MN 55340, eway@maxxon.com)

Commissioning a new acoustic test lab is a non-standardized process and validating test procedures and conditions is complicated by a lack of test samples with established performance. To aid in validating the results at the Maxxon Acoustics Lab, two concrete slabs were fabricated and tested in a Gage Repeatability and Reproducibility-style test design at Intertek Labs. After shipment from York, PA to Hamel, MN, mounting and edge conditions were established (as previously presented) at the Maxxon Labs, and the Gage R&R-style

testing was replicated at the Maxxon Lab. The results indicate sources of variability between labs that can be attributed to the standard test methods, physical differences in reverberation rooms and test chamber design, and sample mounting conditions. In addition to validation of a single lab, the results have lessons for the interpretation of lab test results to improve isolation performance comparison across products and assemblies to inform architectural acoustic design.

9:25

1aAA5. Impact ball testing in a laboratory setting. Evelyn Way (Maxxon Corp., 920 Hamel Rd., Hamel, MN 55340, eway@maxxon.com)

Impact ball test data have been collected across multiple structures by a procedure similar to Annex A of ISO 10140-3:2021. The results are presented at a high level to examine where impact ball testing is useful and what design characteristics of different structural assemblies and floor finishes are highlighted by the impact ball as opposed to other impact sources including the standard tapping machine.

Contributed Papers

9:45

1aAA6. On the vibroacoustics of cross-laminated timber embedded with acoustic black holes. Temitope Akinpelu (Concordia Univ., Montreal, Montreal, QC, Canada), Joonhee Lee (Concordia Univ., Montreal, EV 6.231, 1515 Rue Sainte-Catherine O, Montreal, QC H3H1Y3, Canada, joonhee.lee@concordia.ca), and Behrooz Yousefzadeh (Concordia Univ., Montreal, Montreal, QC, Canada)

The adoption of cross-laminated timber (CLT) in building construction is challenged by low-frequency impact noise. Conventional treatments typically involve increasing the mass of the floor assembly or employing structural decoupling techniques, both of which result in increased floor thickness. This study explores an alternative solution by examining the feasibility of embedding acoustic black holes (ABHs) into CLT floors. The ABH acts as a passive waveguide, concentrating vibrational energy in specific regions where it is transferred to damping layers and reduced. Using the finite element method, we investigated the vibroacoustic behavior of CLT plates within the 50–600 Hz frequency range. Our primary focus was on the influence of the ABH's outer radius and the number of ABH inclusions within a CLT plate. The findings indicate that increasing the outer diameter of the ABH lowers the cut-on frequency and enhances the effective damping of the CLT plate. The increase in damping extends the effectiveness of the ABH to a broader frequency range, with the most noticeable effect above the cut-on frequency. The number of ABH inclusions impacts the structural modes below the cut-on frequency, and the optimal number of inclusions depends on the frequency of interest.

10:00–10:15 Break

10:15

1aAA7. On-site noise and vibration measurements on diverse gym acoustic solutions and noise sources. Marina Rodrigues (CDM Stravitec, Reutenbeek 9 11, Overijse 3090, Belgium, m.rodrigues@cdm-stravitec.com) and Alfredo Rodrigues (CDM Stravitec, Toronto, ON, Canada)

The current market development for fitness spaces has led to a greater proximity to adjacent residential areas and has underscored the importance of a proper acoustic separation between spaces to ensure space use compatibility. While a good acoustic design is necessary to facilitate the compatibility of fitness spaces within residential buildings, both the novelty of the issue and the limited guidance on proper design methodologies for the location of fitness spaces and the equipment used, within a building structure, make designing effective acoustical solutions difficult. To validate and enhance the understanding of methodology for selecting an acoustic solution, a major gym chain operator in collaboration with CDM Stravitec conducted in-depth measurements at two gym locations. The measurements aimed to determine if equipment noise, both direct drops and synchronized repetitive excitation, correlates with the sound produced by a hard but relatively light object. Various equipments were analyzed with different acoustic solutions, ranging from equipment isolation pads, continuous rubber mats and tiles, and lightweight floors supported by continuous and discrete bearings (both elastomeric and springs). This paper presents the outcomes of the comprehensive noise and vibration measurement campaign.

10:30

1aAA8. The relationship between noise sensitivity and the perception of floor impact noise. Hiroshi Sato (Dept. of Information Technol. and Human Factors, AIST, 1 1 1 Umezono, AIST Tsukuba Headquaters, Tsukuba, Ibaraki 305-8560, Japan, sato.hiro@aist.go.jp), Manabu Chikai (Dept. of Information Technol. and Human Factors, AIST, Tsukuba, Ibaraki, Japan), Jeffrey Mahn, Iara B. Cunha, Markus Mueller-Trapet, and Sabrina Skoda (Constr. Res. Ctr., National Res. Council, Duesseldorf, Germany)

Noise sensitivity is a factor that determines how an individual perceives how uncomfortable a sound is, and noise sensitivity is believed to affect the annoyance response to floor impact noise in dwellings. In this study, we conducted a survey using the Weinstein Noise Sensitivity Scale given to subjects who participated in an evaluation of annoyance to floor impact noise and investigated the relationship with the results of the annoyance judgments. As with previous research, it is predicted that the reaction to annoyance to floor impact sound varies greatly depending on the listener's noise sensitivity and that a reaction of "annoy" is more likely to occur in the high sensitivity group, while a "annoy" judgment is less likely to occur in the low sensitivity group.

10:45

1aAA9. Sound flanking through common low-voltage electrical conduit in multi-family residential buildings. Michael Kundakcioglu (HGC Eng., 2000 Argentia Rd., Plaza 1, Ste. 203, Mississauga, ON L5N 1P7, Canada, mkundakcioglu@hgcengineering.com), Adam Doiron, and Jessica Tinianov (HGC Eng., Mississauga, ON, Canada)

Sound flanking between suites in multi-family residential buildings is a prevalent issue in architectural acoustics and building construction that can decrease the sound insulation performance of suite-demising partitions, compromising acoustical privacy and comfort for residents. In recent times, a specific deficiency regarding the sealing of common low-voltage electrical conduit routed between residential suites has been increasing in frequency in construction of residential buildings, resulting in otherwise appropriately designed suite demising configurations performing poorly *in situ*, and in some cases, even failing Ontario Building Code requirements for sound insulation between dwelling spaces during site testing. This article presents test data sampled from multiple residential buildings in Ontario, highlighting the extent of this issue and its implications, along with discussion regarding the proactive prevention and post-construction rectification of this issue.

11:00

1aAA10. Flanking noise at stacked townhouse entrance stairs. Gregory E. Clunis (Integral DX Eng. Ltd., Ottawa, ON, Canada, greg@integraldxengineering.ca) and Pier-Gui Lalonde (Integral DX Eng. Ltd., Ottawa, ON, Canada)

In stacked, wood-framed townhouses, entry stairs for upper units pass through the footprint of lower units. The demising construction between the units, therefore, includes the stairs and floor-ceiling below, as well as the stairwell sidewalls. This irregular construction creates the possibility of

significant sound flanking paths between upper and lower units, potentially leading to poor noise isolation performance and occupant dissatisfaction. This paper present practical solutions that have been used to address sound flanking in these areas, including the results of comparative field measurements to evaluate effectiveness. It is concluded that improvements are possible, although wood-framed entry stairs as typically designed are likely to remain an acoustical weakness.

11:15

1aAA11. Comparison of ASTC results with the calculation performed using the NRC soundPATHS calculator. Nicolas Leveque (MJM Acoust. Consultants, 753 Ste-Hélène, Longueuil, QC J4K 3R5, Canada, nleveque@mjm.qc.ca)

The NRC has developed a calculator that predict the apparent Sound insulation, as described in the procedure described in Section 5.8.1.4. or 5.8.1.5. of the 2015 edition of the National Building Code of Canada. Since 1984, MJM Acoustical Consultants has performed a lot of ASTC tests in the field on the huge variety of building structures (wood, steel, concrete, steel/concrete, etc.) and on different types of partitions (wood or steel). Since the 2015 National Building Code of Canada has been enforced in January 2022 in the Quebec province, there will be in increased demand to perform calculation the soundPaths calculator. Therefore, the performance of the calculator compared to real life situations should be assessed. The purpose of this paper is to present several comparison of the ASTC rating measured in the field and the rating provided by the soundPaths calculator using the same partitions compositions and building structure . We will then discuss the difference between the results and proposed some improvements of the NRC SoundPaths calculator.

11:30

1aAA12. Traffic noise transmitted indoors. Berndt Zeitler (Univ. of Appl. Sci. Stuttgart, Schellingstr. 24, Stuttgart 70174, Germany, berndt.zeitler@hft-stuttgart.de), Iara Batista da Cunha (National Res. Council Canada, Ottawa, ON, Canada), Martin Schneider (Univ. of Appl. Sci. Stuttgart, Stuttgart, Baden-Württemberg, Germany), and Markus Mueller-Trapet (National Res. Council Canada, Ottawa, ON, Canada)

In numerous countries, traffic noise is widely acknowledged as a highly disturbing form of pollution. Given that individuals spend a substantial portion of their day indoors, the impact of traffic noise perceived indoors is of considerable significance. To establish appropriate sound insulation requirements for buildings, it is essential to correlate subjective annoyance with objective ratings. This paper aims to present the regulations implemented by various countries and present preliminary findings from ongoing studies that involve listening tests using either measured or simulated traffic noise in indoor environments.

11:45

1aAA13. The cost of transparency: Balancing acoustic, financial, and sustainability considerations for glazed office partitions. Caroline Harvey (Arup Canada Inc., 121 Bloor St. East, Ste. 900, Toronto, ON M4W 3M5, Canada, caroline.harvey@arup.com), Vincent Jurdic (Arup Canada Inc., Montréal, QC, Canada), Chris Pollock, and Willem Boning (Arup USA, Inc., New York, NY)

Substantial areas of modern offices may be glazed, including office partitions and doors. The desire for transparent connection between adjacent spaces may be driven by a need for natural lighting, the aesthetics of glass, and a desire for inclusivity and openness within organizations. At the same time, organizations require speech privacy for confidential communications, a requirement that relies on good sound isolation performance from glazing and seals. This paper examines the cost of transparency for modern offices, with a focus on balancing the acoustic performance of glazed partitions with spatial planning, post-pandemic occupancy patterns, financial costs, and the carbon cost of extensive glazing. Drawing on recent work to address poor sound isolation in a building with multiple small private offices with glazed partitions onto open office areas, this paper examines the impact of low Noise Isolation Class (NIC) values between adjacent spaces, including voice privacy concerns, acoustic discomfort and enforced changes to occupancy patterns. The design of glazed partitions should address a range of privacy needs while balancing the benefits and costs of a “transparent” workplace in terms of acoustics, construction costs, and embodied carbon.

Session 1aBAa

Biomedical Acoustics and Physical Acoustics: Sonobiopsy for Noninvasive Molecular Diagnosis

Hong Chen, Chair
Washington Univ. in St. Louis, 4511 Forest Park Ave., St. Louis, MO 63108

Invited Papers

8:00

1aBAa1. Sonobiopsy for noninvasive molecular diagnosis of diseases. Hong Chen (Washington Univ. in St. Louis, 6338 Washington Ave., University City, MO 63130, chenhongxjtu@gmail.com)

Ultrasound technology has traditionally been pivotal in diagnostic imaging and therapeutic treatments. However, the advent of "sonobiopsy" has expand its application into molecular diagnosis. Sonobiopsy leverages ultrasound to noninvasively release disease-specific biomarkers into the bloodstream. This innovative approach enables spatially targeted and temporally controlled detection of disease-specific circulating biomarkers, offering a significant advancement in disease diagnosis. Initially conceptualized in 2009 by Gary Glazer's team at Stanford University, sonobiopsy has rapidly emerged as a groundbreaking technique. It is particularly promising in diagnosing brain diseases and other conditions, offering a less invasive alternative to traditional methods. This presentation aims to provide a comprehensive overview of the sonobiopsy field. It will cover its development, current applications, and potential future implications in medical diagnostics. The discussion will include insight into how sonobiopsy offers a more precise, controlled, and patient-friendly approach to disease detection, underscoring its growing importance in medicine.

8:30

1aBAa2. Releasing genetic biomarkers from cells and tissues with ultrasound. Roger J. Zemp (Electr. & Comput. Eng., Univ. of Alberta, 2nd Fl. ECERF, 9107-116 St., Edmonton, AB T6G 2V4, Canada, rzemp@ualberta.ca), Pradyumna Kedarisetti, and Joy Wang (Electr. & Comput. Eng., Univ. of Alberta, Edmonton, AB, Canada)

Ultrasound sonoporation has long been investigated as a means of enhancing therapeutic delivery into cells. More recently, however, ultrasound has been explored as means to liberate biomarkers out of cells. I will discuss our past efforts to enhance ultrasound biomarker release using microbubbles and nanodroplets both *in vivo* and *ex vivo*. Our recent *in vivo* work has demonstrated that ultrasound and nanodroplets can enhance extracellular vesicles and tumor DNA/RNA in the blood by 100-1000-fold. Because we can compare post-sonication blood biomarker levels to a pre-sonication blood draw baseline, and because any increase in biomarkers must be due to ultrasound treatment, this approach is a powerful alternative to biopies with much less tissue damage. In a different paradigm, we are exploring contrast-agent-free ablative acoustic mechanisms on a micro-scale to enhance biomarker release even further. Our group is further exploring novel avenues to detect circulating tumor cells in blood samples, where ultrasound treatment is applied to a separated blood fraction. By detecting biomarkers released from circulating tumor cells but primarily absent in blood cells, our approach is sensitive to single cells. Ongoing work in prostate cancer patients suggest promise for sensitive and specific detection of clinically significant prostate cancer. Ultrasound for biomarker release is a promising avenue of research which could lead to many important clinical applications given additional work and validation.

9:00

1aBAa3. Intersections of focused ultrasound and extracellular vesicles: Toward diversified biomarker enrichment and discovery. Natasha D. Sheybani (Biomedical Eng., Univ. of Virginia, Medical Res. Bldg. 5 (MR-5), 415 Ln. Rd., Charlottesville, VA 22908, nds3sa@virginia.edu)

Extracellular vesicles (EVs) are heterogeneous, membrane-bound structures shed by most fluid-interfacing cell types and play a critical role in orchestrating pathophysiological processes. With the advancement of liquid biopsy for cancer diagnosis and surveillance, EVs have burgeoned as a powerful asset toward circulating biomarker discovery owing to their diverse cargo (i.e., proteins, metabolites, nucleic acids, lipids). With the advent of focused ultrasound (FUS) technology as a tool for "sonobiopsy" in solid tumors, we recognize the increasing importance of deconvolving how the diverse bioeffects of FUS may influence EV quantity and quality. We have examined the impacts of thermal and mechanical FUS regimens on EV release and profile across cancer and immune cell contexts. Our observations suggest that both hyperthermia and microbubble-assisted FUS augment EV release acutely. On closer examination, hyperthermia-exposed EVs also display shifts in proteomic profile and differential immunomodulatory capacity. Ongoing studies are interrogating the impact of other FUS regimens on EV biology. Meanwhile, we are also directly investigating EVs from liquid biopsy specimens spanning murine cancer models, veterinary oncology (spontaneous canine cancers), and clinical trials. We expect that a continued work at the functional intersection of FUS and EVs will yield important, timely insight for this rapidly evolving field.

9:30

1aBAa4. Extracellular vesicle response following focused ultrasound-mediated blood–brain barrier opening for the detection of Alzheimer's pathology. Alina Kline-Schoder, Fotios Tsitsos, Alec Batts, Melody DiBenedetto, Sua Bae, Keyu Liu (Dept. of Biomedical Eng., Columbia Univ., New York, NY), and Elisa Konofagou (Dept. of Biomedical Eng., Columbia Univ., 630 West 168th St., Physicians & Surgeons 19-418, New York, NY 10032, ek2191@columbia.edu)

Focused ultrasound (FUS), in conjunction with systemically administered microbubbles, has been shown to induce transient and non-invasive blood–brain barrier opening (BBBO). Originally developed as a method of improving targeted drug delivery, FUS-BBBO has recently been proposed as a means of amplifying detection of disease biomarkers in the bloodstream. We hereby describe a method that can detect extracellular vesicles (EVs) after BBB opening in Alzheimer's disease (AD) mice and patients. BBBO is shown to enhance the release of EVs with disease-specific cargo to the bloodstream when targeting amyloid affected regions and shown to be more sensitive than cell free DNA. In this presentation, we will describe the overall methodology and quantify the concentration and content of EVs isolated from the serum of mice following drug-free FUS-BBBO in both mice and patients. In AD mice and patients, an average EV concentration increase of 164% and 100% 1 h after FUS-BBBO containing both amyloid and tau was, respectively, found compared to 0% under sham conditions and highly correlated with BBBO volume. Whole genome RNA sequencing and mass spectrometry protein identification demonstrated inflammatory and proliferation gene upregulation. These findings highlight the potential of FUS-BBBO to increase EV serum concentration towards early AD detection.

10:00–10:15 Break

10:15

1aBAa5. Microbubble-enhanced and activity-informed FUS liquid biopsy in infiltrating gliomas. Pavlos Anastasiadis (Neurosurgery, Univ. of Maryland School of Medicine, 670 W Baltimore St., Baltimore, MD 21201, panast@som.umaryland.edu), Chetan Bettegowda, and Graeme F. Woodworth (Neurosurgery, Univ. of Maryland School of Medicine, Baltimore, MD)

In the absence of clinically available biomarkers for CNS malignancies, the conventional method for disease monitoring in glioma patients is radiologic. While most cancers shed cell-free molecules of tumor-derived DNA into the circulation, brain tumors form the exception to this rule, rarely shedding detectable levels of tumor DNA into the bloodstream. Microbubble-enhanced focused ultrasound (MB-FUS) is a rapidly emerging clinical tool to safely open the blood-brain barrier (BBB) for enhanced neurotherapeutic delivery and liquid biopsy. Previous work demonstrated MB-FUS treatments in infiltrating gliomas using multi-transducer systems with embedded acoustic emissions (AEMs) monitoring. In this study, we analyzed serum samples from a Phase 1 clinical trial of combined MB-FUS BBB opening with adjuvant temozolomide treatment of the tumor-infiltrated brain regions surrounding the surgical resection cavity of patients with newly diagnosed glioblastoma. The analysis evaluated the technical and treatment parameters associated with BBB opening, circulating DNA levels, and other biomarkers of MB-FUS effects. We analyzed serum samples using Real-SeqS and ELISA to quantify biomarker levels in longitudinal serum fluid samples derived from 13 patients undergoing repeated cycles of MB-FUS plus temozolomide. Our work and others in the field establish the potential for MB-FUS-enabled, non-invasive biopsy of gliomas, which would fundamentally advance the diagnosis, monitoring, and management of brain tumors.

10:45

1aBAa6. Monitoring gene expression in the brain with synthetic serum markers. Jerzy O. Szablowski (Bioengineering, Rice Univ., 6500 Main St., MS 142 BRC 869, Houston, TX 77005, js170@rice.edu), Joon Pyung Seo (Appl. Phys. Program, Rice Univ., Houston, TX), Sangsin Lee, Shirin Nouraein, Zhimin Huang, James Kwon, James S. Trippett, and Ryan Wang (Bioengineering, Rice Univ., Houston, TX)

Blood tests are among the most common clinical tools due to their low cost, simplicity, and ability to observe many markers at once. However, currently, blood tests can only monitor a fraction of physiological processes that happen to have a serum marker. Here, we will present our work on the development of synthetic serum markers that can track gene expression in intact brain cells with a simple blood test. The synthetic marker approach has several advantages. First, detection of existing markers may be challenging due to their low levels in blood. With a synthetic marker, one can design a reporter that is easy to detect allowing for superior sensitivity. Second, there are large numbers of genes in each cell, but available brain imaging techniques can at most represent only a few signals (e.g., a few colors of fluorescent proteins, or types of MRI contrast). Synthetic serum markers use biochemical detection and can be designed to be massively multiplexed similarly to how thousands of proteins can be detected in blood simultaneously with mass spectrometry. Third, synthetic serum markers can surveil large brain regions, unlike invasive locally implanted devices. Finally, these markers have the potential for inexpensive and simple readout. We show applications of this concept with markers that can cross through an intact blood–brain barrier to report on gene expression and markers whose release from the brain relies on sonobiopsy to provide spatial selectivity.

11:15

1aBAa7. Monitoring of ultrasound targeted drug delivery with liquid biopsy. Victor Menezes, Hohyun Lee (School of Mech. Eng., Dept. of Biomedical Eng., Georgia Inst. of Technol., Atlanta, GA), and Costas Arvanitis (School of Mech. Eng., Dept. of Biomedical Eng., Georgia Inst. of Technol., 311 Ferst Dr. Northwest, Atlanta, GA 30332, costas.arvanitis@gatech.edu)

The characterization of brain cancer genetic and molecular characteristics along with monitoring their response to therapeutic interventions is critical for effective treatment. Liquid biopsy—a minimally invasive technique that uses cancer soluble markers, such as cell-free DNA, in blood samples—offers a unique method to characterize the molecular profile of brain tumors and monitor the response treatment, as their location limits the effectiveness of traditional methods, such as needle biopsy. However, due to the limited secretion of cancer soluble molecules from the tumor core and the bloodstream, presumably due to the presence of the blood–brain barrier (BBB), liquid biopsy suffers from low sensitivity. Here, we show that microbubble enhanced focused ultrasound (MB-FUS) can significantly enhance the secretion of macromolecules from the tumor core to the blood. We also observed that MB-FUS in combination with liquid biopsy can be used to assess chemotherapy effect on cell death type in brain tumors. Together our findings highlight the potential of liquid biopsy in combination with MB-FUS to overcome challenges associated with the limited secretion of cancer soluble molecules in the circulation and support its application for monitoring targeted drug delivery in brain tumors.

Contributed Paper

11:45

1aBAa8. Evaluation of sonobiopsy feasibility and safety in a mouse model of diffuse intrinsic pontine glioma. Dingyue Zhang (Washington Univ. in St. Louis, 1 Brookings Dr., St. Louis, MO 63130, dingyue@wustl.edu), Yan Gong, Leqi Yang, Kevin Xu, Yimei Yue, Jinyun Yuan, and Hong Chen (Washington Univ. in St. Louis, St. Louis, MO)

Diffuse intrinsic pontine glioma (DIPG) is a leading cause of pediatric brain cancer mortality, emphasizing the need for precise molecular diagnosis to advance therapeutic development. The tumor's location in the brainstem poses challenges for invasive biopsy, prompting the exploration of liquid biopsies. However, both blood-based and cerebrospinal fluid (CSF)-based liquid biopsies exhibit limited sensitivity in diagnosing DIPG. Sonobiopsy, an innovative technique utilizing focused ultrasound and microbubbles, aims to enhance the sensitivity of liquid biopsies. This study assessed the feasibility and safety of sonobiopsy in a DIPG mouse model. The model was established by intracranial injection of enhanced green-fluorescent-protein (eGFP) transduced DIPG tumor cells. Mice were divided into sonobiopsy and conventional liquid biopsy groups. Sonobiopsy, employing MRI-guided focused ultrasound and microbubbles, demonstrated increases in plasma and CSF eGFP DNA, RNA, and protein concentrations compared to conventional liquid biopsy. The enrichment effect varied based on the biomarker type. No observed hemorrhage or tissue damage attributed to sonication suggests the safety of this approach. These results affirm the feasibility and safety of sonobiopsy in enriching DIPG-specific biomarkers in both plasma and CSF, suggesting sonobiopsy as a promising noninvasive molecular diagnostic tool for DIPG.

MONDAY MORNING, 13 MAY 2024 ROOM 212, 8:55 A.M. TO 11:50 A.M.

Session 1aBAb

Biomedical Acoustics, Physical Acoustics, Signal Processing in Acoustics, and Engineering Acoustics: Ultrasound Brain and Super-Resolution Imaging I

Chengzhi Shi, Cochair
School of Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr. NW, Atlanta, GA 30332

Fabian Kiessling, Cochair
Exp. Mol. Imaging, RWTH Aachen Univ., Forckenbeckstrasse 55, n.a., Aachen 52074, Germany

Chair's Introduction—8:55

Invited Papers

9:00

1aBAb1. Super-resolution ultrasound imaging with monodisperse microbubbles in a chicken embryo model. Redouane Ternifi, Alexis Vivien, Anne Lassus, Mina Lykakis, Alexandre Helbert (Res. and Development, Bracco, Plan-les-Ouates, Switzerland), Victor Jeannot (Res. and Development, Bracco, Rte. de la galaise, 31, Plan les Ouates 1228, Switzerland, victor.jeannot@bracco.com), and Emmanuel Gaud (Res. and Development, Bracco, Plan-les-Ouates, Switzerland)

ULM is a super-resolution imaging method that has transformed ultrasound imaging by beating the diffraction limit, enabling the visualization of blood vessel down to the capillary size. The development of innovative ultrasound responsive agents may allow to further improve the performance of this technology. Bracco is engaged in the formulation and the evaluation of various ultrasound responsive agents for ULM including monodisperse microbubbles. Our recent studies have shown that monodisperse microbubbles increase imaging sensitivity by an order of magnitude in comparison to polydisperse microbubbles. The benefit of this for ultrasound localization microscopy (ULM) has been tested *in vitro* in flow phantom and in a chicken embryo chorio-allantoic membrane (CAM) model. The performances of the monodisperse versus polydisperse formulations were evaluated through a collection of numeric metrics extracted from the super-resolved reconstructed CAM images, including not only quantitative criteria, such as the number of detected microbubbles or the rate of successful tracking, but also geometrical considerations such as vessel density or number of segments of the reconstructed vasculature. The study showed that monodisperse formulations outperformed the other polydisperse formulations, particularly using low to moderate microbubble concentrations.

9:20

1aBAb2. Functional ultrasound localization microscopy in the murine brain: Challenges and new techniques. Yike Wang, YiRang Shin (Electr. and Comput. Eng., Univ. of Illinois Urbana-Champaign, Urbana, IL), Qi You (Bioengineering, Univ. of Illinois Urbana-Champaign, Urbana, IL), Bing-Ze Lin, Matthew R. Lowerison (Electr. and Comput. Eng., Univ. of Illinois Urbana-Champaign, Urbana, IL), and Pengfei Song (Elec. and Comput. Eng., Univ. of Illinois Urbana-Champaign, 405 N. Mathews Ave., Beckman Inst. 4041, Urbana, IL 61801, songp@illinois.edu)

Functional ultrasound localization microscopy (fULM) is a new technique that combines the principles of ULM and functional ultrasound (fUS) to achieve brain-wide and micrometer-scale mapping of brain neural activities based on neurovascular coupling. The unique combination of high imaging spatial resolution, large imaging field-of-view, and deep imaging depth of penetration makes fULM a potentially transformative technology for numerous neuroscience applications where activities from both global neural networks and local neurocircuits need to be recorded simultaneously and continuously. At present, however, fULM suffers from many technical and pragmatic challenges, including low sensitivity and specificity to neural activities, the need of long data acquisition with continuous infusion of microbubbles and repeated simulations, and the lack of viable 3D imaging solutions that are essential for neuroscience research. In this presentation, I will first introduce the principles and technical challenges of fULM, followed by recent advances achieved by our group including (1) enhanced microbubble localization, tracking, and other post-processing techniques to boost fULM's sensitivity to neural activities; (2) 3D fULM based on 2D matrix arrays that are compatible with mainstream 256-channel ultrasound systems; and (3) an awake fULM imaging platform for mice and rats that allows whole-brain, microscopic-scale recording of functional neural activities in awake animals.

9:40

1aBAb3. Monitoring of neoadjuvant chemotherapy response of breast cancer with ultrasound localization microscopy. Celine Porte (Exp. Mol. Imaging, RWTH Aachen Univ., Aachen, Germany), Matthias Kohlen (Dept. of Gynecology and Obstetrics, Univ. Clinic Aachen, RWTH Aachen Univ., Aachen, Germany), Thomas Lisson (Med. Eng., Dept. of Elec. Eng. and Information Technol., Ruhr Univ. Bochum, Bochum, Germany), Zuzanna Magnuska (Exp. Mol. Imaging, RWTH Aachen Univ., Aachen, Germany), Brigitte Sophia Winkler (Dept. of Gynecology and Obstetrics, Univ. Clinic Aachen, RWTH Aachen Univ., Aachen, Germany), Anne Rix (Exp. Mol. Imaging, RWTH Aachen Univ., Aachen, Germany), Stefanie Dencks (Med. Eng., Dept. of Elec. Eng. and Information Technol., Ruhr Univ. Bochum, Bochum, Germany), Patrick Koczera (Exp. Mol. Imaging, RWTH Aachen Univ., Aachen, Germany), Georg Schmitz (Med. Eng., Dept. of Elec. Eng. and Information Technol., Ruhr Univ. Bochum, Bochum, Germany), Elmar Stickeler (Dept. of Gynecol. and Obstetr., Univ. Clinic Aachen, RWTH Aachen Univ., Aachen, Germany), and Fabian Kiessling (Exp. Mol. Imaging, RWTH Aachen Univ., Forckenbeckstrasse 55, n.a., Aachen 52074, Germany, fkiessling@ukaachen.de)

Neoadjuvant chemotherapy (NAC) is the standard treatment for high-risk breast cancer, but less than 30% of patients exhibit a complete response, necessitating better preselection and monitoring. Ultrasound localization microscopy (ULM) offers potential improvements by providing super-resolution images of vasculature and functional features. To assess its efficacy, we applied ULM to breast cancer patients undergoing NAC, comparing their responses based on ULM features. Patients were evaluated before their first, second, and fourth NAC cycles, assessing tumor volume and employing contrast-enhanced ultrasound (CEUS) in split-screen mode. CEUS videos underwent postprocessing, involving MB localization and tracking, and motion compensation. ULM images, derived from tracked MBs, generated parameter maps for analyzing vascular changes and differences between responders and non-responders. Preliminary results of 14 patients revealed that during the first three cycles of NAC, the tumor volume as well as vessel coverage (i.e., ratio of vessel to tumor surface) of responders noticeably dropped. In contrast, non-responders showed only minor tumor size reduction and overall no decrease in coverage. Furthermore, the mean distance to the closest track was noticeably higher in non-responders. These findings, though preliminary, hint at ULM's potential for enhancing preselection and monitoring of NAC patients.

10:00

1aBAb4. Model-based deep learning for ultrasound localization microscopy. Tristan S. Stevens (Elec. Eng., Eindhoven Univ. of Technol., 's Gravesandestraat 7s, Eindhoven, Noord-Brabant 5612JM, Netherlands, t.s.w.stevens@tue.nl), Ben Luijten, Ivar R. de Vries, and Ruud J. Sloun (Elec. Eng., Eindhoven Univ. of Technol., Eindhoven, Noord-Brabant, Netherlands)

Ultrasound localization microscopy (ULM) effectively visualizes vasculature through localization of microbubbles using ultrasound. However, challenges arise from the domain gap between clean data from controlled lab environments and real-world scenarios. Factors, such as high bubble densities, reduced frame rates, and poor image quality (e.g., due to aberration), impede clinical adoption. Traditional methods face limitations in handling the challenging nature of ULM, which have spurred the integration of data-driven techniques. We specifically focus on model-based deep learning, emphasizing statistical inference techniques for increased robustness on out-of-distribution data. The localization task is expressed as an inverse problem $\mathbf{y} = \mathbf{H}\mathbf{x} + \mathbf{n}$, where $\mathbf{y}$, $\mathbf{x}$, and $\mathbf{n}$ represent the observed ultrasound signal, underlying microbubble localizations, and noise. Inaccuracies in the measurement matrix $\mathbf{H}$ arise from oversimplified ultrasound acquisition models (i.e., aberration distorting the point-spread function), necessitating the application of spatial priors to regularize the forward model. The ISTA algorithm can enforce sparsity on $\mathbf{x}$ and help solve the ill-posed inverse problem. A deep learning extension, namely learned ISTA, can additionally mitigate errors in the forward model. Beyond spatial priors, imposing temporal priors on individual microbubbles enhances accuracy. KalmanNet, integrating Kalman filtering with deep learning, exemplifies this approach by effectively learning the temporal dynamics of microbubbles.

10:20–10:35 Break

10:35

1aBAb5. Towards reduced ultrasound localization microscopy acquisition time by means of monodisperse microbubbles uncoupling. Giulia Tuccio (DISI, Univ. of Trento, via Sommarive, 5, Povo, Trento 38123, Italy, giulia.tuccio@unitn.it), Lisa te Winkle, Wim van Hoeve (Solstice, Enschede, Netherlands), and Libertario Demi (Information Eng. and Comput. Sci., Univ. of Trento, Trento, Italy)

Ultrasound localization microscopy (ULM) unveils the microvascular structures using microbubbles (MBs) flowing in the circulatory system. As ULM relies on the precise localization and tracking of individual MBs, using high MB concentrations yields to high localization errors and ULM failure. ULM is, therefore, constrained to low MBs concentrations, leading to long acquisition times. To tackle this limitation, in this study, we propose an approach based on the injection of distinct monodisperse MBs, each characterized by a specific resonance behavior. As a proof of concept, we acquired and analyzed ultrasound data from a vascular phantom, where we singularly injected two monodisperse MB populations. Data were collected using an ULA-OP equipped with an Esaote LA533 probe. Pulses with a center frequency of 3 MHz and a bandwidth of 1MHz were utilized for imaging. MBs with diameters equal to 2.5 and 4.1 μm were injected in the phantom. MBs uncoupling was then performed exploiting the differences in the second harmonic signal intensity generated by the two MB populations. Results demonstrate the feasibility of monodisperse MBs uncoupling, enabling the use of higher microbubble concentrations for ULM, and thus, reducing acquisition time.

10:50

1aBAb6. Acoustic emissions based estimation of the temporal changes in microbubble radius during ultrasonic excitation. Hohyun Lee (Mech. Eng., Georgia Inst. of Technol., 901 State St. NW, Atlanta, GA 30332, hlee649@gatech.edu) and Costas Arvanitis (School of Mech. Eng., Dept. of Biomed. Eng., Georgia Inst. of Technol., Atlanta, GA)

Our ability to study microbubble dynamics *in vivo* and link them to distinct mechano-biological effects hinges on our ability to accurately estimate the temporal changes in MB radius during ultrasonic excitation. Here, we hypothesize that real-time passive cavitation detection (PCD) monitoring combined with linear acoustic wave propagation theory can accurately estimate stable MB radius dynamics under *in vivo* conditions. To test our hypothesis, we employed numerical simulations, based on Rayleigh–Plesset modeling, followed by experimental validation, using calibrated PCDs with concurrent optical imaging of MB dynamics using high frame rate microscopy. Our method termed linear acoustic wave propagation and superposition (LAWPS) algorithm, combines Fourier series expansion with Euler's relationship to estimate the acoustic emission (AE) from single MB, which is considered as a monopole source (i.e., $R_0 + \Delta R < \lambda$). LAWPS algorithm, which is independent of MB properties and, thus, can be linearly reversed to calculate MB oscillation radius from AE, was able to accurately capture the radiated pressure generated from a Rayleigh–Plesset MB modeling. Crucially, inverse LAWPS algorithm onto AE generated by Vokurka *et al.* resulted in the original MB model. Experimental observation using monodisperse MBs was able to accurately estimate the temporal changes in MB radius during 0.5 MHz ultrasonic excitation

11:05

1aBAb7. Investigating the resonance response of a system of two ultrasound-driven lipid encapsulated microbubbles confined within a viscoelastic vessel. Hossein Yusefi (Phys., Concordia Univ., 7141 Sherbrooke St. W, Dept. of Phys., Concordia University, Montreal, QC H4B 1R6, Canada, hossein.yusefi@concordia.ca) and Brandon Helfield (Phys., Concordia Univ., Montreal, QC, Canada)

Lipid encapsulated microbubbles, strongly characterized by the initial phospholipid concentration, are clinical ultrasound imaging agents. These microbubbles are commonly used as an ultrasound contrast agent and are being developed for therapeutic applications. In this work, we investigate the vibration dynamics of two microbubbles confined within a viscoelastic vessel. We developed a finite element model to study radial microbubble dynamics in a two-bubble system, as typical clinical doses result in closely spaced bubbles. Specifically, we study the effect of the vessel wall viscosity on the resonance behaviour of microbubbles in the frequency range of 2–8 MHz. For two identical microbubbles, we observed a decrease in 50% in resonance amplitude as we increased the vessel viscosity from 0.1 to 1 Pa s as well as an 8% shift of peak resonance activity towards higher frequencies. Furthermore, we investigated the same system consisting of microbubbles with a lower initial phospholipid concentration and showed the same decrease in amplitude; however, the direction of shift of peak resonance was towards lower frequencies by 4%. Our work suggests that microbubble resonance behaviour is greatly affected by the vessel viscosity and that the extent and direction of bubble dynamics changes are dependent on shell characteristics.

11:20

1aBAb8. Distortion map and correction for ultrasound localization microscopy imaging with a row–column array. Joseph Thomas T. Hansen-Shearer (Bioengineering, Royal School of Mines, Imperial College London, London SW7 2AZ, United Kingdom, jh12718@ic.ac.uk) and Meng-Xing Tang (Bioengineering, Imperial College London, London, United Kingdom)

The row–column array is an emerging technology which can enable large fields of view without the need for a large number of electronic channels. In ultrasound localization microscopy, sparse microbubbles are localized and tracked to produce high-resolution images of the microvasculature. During the localization of the microbubbles, it is typically assumed that the center of the target observed can be used as a substitute for the true microbubble location. However, the true microbubble will be located at the onset of the received signal. This distortion between the true location and the observed location will result in a shift of the microbubble away from the true location, thus altering the accuracy of the images produced. In most forms of ultrasound imaging, this shift mostly manifests as a vertical shift so can often be safely ignored. However, we have observed that in row–column array imaging this distortion will be spatially variable and will result in distortions in the axial, elevational, and lateral directions, which can be on the order of a few wavelengths. Consequently, this distortion, if ignored, will result in inaccuracies in the resulting ultrasound localization microscopy images. Here, we present a technique to map and correct for these distortions.

11:35

1aBAb9. A parametrization study of TEULM's image reconstruction performance. Giulia Tuccio (DISI, Univ. of Trento, via Sommarive, 5, Povo, Trento 38123, Italy, giulia.tuccio@unitn.it), Sajjad Afrakhteh (Univ. of Trento, Trento, Trentino, Italy), and Libertario Demi (Information Eng. and Comput. Sci., Univ. of Trento, Trento, Italy)

Time efficient ultrasound localization microscopy (TEULM) allows imaging the microvascular structures with reduced frame rates compared to standard ULM. The quality of reconstructed images relies on different parameters. In this study, we aim at understanding the links between TEULM performance and three key parameters: SVD k-value (representing the number of components considered as noise), the maximum linking distance (indicating the assumed maximum distance that a microbubble can cover between two successive frames), and the minimum length of a track. We generated TEULM density and dynamics maps, singularly varying these three parameters. Next, we evaluated the quality of TEULM images by means of a root mean squared error (RMSE), dice score, and Fourier ring correlation (FRC) analysis. A publicly available preclinical *in-vivo* dataset was used for this study. Results demonstrate that minimum track length and k-value have the greatest impact on both density and dynamic maps RMSE, whereas the minimum linking distance is the key factor influencing the dice score. Additionally, the relationship between these parameters and metrics is non-linear. Furthermore, the resolution measured by the FRC is negatively affected by increasing the SVD k-value and by increasing the maximum linking distance but positively affected by increasing the minimum track length.

Session 1aCA

Computational Acoustics, Biomedical Acoustics, Physical Acoustics, Engineering Acoustics, and Structural Acoustics and Vibration: Computational Methods for Acoustic Absorption in Materials

Shung H. Sung, Cochair
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D. Keith Wilson, Cochair
Cold Regions Res. and Eng. Lab., U.S. Army Eng. Res. and Dev. Center, U.S Army ERDC-CRREL, 72 Lyme Rd., Hanover, NH 03755-1290

Invited Papers

9:30

1aCA1. The acoustics of absorbers comprising a flexible perforated membrane backed by a stack of granular material. Zhuang Mo (Ray W. Herrick Laboratories/School of Mech. Eng., Purdue Univ., Ray W. Herrick Labs., Purdue University, 177 S. Russell St., West Lafayette, IN 47907-2099, mozh-mcfly@qq.com), Guochenhao Song (Ray W. Herrick Laboratories/School of Mech. Eng., Purdue Univ., West Lafayette, IN), Huawei Yang, Tongyang Shi (Inst. of Acoust. Chinese Acad. of Sci., Beijing, China), and J. S. Bolton (Ray W. Herrick Laboratories/School of Mech. Eng., Purdue Univ., West Lafayette, IN)

It has been found that when a layer of activated carbon partially fills the space behind a finite, edge-constrained, tensioned impermeable membrane, the absorption peaks due to the modal response of the membrane can be significantly enhanced in the low frequency range. In the present work, the modeling aspects of the latter work have been extended to allow the membrane to be both tensioned and flexurally stiff, and furthermore, to be micro-perforated, thus expanding the treatment design space. Second, the particle layer behind the membrane is modeled by using a two-dimensional finite difference implementation of the Biot poro-elastic theory which then accounts for the interaction of the particle layer and walls that contain it: i.e., the solid phase of the particle stack itself is allowed to exhibit modal behavior in the radial direction. The interaction of the membrane nearfield and the particle stack, which creates a nearfield damping effect, is also fully captured. Finally, the model accounts for the hierarchical porosity of activated carbon. The model has been verified by comparison with measurements and it has been found that strikingly high levels of low frequency absorption can be realized by appropriate optimization of the membrane and particle stack properties and geometry.

9:50

1aCA2. Granular activated carbon sound absorption predictions made using measured material parameters. Huawei Yang (Inst. of Acoust., Chinese Acad. of Sci., Beijing, China), Tongyang Shi (Inst. of Acoust., Chinese Acad. of Sci., No. 21 North 4th Ring Rd., Haidian District, Beijing 100190, China, shitongyang@mail.ioa.ac.cn), and J. S. Bolton (Ray W. Herrick Laboratories/School of Mech. Eng., Purdue Univ., West Lafayette, IN)

Granular activated carbon (GAC) is hierarchical porosity material that yields better in low frequency sound absorption performance compared with many traditional porous materials due to the sorption effect created by nanometer-scale pores within the particles. In 2017, a triple porosity model accounting for the sorption effect in the micropores was proposed by Venegas *et al*. In the proposed model, some of input parameters (i.e., mesopore size and micropore effective diffusion coefficient) were fitted by matching the model to measured the surface impedance of a GAC stack; however, some fitted values are questionable from an inorganic material perspective. In the present work, the GAC material parameters, such as micro- and meso-pore sizes, were estimated from a standard isotherm measurement. The measurement results showed a standard Langmuir type isotherm behavior at different temperatures: i.e., 273.15 and 293.15 K. Thus, if the isotherms can be described with the Langmuir model, then the Langmuir constant and heat of adsorption can be estimated based on the isotherms. Finally, these parameters were used as the input to the GAC model and the absorption coefficient was calculated. The calculated result was also compared with the absorption coefficient measured following the E1050 standard.

10:10

1aCA3. Improvement of sound absorption of porous materials through periodic holes of sinusoidal decreasing profile. Zacharie Laly (Dept. of Mech. Eng., Sherbrooke Univ., 2500, boul. de l'Université, Sherbrooke, QC J1K 2R1, Canada, zacharie.laly@usherbrooke.ca), Raymond Panneton (Dept. of Mech. Eng., Sherbrooke Univ., Sherbrooke, QC, Canada), Noureddine Atalla (Dept. of Mech. Eng., Sherbrooke Univ., Sherbrooke, QC, Canada), Sebastian Ghinet (Aerosp., Natl. Res. Council Canada, Ottawa, ON, Canada), and Chris Mechefske (Dept. of Mech. and Mater. Eng., Queen's Univ., Kingston, ON, Canada)

Porous materials used in several engineering applications for noise attenuation present poor acoustic performance at low frequencies. In this paper, a design of porous materials with various periodic decreasing hole profiles is presented and studied using the finite element method. The sound absorption coefficient and the normalized surface impedance are compared for different periodic decreasing hole profiles. A small drop in sound absorption is observed after the first peak with a decreasing linear profile. Using a sinusoidal decreasing

hole profile, it is shown that the drop is attenuated. The impacts of the amplitude and the period of the sinusoidal profile on the sound absorption coefficient and the surface impedance are demonstrated. The proposed porous material design with periodic holes having a sinusoidal decreasing profile shows a significant improvement in sound absorption over a large frequency band compared to conventional porous materials.

10:30–10:45 Break

10:45

1aCA4. Prediction of the sound pressure response in an enclosed cavity with sound absorbent walls using the CMA and AMA methods. Shung H. Sung (SHS Consulting, LLC, 4178 Drexel Dr., Troy, MI 48098, shsung1972@gmail.com) and Donald J. Nefske (DJN Consulting, LLC, Troy, MI)

The sound pressure response in an enclosed cavity with sound absorbent walls is predicted using the classical modal analysis (CMA) method and the asymptotic modal analysis (AMA) method. The sound absorbing material is represented as an equivalent fluid with frequency dependent bulk material properties. The equivalent fluid properties can be obtained from either impedance tube test data or based on micro-scale Biot material properties. In the low frequency range, the CMA method is implemented to include the frequency dependent material properties. In the medium and high frequency ranges, when modal density is high, the frequency dependence effects become cumbersome using CMA. The AMA method has been developed as an asymptotic extension of the CMA method for the mid- and high-frequency ranges and is extended here to incorporate the frequency dependence of an equivalent fluid properties of the sound absorbing materials. An enclosed rectangular box subjected to loudspeaker excitation is used to evaluate the effect of absorption materials using the CMA method for the low-frequency range and the AMA method for the mid- and high-frequency ranges. Test data have also been obtained for comparisons of the CMA method in the low frequency range. Using these procedures, the CMA and AMA methods can be applied to represent absorbent materials over a wide frequency range for enclosure architectural design.

11:05

1aCA5. The impact of non-monotonic relaxation on numerical accuracy and stability of viscoelastic finite elements. Eric Abercrombie (Dept. of Mech. Eng., Boston Univ., Boston, MA 02215, abere@bu.edu) and James G. McDaniel (Mech. Eng., Boston Univ., Boston, MA)

Analysis of sound and vibration in linear viscoelastic materials is based on a relaxation function that is the time-dependent stress resulting from a unit step in strain. It is well established that relaxation functions decrease monotonically in time. The most common constitutive relaxation functions, such as the Kelvin–Voigt or General Maxwell Model, produce curves that must be monotonic. However, recently, the authors have produced a methodology that allows the use of discrete data for the time-dependent relaxation function. One result of this approach is the possibility of non-monotonic relaxation function input. This non-monotonic data may be the result of test and measurement errors. Alternatively, this non-monotonicity may be an actual anomalous physical result, such as in rock salt (He *et al.* 2019). The authors use validated viscoelastic finite element models in the time domain to determine the instability caused by such non-monotonic datasets. The study uses randomized non-monotonic datasets that maintain the fundamental material trend to find unstable or inaccurate results. A mathematical review highlights the conditions causing numerical instability in modern stepwise time integration solvers. The results provide clarity for viscoelastic analysts considering low-precision viscoelastic measurements or unusual physical properties. [Work supported by ONR under Grant N00014-22-1-2785.]

11:25

1aCA6. Incorporating the effect of frequency-independent attenuation within the on-axis spatial impulse response of a circular piston. Robert J. McGough (Elec. and Comput. Eng., Michigan State Univ., 428 S. Shaw Ln., Rm. 2120, East Lansing, MI 48824, mcgough@egr.msu.edu)

The spatial impulse response is important for numerical simulations of diagnostic ultrasound. The spatial impulse response, which describes transient diffraction due to an impulsive input, yields closed form analytical expressions for various transducer geometries when the medium is lossless. These analytical expressions are advantageous for simulations that repeatedly evaluate these expressions at hundreds of thousands of points. However, spatial impulse responses evaluated for lossy materials typically require additional numerical calculations that substantially increase the computation time. Thus, analytical or rapidly converging numerical expressions for the lossy spatial impulse response are expected to greatly enhance present simulation methods. This motivates the derivation of on-axis spatial impulse responses for a circular piston that model frequency-independent attenuation. Closed-form analytical expressions for the on-axis spatial impulse response are introduced for two closely related frequency-independent attenuation models. The results show that, as the attenuation constant increases, the peak amplitudes of these lossy on-axis spatial impulse responses decrease. The lossy on-axis impulse response also decreases slightly as time increases beyond the initial arrival time, whereas the lossless on-axis spatial impulse response for a circular piston maintains a constant value after the initial arrival time.

Contributed Paper

11:45

1aCA7. Exploring integral bounds for sound absorption: A comprehensive study. Zhe Zhang (Mech. Eng., The Univ. of Hong Kong, Rm. 210, Haking Wong Bldg., Pokfulam Rd., Hong Kong 999077, Hong Kong, zz86736@connect.hku.hk), Ying Hu, Bohua Huang, Xue Han, and Lixi Huang (Mech. Eng., The Univ. of Hong Kong, Hong Kong, Hong Kong)

Sound absorption across a wide range of frequencies is a focus in contemporary acoustics. Recently, integral bounds of absorption or reflection coefficients were introduced as a guide of design optimization following the footsteps of electromagnetics, where integral relations were derived based on system causality considerations. This talk examines the relation between the integral bound and the effects of various physical boundary conditions and the bulk absorber material, with the hope to raise the bound. Effects of various approximations made during mathematical derivation are also examined by comparison with specific numerical examples, and the physics of the bound is thoroughly discussed. The findings are expected to have significant implications for the development of effective noise reduction strategies and the advancement of smart acoustic design.

MONDAY MORNING, 13 MAY 2024 — ROOM 201, 8:00 A.M. TO 11:55 A.M.

Session 1aID

Interdisciplinary and Student Council: Introduction to Technical Committees

Brijonnay Madrigal, Cochair
Marine Mammal Res. Prog., Univ. of Hawai'i at Manoa, 46-007 Lilipuna Rd., Kaneohe, HI 96744

Ian C. Bacon, Cochair
Phys. & Astron., Brigham Young Univ., 333 W 100 S, Provo, UT 84601

Natalie Kukshtel, Cochair
Appl. Ocean Phys. & Eng., Woods Hole Oceanographic Inst., 98 Water St., Woods Hole, MA 02543

Heui Young Park, Cochair
Pennsylvania State Univ., State College, PA 16801

Invited Papers

8:00

1aID1. Architectural acoustics: Buildings and beyond. David S. Woolworth (Roland, Woolworth & Assoc., 356 County Rd. 102, Oxford, MS 38655-8604, dwoolworth@rwaconsultants.net)

Architectural acoustics not only covers buildings and the environment around them but also human perception of the acoustic environment, indoors and outdoors. As a technical committee of the acoustical society, our members are spread over research, academia, practitioners and industry. Architectural acoustics is not reserved for concert halls and opera houses but applies to all occupied spaces and has a direct impact on quality of life of any user of the space. Specific topics within the discipline include but are not limited to environmental sound, speech privacy, and speech intelligibility, simulated acoustic environments, annoyance, human hearing, airborne and structureborne noise, sound and impact isolation, loudspeakers and microphones, room acoustics, soundscape, and acoustical measurements. The technical committee on noise is often a cosponsor of specials sessions by the TCAA, as noise control via architectural means is common practice. This presentation will provide an overview of the TCAA and the field of architectural acoustics and provide examples of current research and projects of interest.

8:20

1aID5. ASA's computational acoustics Technical Committee. D. Keith Wilson (Cold Regions Res. and Eng. Lab., U.S. Army Engineer Res. and Development Ctr., U.S. Army ERDC-CRREL, 72 Lyme Rd., Hanover, NH 03755-1290, D.Keith.Wilson@usace.army.mil)

Computational acoustics is in its second full year as a Technical Committee, after being promoted from a Technical Specialty Group in May 2022. The CA TC provides a forum for researchers interested in numerical methods pertinent to acoustic wave propagation and structures, data analytics, validation, optimization, visualization, as well as application of computational models to engineering, noise control, and other practical problems. Some areas of current and emerging interest include artificial intelligence, uncertainty characterization, model reduction techniques, efficient parallelization, and time-domain treatments of dissipation. The CA TC is also interested in building collaborations with other TCs on repositories and benchmarks for codes and data.

8:40

1aID2. An introduction to the Technical Committee on animal bioacoustics. Laura Kloepper (Dept. of Biological Sci., Univ. of New Hampshire, 230 Spaulding Hall, Durham, NH 03824, laura.kloepper@unh.edu)

Animal bioacoustics as a field of research involves the study of sound in non-human animals. The range of this field is wide and includes all aspects of sound production and reception, communication and associated behaviors, acoustic ecology and effects of noise/sound, and passive and active acoustic methods for monitoring individuals, populations, habitats, and ecosystems. Members of the Technical Committee on Animal Bioacoustics (TCAB) come from diverse backgrounds, including those with training in biology, ecology, engineering, mathematics, oceanography, physics, and psychology. Many TCAB members also participate in the activities of other ASA Technical Committees including Acoustical Oceanography, Engineering Acoustics, Noise, Psychological and Physiological Acoustics, Signal Processing, and Underwater Acoustics, reflecting the interdisciplinary nature of the field. This talk will highlight some of the popular and emerging areas of research in Animal Bioacoustics.

9:00

1aID3. Introduction to the Acoustical Oceanography Technical Committee. David R. Barclay (Dept. of Oceanogr., Dalhousie Univ., PO Box 15000, Halifax, NS B3H 4R2, Canada, dbarclay@dal.ca)

The Acoustical Oceanography Technical Committee is responsible for representing and fostering Acoustical Oceanography within the Acoustical Society of America. It is concerned with the development and use of acoustical techniques to measure and understand the physical, biological, geological, and chemical parameters and processes of the sea. Several acoustical methods are used to quantitatively study various oceanographic processes. Approaches include ocean parameter estimation by acoustical methods, remote sensing by passive and active acoustics, acoustic imaging, inversion, and tomography, and developing acoustical instrumentation for oceanographic studies.

9:20

1aID4. Capillaries, cancer, and cavitation: An introduction to the Biomedical Acoustics Technical Committee. Julianna C. Simon (Penn State Univ., Penn State 201E Appl. Sci. Bldg., University Park, PA 16802, jcsimon@psu.edu)

Biomedical acoustics is a growing field comprised of individuals who study the interaction of sound with biological materials. While ultrasound in medicine is most associated with fetal imaging, researchers and clinicians continue to push the boundaries of ultrasound to improve the diagnosis and treatment of disease. As such, this conference features special sessions on imaging the microvasculature in the brain and beamforming to improve the speed and quality of the ultrasound image. But ultrasound can also be used as a therapeutic due to absorption of the acoustic wave, radiation forces, and cavitation. On the therapy side, special sessions have been organized on: phase-change contrast agents, which can improve imaging contrast or locally deliver medications to cells; sonodynamic therapy, which involves killing cancer cells with ultrasound; sonobiopsy, which uses ultrasound to enhance the release of biomarkers for noninvasive disease diagnosis; and wave propagation in complex media, which helps us to understand our acoustic field for imaging and therapy. BATC has also organized a session on training students in technical writing. Ultrasound is an important diagnostic and treatment modality that continues to advance as researchers and physicians work to improve patient outcomes.

9:40

1aID6. Introduction to the Technical Committee on Engineering Acoustics. Michael R. Haberman (Walker Dept. of Mech. Eng. & Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, haberman@utexas.edu)

This talk will introduce ongoing work in the Technical Committee on Engineering Acoustics (TCEA) of the Acoustical Society of America. Engineering Acoustics encompasses the theory and practice of creating tools to generate and investigate acoustical phenomena and then to apply the knowledge of acoustics to practical utility. This includes the design and modeling of acoustical and vibrational transducers, transducer arrays, and transduction materials or systems in all media and frequency ranges. It is also concerned with the design of acoustical instrumentation, metrology, and the calibration of those systems. It further considers all aspects of measurement, fabrication, and computational techniques as they relate to acoustical phenomena and their utility. The talk will provide an introduction of a broad range of research topics in TCEA, with specific highlights on exciting new areas of research.

10:00–10:15 Break

10:15

1aID7. Overview of the technical area in musical acoustics. Jonas Braasch (School of Architecture, Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, braasj@rpi.edu), Andrew A. Piacsek (Phys., Central Washington Univ., Ellensburg, WA), and Gary Scavone (Music Res., McGill Univ., Montreal, QC, Canada)

Musical acoustics was launched as one of the first Technical Committees of the Acoustical Society of America. The Technical Committee in Musical Acoustics (TCMU) is concerned with applying science and technology to the field of music. The four main areas are (1) physics of musical sound production in musical instruments and the voice, (2) music perception and cognition, (3) analysis and synthesis of musical sounds and compositions, and (4) recording and reproduction technology. The scopes of areas have changed over time; for example, current interests in using groundbreaking methods in artificial intelligence and computational acoustics to solve problems. There is substantial interdisciplinary overlap with other technical committees, such as Architectural Acoustics and Physiological and Psychological Acoustics. Musical acoustic studies sometimes only require relatively moderate equipment. Thus, they lend themselves well as a research entry point for undergraduate and even high school students—especially since there is often a natural interest in music from early on. However, research in the field can also become very complex and often requires cultural understanding and listening skills to interpret technical results and direct research meaningfully. On the practical side, the TCMU sometimes organizes concerts to augment the technical sessions.

10:35

1aID8. Highlights from the technical committee on noise. Alexandra Loubeau (NASA Langley Res. Ctr., MS 463, Hampton, VA 23681, a.loubeau@nasa.gov) and Logan T. Mathews (Brigham Young Univ., Provo, UT)

The Technical Committee on Noise (TCNS) includes researchers and industry practitioners interested in understanding noise generation from a variety of sources, its propagation, and its impact on structures, objects, and people. The goal of reducing the impact of this noise, or unwanted sound, has led to innovative model development, measurement techniques, mitigation strategies, and input towards the establishment of regulations. Community noise definition and abatement is of particular interest and motivates the advancement of research on a variety of topics, such as transportation noise from traditional and new vehicles, hearing protection, jet and rocket noise, building systems noise and vibration, atmospheric sound propagation, soundscapes, and low-frequency sound. The diverse technical background of TCNS members is essential, given the interdisciplinary nature of this research, and it is mirrored in the topics of special sessions typically held jointly with other ASA technical committees.

10:55

1aID9. An introduction to psychological and physiological acoustics. G. Christopher Stecker (Ctr. for Hearing Res., Boys Town Natl. Res. Hospital, 555 N 30th St., Omaha, NE 68131, cstecker@spatialhearing.org)

The technical committee on Psychological and Physiological acoustics (P&P) is concerned with a wide and multidisciplinary range of topics. P&P members address questions of how the auditory system transforms and processes incoming sound and how sound is used to facilitate behaviors like communication and navigation. Physiological acoustics concerns topics, such as the mechanical and acoustical mechanisms of the middle and inner ear and neurophysiology of peripheral and central auditory systems. Psychological acoustics concerns, for example, behavioral studies of auditory perception and cognition. Within these topics, research covers the basic science of normal function and hearing disorders, as well as clinical, translational, and applied questions related to hearing aids, cochlear implants, and audio technology. Membership and activities of P&P overlap heavily with several other technical committees in ASA, especially Architectural Acoustics, Animal Bioacoustics, Noise, and Speech Communication. This presentation will feature a brief overview and several examples of P&P research.

11:15

1aID10. Introducing the Structural Acoustics and Vibration Technical Committee. Christina J. Naify (Appl. Res. Labs: UT Austin, 4555 Overlook Ave. SW, Washington, DC 20375, christina.naify@gmail.com)

The Structural Acoustics and Vibrations Technical Committee (TCSA) includes scientific study of vibrating structures, excited using either elastic or acoustic waves, and radiated acoustic fields from those structures. The committee members study a diverse range of physics relating to these basic phenomena, from damping and isolation, to active control, to modal response and a variety of techniques are used, including numerical modeling, analytical techniques, and experimental measurements. Due to the wide range of applications in which vibrating structures are found, TCSA is multidisciplinary, with many of our meeting sessions co-chaired across a wide range of complimentary TCs. Additionally, research fields span a range of disciplines in their practice, including industry, academia. and government. This talk will provide a brief overview into TCSA including historical highlights and future directions as well as summaries of some of the recent special sessions presented at Acoustical Society meetings.

11:35

1aID11. Introducing speech communication. Benjamin V. Tucker (Commun. Sci. and Disord., Northern Arizona Univ., Flagstaff, AZ, benjamin.tucker@nau.edu)

Research on speech communication investigates how speech is produced and how it is perceived. Researchers and practitioners who engage with speech communication come from various disciplines, such as audiology, computer science, engineering, linguistics, psychology, and speech pathology. Speech communication researchers seek to understand the acoustic signal as produced by a speaker or perceived by a listener. Research under the speech communication technical committee overlaps with other technical committees. In this presentation, I provide brief illustrations of various topics within speech communication, focusing on how people perceive and produce speech to communicate in the world's languages.

Session 1aPA

Physical Acoustics, Computational Acoustics, Structural Acoustics and Vibration, and Engineering Acoustics: Developments and Applications in Phononic Crystals

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

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

Chair's Introduction—8:00

Invited Papers

8:05

1aPA1. Unilateral transmission in bilinearly coupled systems. Ali Kogani (Concordia Univ., Montreal, 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)

This work investigates the steady-state vibration transmission characteristics of phononic lattices featuring bilinear coupling forces. We focus specifically on the phenomenon of unilateral transmission, which occurs when the transmitted vibrations maintain either complete compression or tension; i.e., the transmitted displacement does not change signs and remains either positive or negative. We consider bilinearly coupled oscillators as the unit cell for a bilinear phononic lattice. We conduct a parametric study for the unit cell and determine the boundary for the onset of unilateral transmission. We then extend the analysis to a bilinear phononic lattice. In both cases, we compute the steady-state response and linear stability of the system, considering bilinear forces with both amplitude-dependent and amplitude-independent properties. We identify regimes of nonreciprocal unilateral transmission when the mirror symmetry of the unit cell is broken. Our findings contribute to a better understanding of unilateral transmission in systems with bilinear elasticity.

8:25

1aPA2. Towards Dirac-like physics in ultrasonic crystals. Nicholas Gangemi (Phys. Acoust., U.S. Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375, nicholas.gangemi@nrl.navy.mil), Caleb F. Sieck (Code 7160, U.S. Naval Res. Lab., Washington, DC), Joseph Vignola, Diego Turo (Mech. Eng., The Catholic Univ. of America, Washington, DC), Jeffrey Baldwin, Steven Liskey, Aaron Edmunds, William Wilson, Douglas Photiadis, and Bernard Matis (Phys. Acoust., U.S. Naval Res. Lab., Washington, DC)

Ultrasonic crystals provide a novel path forward for studying Dirac-like physics in classical systems free from interaction effects and are considered promising candidates for the development of more complex multilayer systems, such as those governed by a Moire pattern. However, no ultrasonic crystal experiment (and to the best of our knowledge no experiment studying classical wave propagation in a hexagonal lattice) has measured linear dispersion along the Dirac cone at the edge of the Brillouin zone where the upper and lower bands touch, which must include a measure of reduced phase and group velocities along the cone. This presentation will discuss experimental and theoretical results on acoustic surface wave propagation in a hexagonal lattice of resonant cavities, which can aid in the observation of such Dirac-like physics for ultrasonic crystals.

8:45

1aPA3. Hierarchical coiled phononic crystals: Reflectionless interfaces by duality. Carson Willey (UES Inc./Air Force Res. Lab., 2179 12th St., Bldg. 654/Rm. 304, Wright-Patterson AFB, OH 45433, carson.willey.ctr@us.af.mil), Vincent Chen (UES Inc./Air Force Res. Lab., WPAFB, OH), and Abigail T. Juhl (Air Force Res. Lab., Wright Patterson Air Force Base, OH)

Duality has previously been demonstrated in twisted Kagome lattices, which are composed of a 2-D unit cell whose configuration can be transformed smoothly by varying a particular unit cell parameter. In the Kagome lattice, the twist angle defines the orientation of two triangular structures comprising the unit cell. In this case, duality expresses the fact that dispersion curves are identical for twist angles that are $+/-\ \delta\psi$ away from the self-dual configuration at $\psi_{;c}$. The term self-dual refers to a certain unit cell configuration where dispersion curve families become degenerate, and thus, overlap. In this talk, a 1-D phononic crystal (PnC), composed of rotation-locked nodes linked by flexible bars modeled by continuum elements, is studied as a function of a twist angle. Interestingly, we demonstrate that a 1-D bar-based PnC also exhibits duality, when subjected to periodic rotation locking at the nodes. We show that particular segments of dual unit cells, having a $+/-\ 90^{o}$ difference in their wave propagation axes may be connected together without reflection and that this enables hierarchically coiled PnC (HCPnC), is contrasted with topologically protected metamaterials that also enable reflectionless wave propagation around a corner.

9:05

1aPA4. Bound states in the continuum in clusters of acoustic and elastic resonators. Daniel Torrent (Universitat Jaume I, Av Vicente Sos Bayntat, Castellon de la Plana, Castellón 12071, Spain, dtorrent@uji.es), Marc Martí Sabaté (Imperial College, London, United Kingdom), Junfei Li (Elec. and Comput. Eng., Duke Univ., Durham, NC), Steven Cummer (Elec. and Comput. Eng., Duke Univ., Durham, NC), and Bahram Djafari-Rouhani (Univ. of Lille, Villeneuve d'Asq, France)

This study investigates the localization of waves within highly symmetric clusters of scatterers, focusing on the resonances that emerge when resonators are strategically positioned along the perimeter of a circumference. Notably, the quality factor of these resonances exhibits a noteworthy enhancement with an increasing number of resonators. We demonstrate that, specifically for flexural waves, as the number of scatterers approaches infinity, a defining condition emerges for the formation of bound states in the continuum (BICs). Additionally, we present an analytical expression for the design of these BICs. The findings extend beyond flexural waves, being applicable to various types of classical waves. To validate our theoretical framework, we conduct an experimental study in acoustics, showcasing the realization of an acoustic BIC open resonator. This resonator is constructed by coupling polygonally arranged holes within a two-dimensional waveguide. Our proposed design enables the scanning of the acoustic field, facilitating the retrieval of mode shapes and the estimation of quality factors through the analysis of signal decay rates.

9:25

1aPA5. Comparison between the performance of as-designed and as-built phononic pseudocrystal isolators. S. Hales Swift (Sandia National Labs., P.O. Box 5800, MS 1082, Albuquerque, NM 87123-1082, shswift@sandia.gov), Michael Denison, and Dale E. Cillessen (Sandia National Labs., Albuquerque, NM)

Many phononic crystal studies involve fabrication of a physical object and predicting its vibrational properties and then compare these to the measured properties; however, fewer studies examine whether the differences between the as-designed and as-built versions of the finished object are relevant to its measured performance. This study considers the effects of geometric differences between as-designed and as-built phononic pseudocrystal isolators implemented in steel using laser powder-bed fusion and compares the results of measurement with simulation of the as-designed and as-built versions of the isolator to show that these differences can be significant and typically tend to degrade performance. [*SNL is managed and operated by NTESS under DOE NNSA Contract No. DE-NA0003525.*]

9:45–10:00 Break

10:00

1aPA6. Backscattering-free edge states below all bands in two-dimensional auxetic media. Wenting Cheng (Phys., Univ. of Michigan, Ann Arbor, MI), Kai Qian (Univ. of California San Diego, San Diego, CA), Nan Cheng (Phys., Univ. of Michigan, Ann Arbor, MI), Nicholas Boechler (Univ. of California San Diego, San Diego, CA), Xiaoming Mao (Phys., Univ. of Michigan, 450 Church St., Ann Arbor, MI 48108, maox@umich.edu), and Kai Sun (Phys., Univ. of Michigan, Ann Arbor, MI)

Unidirectional and backscattering-free propagation of sound waves is of fundamental interest in physics and highly sought-after in engineering. Current strategies utilize topologically protected chiral edge modes in bandgaps or complex mechanisms involving active constituents or nonlinearity. Here, we propose a new class of passive, linear, one-way edge states based on spin-momentum locking of Rayleigh waves in two-dimensional media in the limit of vanishing bulk modulus, which provides 100% unidirectional and backscattering-free edge propagation immune to any edge roughness at a broad range of frequencies instead of residing in gaps between bulk bands. We further show that such modes are characterized by a new topological winding number that is analogous to discrete angular momentum eigenvalues in quantum mechanics. These passive and backscattering-free edge waves have the potential to enable a new class of phononic devices in the form of lattices or continua that work in previously inaccessible frequency ranges.

Contributed Papers

10:20

1aPA7. Time-varying phononic metamaterials. Raul Esquivel-Sirvent (Universidad Nacional Autonoma de Mexico, Instituto de Fisica, Circuito de la Investigacion S/N, Mexico, Coyoacan 01000, Mexico, sirventr@gmail.com)

The introduction of the concept of time crystals [1] opened the possibility of looking for classical systems whose properties change in time. These analog systems are known as time-varying materials [2]. In this talk, I will present two examples of time-varying systems. In one case, we consider a homogeneous slab with an acoustic impedance that varies with time in a periodic way with period T; this is, $Z(t) = Z(t + T)$. Although it is a homogeneous medium, a band structure appears due to the time dependence of the impedance. However, rather than having the bands rather than being in frequency are in the wave-number space. The second example is a phononic crystal made of a periodic array of unit cells of two different materials. In this case, one of the materials shows a time-dependent impedance that introduces a second modulation on the system. The possible realization of time-dependent structures and the potential applications will be discussed.
F. Wilkzek, Quantum Time Crystals, Phys. Rev. Lett. 109, 160401 (2012). E. Galiffi, R. Tirole, S. Yin, H. Li, S. Vezzoli *et al*. "Photonics of time-varying media," Adv. Photonics, 4, 014002 (2022). [Work partially supported by DGAPA-UNAM.]

10:35

1aPA8. An investigation of the effect of randomness in trunk size and location on the attenuation of sound through forests generated by Lindenmayer systems. Divyamaan Sahoo (Dept. Elec. & Comput. Eng., Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., North Dartmouth, MA 02747, divyamaansahoo@gmail.com)

This presentation provides an overview of what is known about the influence of randomness in trunk size and location on the attenuation of sound through forests. It also provides an explanation of the Lindenmayer system (L-system) approach to modeling plants and trees in addition to the multiset L-system approach to modeling the dynamics of forest ecosystems, investigating its applicability to the problem of optimizing sound attenuation through forests. This project was completed in partial fulfillment of the requirements for the Diploma in Acoustics and Noise Control, Institute of Acoustics, Milton Keynes, UK, under the mentorship of Dr. Keith Attenborough.

10:50–11:50
Panel Discussion

Session 1aPP

Psychological and Physiological Acoustics: Psychological and Physiological Acoustics Poster Session

Gregory M. Ellis, Chair
Audiol. and Speech Pathol., Walter Reed Natl. Med. Military Ctr., 4494 Palmer Rd. N, Bethesda, MD 20814

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

Contributed Papers

1aPP1. Perceptual weighting of acoustic cues to contrastive prosody for sentences in quiet and in noise. Harley J. Wheeler (Speech-Language-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, wheel488@umn.edu) and Matthew B. Winn (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN)

Prosody is used to mark important information in speech, yet despite this important role in everyday communication it is not a customary part of speech recognition testing in routine audiometry. Listeners might fail to perceive meaningful emphasis on a specific word signaled by prosodic cues despite repeating the words correctly. This study introduces a new paradigm for assessing perception of prosodic cues that are used to signify new/corrective information in a sentence. Stimuli consisted of spoken sentences where one word (in various sentence positions) was emphasized through acoustic modifications across the entire utterance, in a manner that corrected wrong information. Participants used a visual analog scale to mark the timing and degree of emphasis aligned with the target words. Perceptual data in this study are linked with acoustic measures of voice pitch contour, intensity, and duration to characterize how contrastive stress cues are recovered by listeners with and without cochlear implants, who are especially at risk for poorer pitch perception. Follow-up conditions used stimuli where acoustic cues (F0, duration, intensity) were manipulated independently to explore perceptual cue weighting, and the perseverance of these cues through background noise.

1aPP2. Multiple timescales of context shape the perceptual sensitivity to natural pairings of musical pitch and timbre. Christian E. Stilp (Psychol. and Brain Sci., Univ. of Louisville, 317 Life Sci. Bldg., Louisville, KY 40292, christian.stilp@louisville.edu), Isabel Adames, and Anya E. Shorey (Psychol. and Brain Sci., Univ. of Louisville, Louisville, KY)

Musical pitch and timbre (specifically, spectral shape) covary perceptually: lower pitches are associated with darker timbres (less higher-frequency energy) and higher pitches are associated with brighter timbres (more higher-frequency energy). We examined this relationship at multiple timescales of context: trial-level (rates of performance improvement within a testing block), block-level (differences in performance across blocks), session-level (performance as a function of testing block order), and longer-term experience (musical training / background). Musical pitches were labeled as low (C4) or high (G4) from various instruments (trumpet, oboe, trombone, tuba). Testing blocks paired pitch and timbre together to either respect (Consistent) or violate (Reversed) their covariance. All timescales of context contributed to performance, producing clear patterns across experiments. Pitch labeling was poorer in Reversed blocks, but showed steeper rates of improvement than did Consistent blocks (which were often near/at ceiling levels). These patterns were moderated by which condition was tested first in the experiment and by the introduction of trial-by-trial feedback. Greater musical training was consistently correlated with higher accuracy, but adding feedback extinguished this pattern for Consistent blocks. Thus, context on a host of timescales shapes perceptual sensitivity to the natural covariance between musical pitch and timbre.

1aPP3. The effects of preceding sound on psychoacoustic tuning curves measured in simultaneous and forward masking. Elizabeth A. Strickland (SLHS, Purdue Univ., 715 Clinic Dr., West Lafayette, IN 47907, estrick@purdue.edu) and Heesun Park (SLHS, Purdue Univ., West Lafayette, IN)

The medial olivocochlear reflex (MOCR) decreases the gain of the cochlear active process in response to sound. We have used psychoacoustic techniques to show behavioral effects of gain reduction, which could be consistent with the MOCR. We have used paradigms understood to measure frequency selectivity and the input/output function at the level of the cochlea using stimuli (masker and signal) that should be too short to evoke the MOCR. A precursor sound is then presented before these stimuli to evoke the MOCR. Our most recent studies have used forward masking, to avoid the complicating effects of suppression. The current study was designed to examine the effects of suppression, by comparing the effects of a precursor on frequency selectivity measured using forward masking and simultaneous masking. Psychoacoustic tuning curves were measured using simultaneous and forward masking, with and without a precursor. This allowed us to measure the change in tuning with and without the presence of suppression. Broadband precursors and tonal precursors at masker frequencies were measured because previous studies have shown broadening with broadband precursors and sharpening when tonal maskers are the precursors. Results will be discussed in the context of current understanding of the MOCR.

1aPP4. Influence of social and semantic context in processing speech in noise. Etienne Abassi (McGill Univ., 1491 Rue Gilford, Montreal, QC H2J1S1, Canada, etienne.abassi@gmail.com) and Robert Zatorre (McGill Univ., Montreal, QC, Canada)

Social interactions occupy a substantial part of our life, and listening to others' interactions is critical in understanding our social world. Although the role of semantics in speech comprehension has been studied, the role of social context, and its interaction with semantics, remain unknown. We conducted a series of four perceptual experiments to better understand the processing of multiple-speaker conversations from a third-person viewpoint, manipulating the social and semantic context of a conversation. We used a stimulus set consisting of two-speaker dialogues or one-speaker monologues (factor: social context) arranged in intact or sentence-scrambled order (factor: semantic context). Each stimuli comprised five sentences, with the fifth sentence embedded in multi-talker babbling noise. This fifth sentence was subsequently repeated without noise, with a single word altered or unchanged. Stimuli were presented to healthy young adults, asked whether the repeated sentence was same as or different from previous in-noise sentence. We found significant effects for both social and semantic contexts when processing a conversation. Our findings highlight that both semantic and social aspects of a conversation can modulate the processing of conversations. These results raise new questions regarding predictive or other mechanisms that may be at play when perceiving speech in social contexts.

1aPP5. Sensitivity to spectral and temporal cues in music with a cochlear implant in single-sided deaf listeners. John J. Galvin (House Inst. Foundation, 1127 Wilshire Blvd., Ste. 1620, Los Angeles, CA 90017, jgalvin@hifla.org), Sean Lang (House Inst. Foundation, Los Angeles, CA), Natalia Stupak, and David Landsberger (Dept. of Otolaryngol.—Head and Neck Surgery, NYU Grossman School of Medicine, New York, NY)

Despite the poor sound quality with the cochlear implant (CI), single-sided deaf (SSD) CI users often prefer to listen to music with binaurally combined acoustic and electric hearing. The source of this binaural benefit is unclear. Is the benefit due to gross binaural restoration or synchronous temporal envelope information across ears? Does the detail of the spectral and/or temporal content in the CI ear matter? We investigated sources of binaural benefit for music quality in SSD-CI users by manipulating the input to the CI. Unprocessed music was presented to the acoustic-hearing ear. Experimental stimuli were delivered directly to the CI ear: unprocessed (same as everyday listening), reduced temporal modulations (using a sinewave vocoder to restrict the input to the CI), reduced spectral information (wideband noise modulated by the music temporal envelope), or wideband noise. Preliminary results suggest sensitivity to the spectral and temporal information delivered to the CI ear; however, this sensitivity was highly variable across participants. Binaural benefits were not observed when only temporal cues or white noise were presented to the CI ear. Manipulation of vocoders presented to the CI ear may be a promising approach to evaluate perception of spectral and temporal cues with the CI.

1aPP6. Wave-based signal-to-noise enhancement and the shape of cochlear filters. Alessandro Altoe (Dept. of Otolaryngol., USC, 1640 Marengo St., Los Angeles, CA 90013, altoe.alessandro@gmail.com) and Christopher Shera (Univ. of Southern California, Altadena, CA)

Cochlear transfer functions measured at one location—a.k.a. "cochlear filters"—are characterized by a steep roll-off as the frequency increases above the local best (responding) frequency (BF). The functional role of the sharp "cut-off" is not well understood, although it was previously hypothesized that it plays a major role in encoding sound frequency. Less hypothetical in nature, our work elucidates the role of the steep cut-off in improving signal detection in cochlear-like amplification strategies. We study signal amplification in generic active gain media models and physics-based cochlear models as well. We demonstrate that the optimal strategy for boosting the signal-to-noise ratio (SNR) at a given location "x" requires high gain basal to "x," followed by rapid attenuation (i.e., sharp cut-off) beyond it. This strategy of amplification followed by a sharp cut-off is precisely how the cochlea processes traveling waves of given frequencies; it boils down to a peculiar form of spatial filtering where waves coming from the "signal side" are amplified, while waves coming from the direction where there is noise but no signal are squelched. This noise "squelching" action manifests in the cochlear filters as the steep high-frequency roll-off.

1aPP7. Characterizing inner-hair-cell specific dysfunction from spike-train-derived transduction functions using a phenomenological auditory-nerve model. Madhurima Patra (Biomedical Eng., Purdue Univ., 715 Clinic Dr., West Lafayette, IN 47907, mpatra@purdue.edu), Adarsh Mukesh, and Michael G. Heinz (Speech Lang. and Hearing Sci., Purdue Univ., West Lafayette, IN)

The impact of outer-hair-cell damage on sensorineural hearing loss (SNHL) has been extensively studied compared with inner-hair-cell (IHC) damage (e.g., stereocilial). Pre-clinical SNHL animal models provide unique data to directly address IHC-specific deficits (e.g., carboplatin-exposed chinchillas). Spike-train data (period histograms) to low sound level tones can be used to derive IHC-transduction functions by mapping instantaneous spike rates to corresponding sinusoidal pressure values (Horst *et al*., 2018). However, this approach depends on sensation level and spontaneous rate, which are both affected by IHC dysfunction. To better understand the effect of these dependencies, we used a phenomenological auditory-nerve (AN) model (Bruce *et al*., 2018) that provides parametric control over IHC and OHC dysfunction allowing exploration of optimal experimental design. Here, we explore the utility of spike-train-derived IHC-transduction functions in capturing parametric changes in IHC dysfunction as currently implemented in the AN model. We used unsupervised methods and information-theoretic approaches to quantify the dependency of transduction functions on IHC dysfunction. We also compared model findings with AN-fiber data recorded from carboplatin-exposed chinchillas. Preliminary findings suggest that transduction functions obtained from the AN model can capture IHC-specific dysfunction within specific regimes of controllable model parameters, thus showing promise for characterizing IHC dysfunction.

1aPP8. Hypercompression in a tapered, viscous, nonlinear cochlear model. Vipin Agarwal (Mech. Eng., Univ. of Memphis, Memphis, TN), Wen Cai (Mech. Eng., Univ. of Michigan, Ann Arbor, MI), and Karl Grosh (Mech. Eng., Univ. of Michigan, 2350 Hayward St., Ann Arbor, MI 48105, grosh@umich.edu)

In our current research, we focus on predicting the stationary nonlinear response of a cochlear model that extends from the base to the apex when subjected to harmonic input, taking into account the tapering of the cochlear scalae along with electromechanical coupling of outer hair cells and the microstructures of the organ of Corti. We are interested in explaining the paradoxical phenomenon of hypercompression in the motion of the reticular lamina (RL), whereby the response decreases with an increase in the acoustic excitation at the stapes. We derive frequency–response curves for both the basilar membrane and RL at various locations, exploring a wide range of excitation frequencies and amplitudes. In accordance with experimental data, we find that the RL exhibits hypercompression at the base of the cochlea. This behavior is found to arise from the interaction of saturating nonlinearity arising from the active process and the linear response identified as the passive response to acoustic stimulus. These two responses are not synchronized in phase. As the excitation level increases, the two effects tend to partially cancel, giving rise to lower responses with increasing sound pressure levels. [Work supported by NIH-NIDCD R01 04084.]

1aPP9. Adjusting listening effort relative to expected value of listening. Matthew B. Winn (Speech-Language-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr. SE, Shevlin Hall Rm. 115, Minneapolis, MN 55455, mwinn@umn.edu)

Listening effort is a commonly reported difficulty among those who have hearing loss. A person's ability to engage or disengage mental resources at strategic times could be an important signature of a person's capacity to guard against wasted effort. This experiment gave listeners an opportunity to voluntarily reduce effort at specific moments based on the expected value of the incoming signal. First, sentences presented in the clear were immediately repeated, inviting reduced attention to the first presentation. In a second condition, the immediate repetition was randomly dropped on some trials, inviting increased vigilance to the first presentation. In a third condition, the first sentence was presented with degraded quality, inviting increased attention to the repetition to disambiguate misperceived words. Pupillometry was used as an index of moment-to-moment changes in listening effort. Data showed elevated pupil dilations linked in time with the disambiguated words and diminished pupil size specifically when there was less value in attending to the signal. These results support the need to expand the concept of listening effort beyond a "more" or "less" framework, toward a framework of efficiency.

1aPP10. Assessing the role of auditory access in infancy. Kristin M. Uhler (PM&R, Univ. of Colorado Anschutz Medical Campus, 12631 East 17th Ave., Academic Office 1, Ste. 1201, Aurora, CO 80045, kristin.uhler@cuanschutz.edu), Daniel Tollin (Dept. of Physiol. & Biophys., CU Anschutz, Aurora, CO), Angela M. Madrid (Phys. Med. and Rehab., Univ. of Colorado Anschutz Medical Campus, Aurora, CO), and Phillip Gilley (PM&R, Univ. of Colorado Anschutz Medical Campus, Boulder, CO)

Even with the widespread adoption of newborn hearing screening and Early Hearing Detection and Intervention systems, infants who are hard-of-hearing (IHH) remain at increased risk for poor or delayed development of auditory and speech perception skills. Speech perception is positively correlated with better auditory development on functional auditory skills and a variety of language outcomes. However, clinical utilization of speech perception tests has not been broadly accepted. One clinically viable tool to

assess speech perception during infancy is our electroencephalography (EEG) procedure, which has been validated as an objective measure of speech discrimination. However, we have not examined the development of infant speech perception among IHH and infants with normal hearing (INH). Perceptual attunement—or "narrowing"—is a model of perceptual learning that posits that perceptual abilities are shaped by environmental experiences over the first year of life. Here, we describe our adapted EEG methods to examine the development of infant speech perception to study periods of perceptual attunement between IHH and INH. Preliminary findings confirm that at 3 months of age, infants demonstrate neural encoding of native and non-native speech sounds, and at 6 months of age results suggest a decrease in non-native speech sound encoding.

1aPP11. Measuring and modeling Gaussian noise disruption across frequency, level, and hearing status. Adam Svec (Audiol., San Jose State Univ., 448 N 2nd St., San Jose, CA 95112, adamsvec@gmail.com), Marc Brennan (Special Educ. Commun. Disord., Univ. of Nebraska-Lincoln, Lincoln, NE), Afagh Farhadi (Speech, Lang., and Hearing Sci., Purdue Univ., West Lafayette, IN), Braden Maxwell (Psych., Univ. of Minnesota - Twin Cities, Rochester, NY), Sarah Garvey (Special Educ. and Commun. Disord., Univ. of Nebraska-Lincoln, Lincoln, NE), Matthew Pascua (Audiol., San Jose State Univ., San Jose, CA), and Laurel H. Carney (Biomed. Eng., Univ. of Rochester, Rochester, NY)

Recent work suggested that older listeners with minimal hearing loss exhibit more forward masking for a highly fluctuating Gaussian noise (GN) masker relative to a low-fluctuation noise (LFN) masker. This threshold difference (GN–LFN) is referred to as GN disruption and was previously attributed to time-varying cochlear gain resulting from feedback control by the medial olivocochlear (MOC) system. However, thus far, masker conditions have been limited to 80 dB SPL at 4 kHz. The current study aims to challenge the hypothesis that the physiological basis of GN disruption is the MOC system using behavioral measures and computational models across multiple target and masker frequencies (0.5, 1, 2, and 4 kHz), masker levels (80-, 65-, and 50-dB SPL), masker-signal delays (25- and 75-ms), participant age ranges (18–30 and 60–75), and hearing statuses (normal hearing, sensorineural hearing loss). Additionally, speech recognition using IEEE sentences will be measured with GN and LFN maskers. Although testing will not be completed until 2025, preliminary data from 20 younger participants with normal hearing suggest that GN disruption is greater for higher than lower masker and probe frequencies, that GN disruption is similar across masker levels, and that refinements to the computational model are needed to more accurately predict GN disruption.

1aPP12. Factors influencing the minimum audible change in head orientation of a talker. Brian B. Monson (Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, 901 S Sixth St., Champaign, IL 61820, monson@illinois.edu)

Humans can detect changes in the head orientation (a.k.a. facing angle) of a talker using only auditory cues. Head orientation cues are beneficial for determining whether one is the intended recipient of an utterance and for segregating a target talker from background talkers. Because human talkers are directional sound sources, binaural cues and monoaural spectral cues can both contribute to talker head orientation perception. In this study, we assessed listeners' sensitivity to talker head orientation changes using only monaural cues. We tested several factors that could influence the minimum audible change in head orientation: talker and gender (two male, two female), stimulus bandwidth (full-band versus low-pass filtered at 8 or 10 kHz), transducer (loudspeaker versus headphone), stimulus uncertainty (interleaved versus blocked presentation of four talkers), and vocal production mode (speech versus singing). Bandlimiting at 8 or 10 kHz led to worse performance, as did increasing stimulus uncertainty. The effect of transducer was very limited. Performance with speech was better than that with singing. The effect of talker was large and consistent, suggesting individual variability in speech directivity patterns may affect head orientation cues. [Work supported by NIH grant R01DC019745.]

1aPP13. The impact of hearing-aid amplification and its integrated tinnitus feature on tinnitus management. Ieda Ishida (Innovation Ctr. Toronto, Sonova Canada, 3105, Unit 2, Mississauga, ON L5L1J3, Canada, ieda.ishida@sonova.com), Wei Sun (Dept. of Commun. Disord. and Sci. - Ctr. for Hearing and Deafness, State Univ. of New York at Buffalo, Buffalo, NY), Eric Cui (Dept. of Psych., Univ. of Toronto, Toronto, ON, Canada), Maren Stropahl, Matthias Keller (Dept. of Res. and Development, Sonova US, Staefa, Switzerland), Elizabeth Rivera Rosario, Julia Conenna, Celine Wan, Ivy Dimaculangan (Dept. of Commun. Disord. and Sci., Ctr. for Hearing and Deafness, State Univ. of New York at Buffalo, Buffalo, NY), Stefanie Hensler, Teresa Wenhart (Dept. of Res. and Development, Sonova AG, Staefa, Switzerland), and Jinyu Qian (Innovation Ctr. Toronto, Sonova Canada, Mississauga, ON, Canada)

Tinnitus is the potentially debilitating perception of phantom sound with no current cure. Although some management approaches are available, the benefits of hearing aid amplification and its added noiser are uncertain. This study assessed impacts of hearing-aid amplification, and amplification with an added noiser feature, on adults with hearing loss and chronic bothersome tinnitus. Thirty adults [42–75 (Mean = 61.1) years old; 18 males], with mild to moderate sensorineural hearing loss but no previous amplification exposure; and bothersome tinnitus [Tinnitus Functional Index (TFI) baseline scores of at least 20], were randomly assigned to one of two groups in a cross-over study to experience amplification-only, and amplification + noiser for one month each, following one month of no intervention. Study hearing aids were worn minimally for 5-h/day. TFI questionnaires evaluated tinnitus severity before and after each condition. A one-way repeated measures ANOVA with 30 participants revealed a significant effect on TFI across baseline, amplification-only, and amplification + noiser conditions ($p = .023$), suggesting tinnitus improvement across conditions. Further post-hoc tests indicated lower TFI scores in the amplification + noiser condition ($p = 0.048$), identifying its distinct benefit. Preliminary analysis suggests benefit of hearing aid amplification, with a prominent advantage of the noiser feature on tinnitus management.

1aPP14. Exploring the hidden risks: How U.S. infrastructure contributes to hearing loss. Tianle Duan, Qingchun Li (Purdue Univ., West Lafayette, IN), and Noori Kim (Purdue Univ., 401 N. Grant St., Rm. 115, West Lafayette, IN 47907, kim4147@purdue.edu)

Hearing loss is a critical health issue in the US, as approximately 15% of American adults live with hearing loss. Existing factors affecting hearing loss include congenital factors, chronic middle ear infections, noise exposure, age, gender, lifestyle, and ototoxic drugs. However, few studies investigated how infrastructure would affect hearing loss. Forbes Health recently reported cities with the greatest hearing loss risks ranked by the densities of specific infrastructure establishments, such as restaurants and cafeterias, bars and nightclubs, construction, and casinos, where sound can reach damaging levels (i.e., producing noise over 85 decibels on average). To verify this idea, we investigated the correlation between the hearing loss rate and the density of risky establishments at the county level, using the U.S. Census Bureau hearing loss population data and point-of-interest data provided by Safegraph and Openstreet. However, the results show that 81.25% of states (39/48) have statistically significant negative correlations between the densities of risky establishments and hearing loss rates at county levels. In contrast, the other states show no significant correlations. The results imply a different mechanism from Forbes' report of infrastructure damaging hearing and suggest further exploration of how infrastructure contributes to hearing loss in the US.

1aPP15. Digital therapeutics in hearing healthcare. Keisuke A. Nakamura (Purdue Univ., West Lafayette, IN) and Noori Kim (Purdue Univ., 401 N. Grant St., Rm. 115, West Lafayette, IN 47907, kim4147@purdue.edu)

Digital therapeutics (DTx) in hearing research have emerged as a new category of therapies that provide evidence-based intervention via digital means, such as software, smartphone apps, or websites. However, as they

are relatively new, yet not well-established. In this presentation, we review DTx technologies in hearing research fields, focusing on three categories: prevention and diagnosis, aid (assistance), and curing (digital medicine). We observed that the majority of DTx systems require interactions with users (or patients) without clinical professionals' direct support to obtain or collect medical evidence; this makes training (or education) features crucial to the success of the therapy. In this view, we will discuss the education or training functions of the current DTx and their contribution and purposes. The impact of emerging artificial intelligence (AI) on DTx in hearing research is being explored, along with discussions about the future of DTx concerning AI integration. We believe that this work will contribute to a better understanding of the current and future DTx technological advancements, shedding light on the field of hearing research, in particular.

1aPP16. Cognitive and developmental contributions to psychometric functions on three auditory tasks during adolescence and young adulthood. Julia J. Huyck (Speech Pathol. and Audiol., Kent State Univ., 1325 Theatre Dr., CPA A144, Kent, OH 44242, jhuyck@kent.edu), Serena A. Sereki, and Jordin T. Benedict (Speech Pathol. and Audiol., Kent State Univ., Kent, OH)

Auditory perceptual tasks mature at different rates, with temporal tasks having longer developmental courses. We tested 10- to 23-year-olds on frequency discrimination, temporal-interval discrimination, and gap detection using the method of constant stimuli. During task performance, listening effort data were collected via eye-tracker. A cognitive battery was also administered. Thresholds correlated between the frequency and temporal-interval discrimination conditions, which shared the same standard stimulus, but not between other conditions. The slopes of the psychometric functions did not correlate between conditions, suggesting that different or time-varying factors might contribute to slopes on these conditions. Shallower psychometric functions corresponded to poorer thresholds on only the temporal-interval and gap conditions, implying a potential role for attention, engagement, fatigue, or the stability of the stimulus representations on those conditions. Regression modeling indicated a limited role of age: Age predicted thresholds on the temporal-interval discrimination condition and slopes on both the frequency and temporal-interval discrimination conditions. Testing order affected performance on frequency discrimination (threshold) and temporal-interval discrimination (threshold and slope). The measures of listening effort and cognition that predicted thresholds and slopes also differed between those two conditions. On gap detection, neither threshold nor slope was predicted by any variable. [Work funded by NIDCD.]

1aPP17. Individual differences in talker discrimination among adult cochlear implant users. Terrin N. Tamati (Dept. of Otolaryngol., Vanderbilt Univ. Medical Ctr., 1608 Aschinger Blvd, Columbus, OH 43212, terrintamati@gmail.com) and Victoria A. Sevich (Speech and Hearing, The Ohio State Univ., Columbus, OH)

Real-world speech communication involves interacting with talkers with diverse voices and accents. Adult cochlear implant (CI) users often demonstrate poor discrimination of talkers' voices relative to their normal-hearing peers, which may contribute to difficulties in understanding speech produced by multiple talkers. However, the factors underlying poor talker discrimination ability are not well understood. The current study examined auditory and cognitive-linguistic factors that support talker discrimination, and explored the association between talker discrimination and sentence recognition across CI users. In an AX talker discrimination task, 25 adult CI users indicated whether word pairs were produced by the same talker or by two different talkers. Participants also completed measures of spectrotemporal processing, cognitive-linguistic skills, including fluid intelligence, working memory capacity, inhibitory control, and phonological processing, and sentence recognition. Results showed that talker discrimination scores (accuracy, sensitivity) were moderately to strongly correlated with spectrotemporal processing, fluid intelligence, inhibitory control, and phonological processing. Talker discrimination scores were also related to sentence recognition accuracy scores across CI users. These preliminary findings suggest that both auditory and cognitive-linguistic processes support talker discrimination, and, furthermore, may underlie the relationship between talker discrimination and sentence recognition in adult CI users.

1aPP18. Patterns of phonetic cue switching in older adults. Mishaela DiNino (Commun. Disord. and Sci., Univ. at Buffalo, 5000 Forbes Ave., Pittsburgh, PA 15213, mdinino4@gmail.com)

Primary phonetic cues for identifying speech in quiet may be unreliable in background noise, requiring listeners to utilize cues more resistant to masking to accurately recognize speech. Previous experiments demonstrated that individuals rely mainly on voice onset time (VOT) to categorize /bir/-/pir/ sounds in quiet but rely on fundamental frequency (F0) when VOT is masked by noise. As older adults report challenges perceiving speech in noise, the current study investigated whether older adults exhibited poor ability to switch reliance from VOT to F0 in broadband noise. Young (aged 18–30) and older (aged 55+) adults with normal or near-normal hearing categorized /bir/ and /pir/ sounds that were manipulated on a continuum of VOT and F0. No group differences were found in cue weights in quiet or when noise was presented at a signal-to-noise ratio (SNR) of 0 dB. However, when SNR ranged from + 6 to -4 dB, young adults relied on F0 even at the best SNR, whereas older adults continued relying on VOT until significantly poorer SNRs, suggesting that older adults may exhibit maladaptive strategies for phonetic cue switching. Continued use of cues even as they become less reliable may help explain why older adults experience difficulty identifying speech in noise.

1aPP19. Eye track, you listen: Listening effort in cochlear implant users. Anthony Q. Robinson (7545 Valley Mist Dr., Memphis, TN 38133, anthonyqrj@gmail.com), Victoria A. Sevich (Columbus, OH), and Terrin N. Tamati (Nashville, TN)

Cochlear Implant (CI) users can experience mental fatigue from the listening effort needed to process the degraded speech information delivered by the implant. Listening effort is unique to the individual CI user and may provide insight into the strengths and weaknesses of real-world speech communication. To analyze listening effort, an eye tracker lab was built utilizing pupillometry that evaluates pupil dilation to detect cognitive load. My experience working in the lab and potential applications for pupillometry in CI research will be presented. Research will have implications for better understanding factors affecting speech recognition in CI users, including predictive speech, familiarity, and talker context, and top-down strategies.

1aPP20. Abstract withdrawn.

1aPP21. Does hearing impairment affect gait under realistic conditions? Troy Wesson (Otolaryngol.—Head and Neck Surgery, Indiana Univ. School of Medicine, Indianapolis, IN), Jeffrey Haddad, Satyajit Ambike (Health & Kinesiology, Purdue Univ., West Lafayette, IN), Abigail Foley (Speech, Lang. & Hearing Sci., Purdue Univ., West Lafayette, IN), Radha Patel, Sarah Burgin (Otolaryngol.—Head and Neck Surgery, Indiana Univ. School of Medicine, Indianapolis, IN), and Alexander L. Francis (Speech, Lang. & Hearing Sci., Purdue Univ., Lyles-Porter Hall, 715 Clinic Dr., West Lafayette, IN 47907, francisa@purdue.edu)

Hearing loss is associated with increased risk of falling, but little is known about how hearing impairment might affect mobility. Here, we report preliminary results from an ongoing experimental study investigating dynamic properties of gait while walking with and without impaired hearing. Young- and middle-aged adult participants with self-reported normal hearing completed four walking tasks, 2 indoors, 2 outdoors, each conducted with and without a simulated hearing loss. In the impaired condition, participants wore an insert earplug in one ear combined with binaural, circumaural noise-damping headphones. We quantified gait parameters using data from inertial measurement units (IMUs) affixed to participants' ankles and waist, allowing participants to walk farther than in typical gait assessments. Parameters included standard gait metrics, such as limb acceleration as well as a novel spatiotemporal index quantifying variability in step patterns. Data have been collected from 13 of 30 projected participants. Analysis will focus on how temporal-spatial gait parameters and variability differ between indoor and outdoor walking with and without hearing impairment. We expect gait parameters to reflect a more variable and potentially less mobile walking strategy in the outdoor condition and with impaired hearing due to increased cognitive load and/or reduced spatial awareness in those conditions.

1aPP22. Functional changes in brain organization after *de novo* audiomotor learning: An fMRI investigation. Floris van Vugt (Univ. of Montreal, Montreal, QC, Canada), Matthew Masapollo (McGill Univ., 630 William St., Montreal, QC H3C 4C9, Canada, matthew.masapollo@mcgill.ca), and David J. Ostry (McGill Univ., Montreal, QC, Canada)

To learn to talk or play a musical instrument, the brain must acquire a "mapping" of the relationship between movements and their acoustical consequences. However, it remains unclear how this type of audiomotor learning alters the functional organization of the brain, and what neural networks support consolidation of that learning. This study used functional magnetic resonance imaging (fMRI) to measure functional changes in brain organization as a result of audiomotor learning from scratch. We used a novel paradigm in which subjects learned to move a joystick in different directions in a 2D workspace to achieve speech-like acoustical targets. At the end of each movement, subjects received auditory feedback corresponding to the sound associated with the direction that they moved in. Before and after training, we used resting-state fMRI to assess learning-related changes in functional connectivity. Behavioral results show that, over the course of practice with feedback, subjects gradually produced joystick movements with fewer errors, indicative of learning. Furthermore, some of this learning was maintained to the second day, indicating subjects started forming durable performance gains. Analyses of the fMRI data are currently underway and aimed at localizing changes in neural activity that scale with behavioral indices of audiomotor learning.

1aPP23. Multimodal sensorimotor investigation of audio-visual integration in cochlear implant users. Olivier Valentin (Res. Inst. of the McGill Univ. Health Ctr., 2155 rue Guy, Montréal, QC H3H 2R9, Canada, m.olivier.valentin@gmail.com), Nicholas Foster (Université de Montréal, Montreal, QC, Canada), Bastien Intartaglia (Res. Inst. of the McGill Univ. Health Ctr., Montreal, QC, Canada), Marie-Anne Prud'homme (Université de Montréal, Montreal, QC, Canada), Marc Schönwiesner (Leipzig Univ., Leipzig, Germany), Sylvie Nozaradan (Université catholique de Louvain, Louvain-la-Neuve, Belgium), and Alexandre Lehmann (Res. Inst. of the McGill Univ. Health Ctr., Montreal, QC, Canada)

Rhythm is an omnipresent element of many daily activities. Numerous studies in cognitive sciences have highlighted that humans exhibit greater precision in synchronizing their movements with auditory rhythmic stimuli compared to visual ones. Deaf individuals were shown to excel in synchronizing with visual cues, surpassing those with normal hearing (NH). Furthermore, it was demonstrated that cochlear implant (CI) users were able to move in time to the beat of music, although not as well as NH controls. This study aims to investigate whether CI users retain a visual synchronization advantage from their pre-implant deafness, while maintaining auditory synchronization skills comparable to those of NH individuals, or if the neural reorganization post-implantation negates the visual synchronization advantage acquired pre-implantation. Specifically, we assessed both unimodal and multimodal auditory and visual abilities in CI users compared to NH controls using a standard sensorimotor synchronization paradigm. Results revealed that CI users exhibit comparable auditory rhythmic synchronization abilities to NH individuals, which is consistent with existing research, while not displaying a superior ability in synchronizing with visual rhythms, likely due to neural reorganization following implantation. This shift in audio-visual integration among CI users suggests that the post-implant reorganization of their auditory cortex might hinder the effective integration of temporal auditory input from the implant with visual information.

1aPP24. Visual speech cues relieve listening effort selectively for listeners with hearing loss. Justin T. Fleming (Speech-Language-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, jtf@umn.edu) and Matthew B. Winn (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN)

Elevated listening effort is one of the most commonly reported and widely impactful problems among individuals with hearing loss. The ability to see a talker's face may help mitigate listening effort, but research on this topic has generally excluded individuals with hearing loss. In this study, we used pupillometry as a time-sensitive index of listening effort for audio-visual speech while manipulating the availability of visual cues by selectively blurring the talker's mouth. Blurring removed place-of-articulation information while allowing fair comparison of pupil size across conditions. Effortful listening was elicited using a sentence repetition task in which participants needed to use later context to fill in a missing word from earlier in the sentence. For listeners with typical hearing, visual cues had no measurable effect on listening effort. In contrast, listeners with cochlear implants (CIs) had two distinct benefits from visual cues: first, pupil dilations were smaller, suggesting release from effort. Second, the downstream intelligibility errors caused by the word-repair process were alleviated. These results caution against generalizing findings obtained with typical-hearing participants to a hearing loss population, support deeper analysis of perception error patterns, and demonstrate the importance of multisensory speech cues to relieve listening effort.

1aPP25. Visual cues influence prosody perception among individuals with cochlear implants. Justin T. Fleming (Speech-Lanauage-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, jtf@umn.edu), Harley J. Wheeler, and Matthew B. Winn (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN)

Understanding a talker's intention conveyed through prosody is essential for successful speech communication. Although visual speech cues are typically characterized by lip movements, there are also visual gestures including head tilts and eyebrow raises that signal prosody. Listeners with cochlear implants (CIs) may rely on these visual cues to solidify their prosody perception and guard against misinterpretations, particularly because of their poor perception of voice pitch. This study used audio-visual recordings of sentences in which a talker sometimes focused one word to signify a change in meaning. After characterizing acoustic and visual prosody cues in this stimulus set, we made prosodically mismatched variants of the stimuli by transplanting prosodically focused audio onto broad focus (i.e., prosodically unfocused) video, and vice versa. Visual influences on prosody perception were measured using acoustic analysis of participant reproductions in a vocal mimicry task. Results show reproduction of F0 contour—a key acoustic marker of prosodic focus—is influenced by perception of visual prosody cues. Specifically, visual cues indicating a focused word led to higher peak F0 in vocal reproductions of that word in the sentence. These results underscore the importance of multisensory information in supporting prosody perception for listeners with hearing loss.

1aPP26. Feel sounds with your hands: Exploring tactile frequency-following response. Andreanne Sharp (Université Laval, 1050 rue de la Medecine, QC G1V 0A6, Canada, andreanne.sharp@fmed.ulaval.ca), Ana Belen Carbajal Chavez (Montreal neurological Inst., QC, Canada), Loonan Chauvette (Université Laval, QC, Canada), Robert Zatorre (Montreal Neurological Inst., Montreal, QC, Canada), and Emily Coffey (Concordia Univ., Montreal, QC, Canada)

Auditory perception is often influence by other senses. Prior studies have documented that the auditory cortex can respond to vibration, but the nature of this neural response remains unclear. The frequency-following response (FFR) is a non-invasive evoked brain response that can be used to study the fidelity of periodicity encoding of complex sounds. The main goal of this study was to investigate if tactile processing of sounds retains periodicity information as measured by the FFR. We acquired electroencephalography while participants were presented with repetitions of a synthesized speech syllable /da/ under three conditions (Auditory, Tactile, and both). A technology developed within our laboratory (Multichannel Vibrotactile Glove) was used to present sounds to all fingers. Results reveal that it is possible to measure a tactile FFR using vibrotactile stimulation, which is similar to the auditory FFR, but exhibits somewhat different characteristics as compared to unimodal auditory FFR, including lower amplitude and no sensitivity to harmonics. These findings suggest that these modalities interact, opening up questions about the origins and pathways responsible for the phenomena and introduces potential uses of tactile perception to mitigate the effect of hearing loss in speech and music perception.

1aPP27. The multichannel vibrotactile gloves: A transmodal technology to feel sound through touch. Loonan Chauvette, Eliane Leprohon, Louis-Philippe Perron-Houle (Université Laval, QC, Canada), Valentin Pintat, Aidin Delnavaz, Jeremie Voix (École de technologie supérieure, Université du Québec, Montréal, QC, Canada), and Andreanne Sharp (Université Laval, 1050 rue de la Medecine, QC, QC G1V 0A6, Canada, andreanne.sharp@fmed.ulaval.ca)

Development of devices for transmitting sounds through touch is motivated by needs coming from diverse disciplines. Hard-of-hearing individuals could benefit from vibrations to overcome the limitations of existing hearing technologies. Adding tactile cues can be useful for all in contexts where the acoustic information is limited due to sounds coming from multiple sources or noise. The potential sensory augmentation provided by the technology is also interesting in an entertainment context in order to offer immersive experiences. A transdisciplinary approach based on a framework recently developed in our laboratory was used to design this technology that enables the transmission of acoustic signals through touch. Validation experiments were carried out via electro-acoustic measurements as well as behavioral measurements in human subjects (n = 5). Electro-acoustic and behavioral measures support that the system provides uniform stimulation across hands and actuators. The frequency response curve as well as the summation effect measured via behavioral threshold measurements support that the tactile receptors are accurately stimulated by the devices. The multichannel vibrotactile gloves offer the flexibility to transmit diverse acoustic features to individual actuators, making them a valuable tool for research and a prospective technology capable of substituting, compensating, or extending sensory perception.

1aPP28. Creating a descriptor model to determine the psychoacoustic annoyance (PA) level of users exposed to hospital noise. Semiha Yilmazer (Faculty of Art, Design and Architecture, Dept. of Interior Architecture and Environ. Design, Bilkent Univ., Ankara 06800, Turkey, semiha@bilkent.edu.tr), Arzu Gönenç Sorguç (Architecture, METU, Ankara, Turkey), Cengiz Yilmazer (CSY R&D and Architecture Eng., Ankara, Turkey), Zekiye Şahin (Interior Architecture and Environ. Design, Bilkent Univ., Ankara, Turkey), and Aslı Zeynep Doğan (Architecture, Ankara Yıldırım Beyazıt Univ., Ankara, Turkey)

This study aims to develop a metric model to determine the psychoacoustic annoyance (PA) level of users exposed to hospital noise with a tonal component. Determination will reveal how a solution can be produced according to the spectral characteristics of the sound in the hospital environment. The study was conducted in the oncology polyclinic of Ankara City Hospital, Bilkent, Türkiye. Voluntary oncology outpatients in three locations in the polyclinic were given questionnaires and interviewed. Equivalent Continuous A-weighted sound Level (LAeq) measurements were taken within the interview hours and at 1-h intervals from three locations: the reception area, courtyard area, and corridor. Thirty normal-hearing participants joined in different experimental listening tests in the lab conditions related to the signal-based PA. The ambient broadband noise level varied with NC-20 and NC-40. The signals with different tone levels and tone frequencies were prepared. The PA model was developed to determine the average of the responses to each sound used in the listening test. The metric model was based on Zwicker's "Psychoacoustic Annoyance Model," which uses sound quality metrics. A correlation coefficient was computed to assess the relationship between the estimated psychoacoustic annoyance rating modeled by the created metric model and measured annoyance.

1aPP29. Abstract withdrawn.

1aPP30. Fluctuation and correlation analyses for spontaneous otoacoustic activity from three terrestrial vertebrate groups. Christopher Bergevin (Phys. & Astronomy, York Univ., Petrie 240, 4700 Keele St., Toronto, ON M3J 1P3, Canada, cberge@yorku.ca) and Olha Fedoryk (Phys. & Astronomy, York Univ., Toronto, ON, Canada)

Spontaneous otoacoustic emission (SOAE) is a telltale sign of an active ear that commonly manifest as a set of spectral peaks unique to a given individual. These sounds arise from a variety of species, despite dramatic morphological differences considered important for cochlear function. Furthermore, SOAE peaks exhibit amplitude (AM) and frequency modulations (FM), these fluctuations giving rise to their characteristic widths. There is, however, little consensus on how SOAE activity is generated. Here, we provide a systematic comparative study of SOAE peak fluctuations across three different groups: human, owl, and (anole) lizard. Specifically, we focus on analysis of correlative behavior in AM and FM fluctuations within a given peak (intra-peak, IrP) and across peaks (inter-peak, IPP) for SOAE waveforms measured from individual ears. In general, there were numerous similarities and differences across the three groups. To complement the empirical data, we also considered one model class (coupled limit cycle oscillators grouping into "frequency clusters") to ascertain how well that model captures fluctuations and associated correlative relations. Initial results indicate the model shows some consistencies with data (e.g., IPP correlations strongest for nearest-neighboring peaks) but also inconsistencies (e.g., model commonly exhibits IPP-FM correlations and to a greater degree than AM).

1aPP31. Exploring perceptual segregation cues in polyphonic vocal music under normal and impaired hearing. Lisanne G. Bogaard (Psych., Univ. of Minnesota, 75 E River Pkwy, Minneapolis, MN 55455, bogaa002@umn.edu) and Andrew J. Oxenham (Psych., Univ. of Minnesota, Minneapolis, MN)

Music plays an important role in the lives of many people, but its enjoyment can be compromised by hearing loss. Although the effects of hearing loss on speech are well-studied, its effects on music perception are less well-understood. This study examined normal-hearing and hearing-impaired listeners' ability to hear out individual voices within polyphonic music, while varying the number of voices and the level of inharmonicity in each voice. We hypothesized that hearing loss would hinder listeners' ability to distinguish voices in music due to impaired frequency selectivity and pitch perception but that inharmonicity would impact hearing-impaired listeners less, due to their reduced sensitivity to pitch discrepancies. MIDI-generated female vocal passages with one to five voices and different inharmonicity levels were presented in two tasks: estimating voice count (denumerability) and following a specific voice in the passage. Preliminary results indicate that hearing-impaired listeners struggle more with increasing voices than normal-hearing listeners but are less affected by inharmonicity. These outcomes align with our hypotheses, confirming that hearing loss leads to poorer overall performance and less sensitivity to inharmonicity. [Work supported by NIH grant R01DC005216.]

1aPP32. Do individual differences correlate across speech perception tasks? Weiyi Zhai (Linguist., McGill Univ., 1085 Dr. Penfield Ave. Montreal, QC H3A1A7, Canada, weiyi.zhai@mail.mcgill.ca) and Meghan Clayards (Linguist., McGill Univ., Montreal, QC, Canada)

Speech perception requires listeners to take into account acoustic cues as well as lexical context and phonetic (coarticulatory) context. Individuals have been shown to vary in how they integrate these factors. To better understand the sources of these differences, we conducted three phoneme categorization tasks on speech continua with 82 native Canadian English speakers. Task 1 (lexical + coartic) embedded a /s-ʃ/ continuum in lexically biasing contexts (e.g., a(s)ume, a(ʃ)ure) followed by different coarticulatory contexts (rounded or unrounded vowels). Task 2 (lexical) had only lexical context cues for /ɛ/-/ɪ/ vowel continua (e.g., v(ɛ)st, k(ɪ)t). In task 3 (coartic), a /d/-/g/ stop continuum in nonsense syllables followed different coarticulatory contexts (/ar/ or /al/). We found those who used lexical context more used coarticulatory context less in task 1, consistent with prior research. However, this correlation disappears when examined across tasks 2 and 3. We also found no correlation between individual use of lexical and coarticulatory context across tasks, suggesting task dependency. Participants' use of acoustic continua was positively correlated across tasks, indicating an individual trait for utilizing acoustic cues.

1aPP33. Novel auditory alerts that foster efficient detection and discrimination in complex auditory environments: Dual-task conditions. Mabel L. Cummins (Dept. of Anesthesiology, Vanderbilt Univ. Medical Ctr., 1211 21st Ave. South, Ste. 422 - Anesthesiology, Nashville, TN 37232, mabel.cummins@vanderbilt.edu), Michael Schutz (School of the Arts, McMaster Univ., Hamilton, ON, Canada), Leslie R. Bernstein (Neurosci. and Surgery, Univ. of Connecticut Health Ctr., Farmington, CT), Joshua Shive (Dept. of Psychol. Sci. and Counseling, Tennessee State Univ., Nashville, TN), and Joseph J. Schlesinger (Dept. of Anesthesiology, Vanderbilt Univ. Medical Ctr., Nashville, TN)

In many complex auditory environments, auditory alerts must be perceived and distinguished accurately while distractions are mitigated. Previously, we measured detection and discrimination performance using four novel alerts: one pair of narrowband and one pair of broadband alerts. Within each pair, one alert was perceived as consonant (labeled "friendly") and one as dissonant (labeled "enemy"). The alerts were presented along with maskers consisting of "truck noise" or "truck noise" combined with "speech babble." Separately, for the pairs of alerts, detectability of the alerts and discriminability between "friendly" and "enemy" pairs of alerts were measured as a function of signal-to-noise ratio (S/N). Results indicated that the alerts allow for robust detection and discrimination, even though the "friendly"-"enemy" pairs of alerts occupied similar spectral loci. In the study reported here, we measured the detectability and discriminability of the alerts under single and dual-task "N-back conditions" to simulate more attentionally demanding environments (e.g., in military conflict). The goal was to assess the impact of the N-back task on the robustness of the alerts to convey crucial information. Results will be presented and discussed in terms of the influence of the relevant variables on the form of the dual-task ROC. [Work funded by ONR N000142212184.]

1aPP34. Predictive factors for broad binaural pitch fusion in adults with hearing aids and cochlear implants. Lina A. Reiss (Oregon Health and Sci. Univ., 3181 SW Sam Jackson Park Rd., Portland, OR 97239, reiss@ohsu.edu), Holden D. Sanders, and Nicole Dean (Oregon Health and Sci. Univ., Portland, OR)

Many adults with hearing loss who use hearing aids (HAs) and/or cochlear implants (CIs) have broad binaural pitch fusion, such that sounds with large pitch differences are fused across ears, leading to difficulties separating voices in multi-talker environments. However, there is individual variation. The goal of the current study was to investigate whether auditory experience factors explain this variation in broad binaural fusion. Binaural fusion was measured in adults with various hearing device combinations: 26 bilateral HA, 25 bimodal CI, and 27 bilateral CI users. Fusion ranges were measured by simultaneous, dichotic presentation of reference and comparison stimuli in opposite ears, and varying the comparison stimulus to find the range that fused with the reference stimulus. Factors examined included age at testing, degree of HL, amount of amplification, age of onset of HL, duration of HL without CI, and duration of HA, bimodal CI, and bilateral CI use. Preliminary analyses suggest that broad fusion is correlated with long durations of HA use; further analyses using multivariable models will be described. The findings will indicate how hearing device experience may influence binaural pitch fusion. [Work supported by NIH grant R01 DC013307.]

1aPP35. The theoretical maximum intelligibility improvement that can be provided by hearing aids. Eric M. Johnson (Dept. of Commun. Sci. and Disord., West Virginia Univ., 375 Birch St., Morgantown, WV 26506, eric.johnson5@hsc.wvu.edu)

One goal of hearing aids is to increase the intelligibility of speech by improving the signal-to-noise ratio through the use of directional microphones and/or digital noise reduction. With this objective in mind, an ideal hearing aid would output amplified target speech with a signal-to-noise ratio of $+\infty$. However, even in this ideal scenario, vent effects would allow background noise to enter the ear canal via the direct sound path while also allowing amplified sound to escape from the ear canal. In this experiment, a range of hearing aid fittings were simulated to account for these vent effects. Hearing-impaired listeners were tested using the ideal simulated hearing aid, and the results will show the maximum theoretical hearing aid benefit.

1aPP36. Modified rhyme test performance and error patterns predict self-reported frequency of temporary threshold shifts for normal hearing listeners. Gregory M. Ellis (Audiol. and Speech Pathol., Walter Reed National Military Medical Ctr., 4494 Palmer Rd. N, Bethesda, MD 20814, gellis@alakaina.com), Matthew J. Makashay (Defense Centers for Public Health - Aberdeen, Aberdeen Proving Ground, MD), and Douglas S. Brungart (Audiol. and Speech Pathol., Walter Reed National Military Medical Ctr., Bethesda, MD)

After exposure to a loud sound, listeners may report that their hearing is dull or muffled. If hearing returns to pre-exposure levels, the change is considered a temporary threshold shift (TTS). TTSs are likely associated with damage to the auditory system. How frequently TTSs are experienced has been shown to be related to performance deficits on tasks like the modified rhyme test (MRT), even if audiometric thresholds are normal. The present study seeks to better understand the types of phonetic errors listeners with different frequencies of TTSs make on the MRT. Over 4000 US Service Members (SMs) completed the study. SMs were asked how frequently they experienced a TTS: "Never," "Infrequent" (<1/year), or "Frequent" (≥ 1/year). SMs completed 80 trials of a version of the MRT where SNR and target level were manipulated. Responses were analyzed using information theory and statistical modeling. Overall proportion correct, reaction time, and error rates associated with phonetic features were predictors of TTS frequency for normal hearing listeners. [The views expressed in this abstract are those of the authors and do not necessarily reflect the official policy of the Department of Defense or the U.S. Government.]

1aPP37. Improving speech understanding for face-to-face communication in noise when wearing hearing protectors. Anthony J. Brammer (Medicine, Univ. of Connecticut School of Med., Farmington, CT), Rahim Soleymanpour, Kia Golzari (Biomedical Eng., Univ. of Connecticut, Farmington, CT), Erin Heiney, Hillary Marquis (Medicine, Univ. of Connecticut School of Med., Farmington, CT), and Insoo Kim (Medicine, Univ. of Connecticut School of Med., 263 Farmington Ave., Farmington, CT 06030, ikim@uchc.edu)

Active hearing protection devices (eHPDs) are commercially available that enhance communication at low noise levels by amplifying both speech and noise. A study has been performed to develop algorithms for improving face-to-face speech communication in both low- and high-level noises when wearing eHPDs. The (mixed) environmental noise and speech at frequencies from 200 to 6000 Hz are divided into 16 or 24 contiguous subbands corresponding to 1.5 and 1 times the bandwidth of auditory filters, respectively. Signals are processed in the time domain with the overall group delay intended to maintain synchronization between speech and lip movements. The temporal modulation in each subband is band-limited to 2–16 Hz using zero time delay, moving detrend, high-pass, and moving average low-pass filters. This modulation is compared to the normalized unfiltered modulation in the subband to provide an estimate of the speech signal-to-noise ratio (eSNR). A threshold is selected for the eSNR above which time signals in subbands are combined to produce the instantaneous output and below which subband signals are attenuated. Subjects "wearing" a simulated HPD and listening to speech in industrial noise experience, on average, a 12% increase in intelligibility when using the algorithm. [Work supported by NIOSH and the Alpha Foundation.]

1aPP38. Self-administered, internet-enabled, modified rhyme test (MRT) for evaluating consonant confusion in remote subjects. Anthony J. Brammer (Medicine, Univ. of Connecticut School of Med., Farmington, CT), Kia Golzari, Rahim Soleymanpour (Biomedical Eng., Univ. of Connecticut, Farmington, CT), Erin Heiney, Hillary Marquis (Medicine, Univ. of Connecticut School of Med., Farmington, CT), and Insoo Kim (Medicine, Univ. of Connecticut School of Medicine, 263 Farmington Ave., Farmington, CT 06030, ikim@uchc.edu)

A self-administered MRT has been developed for persons unable to attend on-site for testing. Subjects remotely log on to a website and are instructed to use their own computer and headphones or earphones/earbuds and seek a quiet location without distractions for the test. After inputting personal identification, the software produces the maximum and minimum intensity sounds that will be heard to allow the subject to adjust the audio

gain to a comfortable listening level. The subject is then instructed not to re-adjust the audio gain during the MRT. Each trial is initiated by the subject and consists of the forced choice of one of six rhyming words, which is embedded in a carrier sentence. The pre-recorded speech in a single test, consisting of 25 trials, is replayed at a pre-selected, fixed speech signal-to-noise ratio. The internet-enabled MRT has been validated for subjects with believed normal hearing (N = 18) by comparison with a conventional MRT in our audiology clinic in which subjects' hearing thresholds were determined, and stimuli were presented at a prescribed sensation level (N = 7). Results were also compatible with word scores obtained on-site without audiological controls when subjects wore high-fidelity headphones (N = 6). [Work supported by NIOSH and the Alpha Foundation.]

1aPP39. Role of spatial positioning strategy for speech recognition in a simulated environment: Effects of age and hearing loss. William J. Bologna (Speech-Lang. Pathol. & Audiol., Towson Univ., 8000 York Rd., Towson, MD 21252, wbologna@towson.edu), Katie Esser, Karina Ball, Elaine Shaw, and Courtney King (Speech-Lang. Pathol. & Audiol., Towson Univ., Towson, MD)

In realistic listening environments, the physical position of the listener and target talker, relative to competing talkers, affects the signal-to-noise ratio and availability of spatial cues for release from masking. While the importance of signal-to-noise ratio and spatial cues for speech recognition has been demonstrated in the laboratory, a listener's ability to leverage these acoustic cues by manipulating their position in an environment has received less attention. This study evaluated how changes in listener position within a simulated environment effect speech recognition for four listener groups that differed in age (younger or older) and hearing status (normal or impaired hearing). The simulated environment consisted of 8–10 speech maskers, individually rendered under headphones for specific locations within the space, with target and listener positions that varied across conditions. The intensity of the target varied throughout testing to estimate the psychometric function for keyword recognition of each listener in each position. Subjective reports of preferred listening position were collected before and after testing to determine the extent to which these listener groups strategically position themselves to maximize their speech recognition in noise. Results will be discussed in terms of the potential gains in speech recognition associated with optimal positioning strategies.

1aPP40. Portable automated rapid testing: Validation of automated testing on older listeners from India. Prashanth Prabhu, Vardha Pattundan, Sonal Priya, Dibyendu Das (All India Inst. of Speech and Hearing, Mysuru, India), Chhayakanta Patro (Speech-Lang. Pathol. and Audiol., Towson Univ., Towson, MD), Nirmal Kumar Srinivasan (Speech-Lang. Pathol. and Audiol., Towson Univ., 8000 York Rd., Towson, MD 21252, nsrinivasan@towson.edu), and Frederick J. Gallun (Oregon Hearing Res. Ctr., Oregon Health and Sci. Univ., Portland, OR)

Portable automated rapid testing (PART) is an iPad application (https://braingamecenter.ucr.edu/games/p-a-r-t/) capable of measuring various psychoacoustic thresholds in an automated and rapid format using commercially available headphones. The app has been validated previously in native speakers of English and an adapted version of the app has been validated in young normal hearing listeners from Mexico. Previous research from our labs indicated that the thresholds obtained using non-speech tasks did not differ significantly between the native and the non-native speakers of English while the native speakers of English had significantly better thresholds in the speech task. However, there was no significant difference in the amount of spatial release from masking between the groups. Here, we present psychoacoustic threshold data for a large cohort of older listeners from India with various English proficiencies. The thresholds obtained from this group will be compared against previously published thresholds based on native speakers of English. It is hypothesized that the results would be similar to what was found earlier with younger listeners from India. These results will give us the evidence to start using this app to measure thresholds of various central auditory processing measures in non-native speakers of English.

1aPP41. A novel method for measuring perception of suprasegmental cues in speech. Jing Shen (Commun. Sci. and Disord., Temple Univ., 1701 N. 13th St., Philadelphia, PA 19122, jing.shen@temple.edu) and Yimin D. Zhang (Elec. and Comput. Eng., Temple Univ., Philadelphia, PA)

While the perception of segmental cues in speech has been extensively studied, evidence on the perception of suprasegmental cues is largely missing. A critical barrier in this area is the challenge in measuring listeners' ability to perceive a combination of multiple acoustic cues. The classic paradigm of discrimination is limited by its heavy reliance on auditory short-term memory and a long testing time. This study aims at examining the utility of a novel measure that is potentially more efficient and offers a more fine-grained account of perception of suprasegmental cues. Built on the evidence that listeners can shadow and rapidly imitate speech cues faithfully, we use the degree of alignment between listeners' rapid imitation of suprasegmental cues and that of the speech stimuli as an index of perception quality. This alignment is mathematically quantified by the dynamic time warping (DTW) method, which is modified to simultaneously account for both the intensity and the fundamental frequency sequences to render more reliable alignment results. Data to date provided rich information regarding individual differences in perception, and results obtained from rapid imitation showed improved utility over those from the discrimination. The theoretical and clinical implication of this work is also discussed.

1aPP42. Evaluation of efferent influences on neural coding using preclinical models of sensorineural hearing loss. Afagh Farhadi (Purdue Univ., West Lafayette, IN, afarhadi@purdue.edu), Samantha Hauser, Andrew Sivaprakasam, and Michael G. Heinz (Purdue Univ., West Lafayette, IN)

The medial olivocochlear (MOC) efferent system is less explored than the ascending auditory pathway but likely contributes in important ways to neural coding and perception. These effects are thought to vary across stimulus configurations, anesthetic states, and subtypes of sensorineural hearing loss (SNHL). To explore effective assays of MOC effects on neural coding, we have recorded Interleaved otoacoustic emissions (OAEs) and envelope following responses (EFRs) from several pre-clinical SNHL chinchilla models. Our preliminary observations include increases in OAEs with inner-hair-cell loss and anesthesia, which may be due to reduced efferent strength. Additionally, we observed enhanced EFRs with an added noise masker, which also could be related to efferent effects. We are using sedated and awake OAE comparisons to develop a standard efferent assay for use in neural-coding studies. Interleaved recording of OAEs and EFRs track cochlear-gain and neural-coding changes during acoustic stimuli. A recently developed modeling framework (Farhadi *et al.*, 2023 JASA) that includes different MOC projection pathways, including midbrain modulation-sensitive inputs, is used to guide most-effective stimulus selection. Ultimately, this model-guided experimental framework will provide unique guidance for testing MOC hypotheses related to neural coding.

1aPP43. Predicting self-reported hearing problems using suprathreshold auditory processing and cognitive measures. Tess K. Koerner (VA RR&D NCRAR, 3710 SW US Veterans Hospital Rd., Portland, OR 97239, koern030@gmail.com), Karen Garcia, Lauren Charney, Conner Corbett (Oregon Hearing Res. Ctr., Oregon Health and Sci. Univ., Portland, OR), Sebastian Lelo de Larrea-Mancera (Northeastern Univ., Boston, MA), G. Christopher Stecker (Ctr. for Hearing Res., Boys Town National Res. Hospital, Omaha, NE), Aaron Seitz (Northeastern Univ., Riverside, CA), and Frederick J. Gallun (Oregon Hearing Res. Ctr., Oregon Health and Sci. Univ., Portland, OR)

Measures typically included in standard audiometric test batteries often fail to correlate with self-reported auditory complaints. This is especially common for patients with normal- or near-normal-hearing sensitivity, like those who are aging or have a history of traumatic brain injury (TBI). This discrepancy likely stems from the use of measures that are not sensitive to auditory or cognitive processing deficits underlying patient complaints, which limits effective clinical assessment and management. Recent work

has focused on making common laboratory-based measures more clinically accessible using a tablet-based app called PART. The goal of the current study was to identify PART measures that better reflect patient complaints. Data was collected from adults with (n = 37) or without (n = 36) a history of TBI that were matched in age and hearing sensitivity. Regression modeling was used to determine relationships between self-reported auditory difficulties and measures of auditory processing and cognition, including spectro-temporal modulation detection, frequency modulation detection, spatial release from masking, and the Auditory Visual Divided Attention Task (AVDAT). Items from the Concussion Symptom Subtype Inventory were used to measure self-reported auditory complaints. Results contribute to a better understanding of processing difficulties that underly auditory complaints in those with and without TBI.

1aPP44. Performance of logistic regression analyses in the identification of inner ear synapse pathology. JoAnn McGee (VA Loma Linda Healthcare System, Loma Linda, CA 92357, mcgeej@umn.edu), Xiaohui Lin, Hongzhe Li, Jonathan H. Venezia, Marjorie R. Leek, and Edward J. Walsh (VA Loma Linda Healthcare System, Loma Linda, CA)

Currently, a diagnostic tool to identify cochlear synaptopathy in humans is unavailable. However, several features of electrophysiological responses evoked by auditory stimuli have been evaluated that may serve as useful indicators of synapse pathology. This study is part of a larger project whose goal is to develop a statistical model designed to accurately and reliably detect cochlear synaptopathy in humans. Univariate logistic regression analyses were employed to identify electrophysiological outcomes that predict synapse pathology in a guinea pig model and the relative performance of each model was evaluated. Previous efforts to assess model performance included area under the Receiver Operating Characteristic curve. In this report, metrics, such as F1-score and Matthews Correlation Coefficient (MCC), were included. Expectations are that analyses that incorporate true negatives, as does MCC, will more completely describe the performance of the binary classification of interest here: synapse pathology or synapse normalcy. Findings will be presented in the context of a non-human mammalian model with the ultimate purpose of developing a statistical model that can be used to optimize the diagnosis of synapse pathology in humans. [Work supported by the Department of Defense Award #W81XWH-19-1-0862.]

1aPP45. An open source platform for hearing research. Odile Clavier (Creare LLC, 16 Great Hollow Rd., Hanover, NH 03755, ohc@creare.com), Chris Brooks, and Brian Graybill (Creare LLC, Hanover, NH)

The disparate nature and format of data obtained from existing commercial hearing test systems is a significant impediment to reproducibility of hearing research. Advancement of hearing diagnostic assays requires access to research-grade platforms that are easily customized while enabling broad collaboration across labs with access to varied populations. In this research, we present an open-source platform that combines the Tympan open-source audio processing device with the TabSINT open-source tablet-based hearing software The Tympan uses a Teensy 4.1 processor which leverages the Arduino Development environment, making it accessible to a wide variety of users to create, modify and configure the embedded software. The Tympan is also a powerful audio processing platform that has demonstrated good performance as a hearing aid in a study of 14 adult users with mild to moderate hearing loss. Separately, TabSINT has been used extensively across the Department of Defense for human studies of hearing that include a variety of speech-in-noise tests and questionnaires, with thousands of subjects tested over the past few years. In this project, the two platforms come together, with new hardware extensions, to create a powerful hearing research platform that is extensible and highly accessible, thanks to the low cost of the hardware.

1aPP46. Estimating thresholds with the adaptive scan method of psychophysical testing. E. S. Lelo de Larrea-Mancera (Psych., Northeastern Univ., Ocaso 85, Insurgentes Cuicuilco, Coyoacán, Cdmx 04530, Mexico, elelo001@ucr.edu), Tess K. Koerner (VA RR&D NCRAR, Portland, OR), Eric C. Hoover (Dept. of Hearing and Speech Sci., Univ. of Maryland, Columbia, MD), G. Christopher Stecker (Ctr. for Hearing Res., Boys Town National Res. Hospital, Omaha, NE), Frederick J. Gallun (Oregon Hearing Res. Ctr., Oregon Health and Sci. Univ., Portland, OR), and Aaron Seitz (Psych., Northeastern Univ., Riverside, CA)

We previously introduced an adaptive psychophysical method called the adaptive scan (AS) that is readily available in the digital app PART. AS presents multiple progressive tracks (e.g., "scans") each of which comprises a range of stimulus values whose positioning adapts based on a listener's performance. Each scan is intended to contain trials above and below a listener's detection or discrimination threshold for an auditory cue, which is anticipated to increase familiarity with the tested cues. Previously, AS was validated against other known adaptive psychophysical methods (e.g., up-down staircases) and the method of constant stimulus using the QUEST algorithm. QUEST allowed for targeting the same point in the psychometric function (e.g., 80% threshold) in all methods tested. However, the QUEST procedure is parametric, requires many trials, and is not easily applied to the diversity of psychophysical tests that AS can be useful for. Here, we examine a variety of alternative approaches to estimate threshold from the AS procedure. This work adds to the usability of AS in particular and psychophysical testing in general.

1aPP47. Effects of noise exposure on peripheral auditory function, temporal envelope coding, and speech perception in service members with normal hearing. Angela Monfiletto (Speech Lang. Pathol. and Audiol., Towson Univ., 7903 Glen Dr., Towson, MD 21204, amonfi1@students.towson.edu), Nirmal Kumar Srinivasan (Audiol., Speech-Lang. Pathol., and Deaf Studies, Towson Univ., Towson, MD), and Chhayakanta Patro (Speech Lang. Pathol. and Audiol., Towson Univ., Towson, MD)

Service members face a significant risk of hearing impairment due to prolonged exposure to loud sounds in military settings. Detecting early signs of hearing loss is challenging, as they are often subtle and not easily identified through conventional clinical tests. In our study, we investigated the effects of noise exposure on electrophysiological, behavioral, and self-report measures of hearing damage in young adults, including both military and non-military individuals with normal audiometric thresholds. Participants completed a comprehensive test battery, covering standard and extended high-frequency pure-tone audiometry (0.25–16 kHz), a noise exposure structured interview, electrocochleography (ECochG), amplitude modulation (AM) detection threshold, and spatial release from speech-on-speech masking (SRM). Preliminary ECochG analyses suggested physiological deficits indicative of cochlear synaptopathy in service members. The discussion will focus on the correlation between physiological results and perceptual findings obtained through AM and SRM measures, aiming to understand the links between observed deficits and the perceptual challenges faced by service members in the context of subclinical hearing damage induced by noise exposure.

1aPP48. Hearing in the extended high frequencies and cochlear nerve output in the lower frequencies. Srikanta K. Mishra (Dept. of Speech, Lang., and Hearing Sci., Moody College of Commun., Austin, TX), Angela Monfiletto (Speech Lang. Pathol. and Audiol., Towson Univ., Towson, MD), and Chhayakanta Patro (Speech Lang. Pathol. and Audiol., Towson Univ., 326 Stevenson Ln., B8, Towson, MD 21204, cpatro@towson.edu)

Emerging studies suggest the perceptual importance of hearing in extended high frequencies (EHFs), such as 10–16 kHz. However, the

mechanisms by which EHFs may affect auditory functioning remain unclear. There is some evidence that hearing in the EHFs may be associated with reduced cochlear nerve output at lower frequencies. The objective of the present study was to examine the relationship between hearing in the EHFs and response parameters of electrocochleography (ECochG) reflecting hair cell functioning (summating potentials) and cochlear neurons (action potentials). We hypothesized that the relationship between hearing in the EHFs and cochlear nerve output is primarily due to altered cochlear non-linearity at lower frequencies. Tone-burst evoked ECochG and distortion product otoacoustic emissions (DPOAEs) input/output functions at 1000, 2000, and 4000 Hz were measured in adults with clinically normal audiograms. Hearing thresholds were also measured at 10, 11.3, 12.5, 14, and 16 kHz. Preliminary findings suggest that adults with normal audiograms can have elevated hearing in the EHFs. Results relating EHF thresholds, ECochG parameters, and DPOAE i/o functioning across frequencies will be presented and discussed in the context of early, subclinical cochlear damage. These findings will also help examine the claims of hidden hearing loss in humans.

MONDAY MORNING, 13 MAY 2024 ROOM 203, 10:00 A.M. TO 11:45 A.M.

Session 1aSA

Structural Acoustics and Vibration, Structural Acoustics and Vibration, Physical Acoustics, and Engineering Acoustics: Eric Ungar's Contributions to Structural Acoustics Research and Applications

Anthony L. Bonomo, Cochair
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School of Architecture, Rensselaer Polytechnic Inst., 110 Eighth St., Troy, NY 12180

Chair's Introduction—10:00

Invited Papers

10:05

1aSA1. Eric Ungar: Austria to Acentech—Highlights, contributions, reflections, and stories. Jeffrey A. Zapfe (Acentech, 33 Moulton St., Cambridge, MA 02138-1118, jzapfe@acentech.com)

This paper will describe Eric Ungar's journey from boyhood in 1920's Austria to his stature as one of the most significant contributors to the field of acoustics and vibration. I will go over some of Eric's highlights, major awards, and contributions. I will also describe how Eric has influenced me both before and after our time together at Acentech. And what discussion of Eric's career would be complete without some of the stories and anecdotes that show off the not so technical side of Eric as well.

10:30

1aSA2. Eric Ungar and vibration damping. James G. McDaniel (Dept. of Mech. Eng., Boston Univ., Boston, MA 02215, jgm@bu.edu)

This presentation honors Eric Ungar's contributions to vibration damping. The classic paper titled "Loss Factors of Viscoelastic Systems in Terms of Energy Concepts," which he co-authored with Edward Kerwin in 1962, takes center stage. Two significant contributions of the paper deserve detailed discussion. The first is the time dependence of total energy for viscoelastic systems with added mass, which results in ambiguous loss factor definitions off resonance. The second and most important contribution is an equation for the loss factor of a network of ideal viscoelastic springs. This equation reveals that the system loss factor is a weighted average of the loss factors of the viscoelastic springs, where the weights are the total energies in the viscoelastic springs. Next, the value of this work is illustrated by some recent work inspired by the 1962 paper. This recent work, which was published 60 years after the 1962 paper, involves the calculation of loss factors from finite element models of complex structures with viscoelastic elements. The presentation concludes with recollections that celebrate Eric Ungar's talent in educating and inspiring generations of engineers and researchers. [Work supported by ONR under Grant N00014-22-1-2785.]

10:55

1aSA3. A comparative analysis of damping models for the vibratory properties of a bent beam. Jerry H. Ginsberg (5661 Woodsong Dr., Dunwoody, GA 30338, j.h.ginsberg@comcast.net)

Much of Eric Unger's career was devoted to the study of dissipation mechanisms. This paper will explore the effect of various ways that damping effects have been incorporated in the solution of equations of motion. The analytical model of an L-shaped beam, cantilvered at one end and loaded at its free end by a force that is transverse to the plane of the bent beam, is the basis for the study. Equations of motion that account for coupling of flexure and torsion are created by merging the Ritz series method and the method of Lagrange multipliers. Energy is dissipated by a dashpot transversely mounted at the corner. An "exact" solution of the equations of motion using state-space modal analysis provides the reference for approximate methods including proportional damping, modal decoupling, and structural damping. Impulse response and complex frequency response at various levels of damping will be addressed.

11:20

1aSA4. Abstract withdrawn.

MONDAY MORNING, 13 MAY 2024 ROOM 214, 9:00 A.M. TO 11:00 A.M.

Session 1aSP

Signal Processing in Acoustics: Signal Processing in Acoustics Poster Potpourri

Trevor Jerome, Chair
Naval Surface Warfare Ctr., Carderock Div., 9500 MacArthur Blvd.,
BLDG. 3 #329, West Bethesda, MD 20817

All posters will be on display and all authors will be at their posters from 9:00 a.m. to 11:00 a.m.

Contributed Papers

1aSP1. Delay-constrained hearing aid speech enhancement using a wireless remote microphone. Austin Lu (Univ. of Illinois Urbana-Champaign, 1308 W Main St. MC 228, Urbana, IL 61801, austinl8@illinois.edu), Yongjie Zhuang (Elec. and Comput. Eng., Stony Brook Univ., Stony Brook, NY), Ryan M. Corey (Elec. and Comput. Eng., Univ. of Illinois Chicago, Chicago, IL), and Andrew C. Singer (Elec. and Comput. Eng., Stony Brook Univ., Stony Brook, NY)

Hearing aids have been shown in literature to offer better noise reduction and spatial cue preservation when used alongside a remote microphone. Oftentimes, this is in ideal conditions where both devices are synchronized and can transmit data losslessly and instantaneously. In practice, hearing aids tend to receive remote microphone data via wireless link, which incurs a slew of adverse effects. Among these, transmission delay is of particular importance, as hearing aids already have tight constraints on allowable end-to-end delay. Typically, this delay constraint is under 10 ms for perceptual reasons. In this work, we study a filter designed for hearing aids with a network-delayed remote microphone. Originally, the filter is derived and analyzed in the time frequency domain without explicit consideration of the delay constraints. We rederive the filter to address this gap, and seek to measure the effect of the network delay and delay constraints on overall performance.

1aSP2. Deep learning based recognition of multitarget underwater acoustic signals. PANXIANG Pan (Zhejiang Univ., 38 Zheda Rd., Hangzhou, Zhejiang 310027, China, panxiang@zju.edu.cn) and Jiabao Tan (Zhejiang Univ., Hangzhou, Zhejiang, China)

For enhancement of recognition of surface ships, a multitarget recognition framework is proposed based on combination of features confusion and Deep Learning. The AMS (Amplitude Modulation Spectrogram), MFCC (Mel Frequency Cepstral Coefficient), RASTA-PLP (Relative Spectral Transform-Perceptual Linear Prediction), GFCC (Gammatone Frequency Cepstral Coefficient), and Delta features are confused to extract static and dynamic features from ship- radiated noise. Due to the improvement of intra-class compactness and inter-class separability, better target recognition performance can be achieved in identifying five types of ships. Further data augmentation techniques, such as pitch shift, time stretch, and noise addition, are utilized to increase the amount of training data. The parameters of augmentation methods are chosen according to time delay spread and Doppler spread of underwater acoustic channels. The target recognition model's robustness can be enhanced due to increasing the dataset size. In design of the recognition framework based on deep learning, the residual structures are utilized to improve the speed of parameter optimization, and a multi-scale structure is integrated to extract local features. Meanwhile, an attention mechanism is implemented to focus the important information provided by key feature channels in the frequency domain, and online label smoothing is adopted to improve anti-overfitting ability. The effectiveness of the multitarget recognition framework is verified by the ShipsEar dataset .

1aSP3. Tone selection for cochlear implants using peak-finding and frequency bin overlap. James H. Keen (Phys., Univ. of New Orleans, 2000 Lakeshore Dr., New Orleans, LA 70148, jhkeen1@uno.edu) and Sanichiro Yoshida (Phys., Southeastern Louisiana Univ., Hammond, LA)

The Advanced Bionics cochlear implant devices employ a method of allocating static frequency bins to electrodes based on the natural tonotopic organization of the cochlea. Many studies show that there is a significant frequency-to-place mismatch in many subjects, but that over time the brain adjusts to these mismatches. Also, most of the tonal information in human speech is present in the lower end of the frequency spectrum, where there are few frequency bins and electrodes available to represent this information. Here, adjustments to the frequency bin allocation algorithm used in the crowdsourced CI Hackathon code are made to allow a more accurate representation of the original signal in the lower registers. A peak-finding and selecting method is used to find significant peaks regardless of frequency bin, then bins are overlapped to allow more accurate representation of tones near the borders between channels and peaks which may be near to each other.

MONDAY MORNING, 13 MAY 2024 — ROOM 215, 8:00 A.M. TO 11:50 A.M.

Session 1aUW

Underwater Acoustics, Acoustical Oceanography, Signal Processing in Acoustics, and Computational Acoustics: Data Science in Ocean Acoustics I

Alexander S. Douglass, Cochair
Oceanography, Univ. of Washington, 1501 NE Boat St., MSB 206, Seattle, WA 98195

Zoi-Heleni Michalopoulou, Cochair
Dept. of Math. Sci., New Jersey Inst. of Technol., 323 M. L. King Boulevard, Newark, NJ 07102

Tracianne B. Neilsen, Cochair
Phys. and Astronomy, Brigham Young Univ., N251 ESC, Provo, UT 84602

Haiqiang Niu, Cochair
Inst. of Acoust., Chinese Acad. of Sci., No. 21 North 4th Ring Road, Beijing 100190, China

Chair's Introduction—8:00

Invited Papers

8:05

1aUW1. Physics-informed neural network-based predictions of ocean acoustic pressure fields. Yongsung Park (Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0238, yongsungpark@ucsd.edu), Seunghyun Yoon (Seoul National Univ., Seoul, Republic of Korea), Peter Gerstoft (Univ. of California, San Diego, La Jolla, CA), and Woojae Seong (Seoul National Univ., Seoul, Republic of Korea)

Physics-informed neural network (PINN) trains the network using sampled data and encodes the underlying physical laws governing the dataset, such as partial differential equations (PDEs). A trained PINN can predict data at locations beyond the sampled data positions. The ocean acoustic pressure field satisfies PDEs, Helmholtz equations. We present a method utilizing PINN for predicting the underwater acoustic pressure field. Our approach trains the network by fitting sampled data, embedding PDEs, and enforcing pressure-release surface boundary conditions. We demonstrate our approach under various scenarios. By incorporating PDE information into a neural network, our method captures more accurate solutions than purely data-driven methods. This approach helps enhance the information content of sampled data when dealing with a limited amount of data.

8:25

1aUW2. Evaluation of data-driven neural operators in ocean acoustic propagation modeling. Haiqiang Niu (Inst. of Acoust., Chinese Acad. of Sci., No. 21 North 4th Ring Rd., Beijing 100190, China, nhq@mail.ioa.ac.cn)

Various types of neural networks have been employed to tackle a wide range of complex partial differential equations (PDEs) and ordinary differential equations (ODEs). Notably, neural operators such as DeepONet and FNO show promise in handling these problems, offering potential real-time prediction capabilities. In contrast to physics-informed neural network (PINN) methods, neural operators primarily derive insight from extensive, well-prepared datasets. In the context of ocean acoustic propagation modeling, where the challenge involves solving the wave or Helmholtz equation given specific boundary conditions, this study focuses on assessing the performance of data-driven neural operators in predicting sound pressure. Unlike conventional approaches that map between finite-dimensional Euclidean spaces, neural operators excel in learning mappings between infinite-dimensional function spaces—a particularly advantageous feature in sound propagation modeling tasks. This research specifically delves into evaluating the generalization capabilities of neural operators when applied to sound propagation modeling in a range-independent shallow water environment. By exploring the neural operators' effectiveness in this domain, the study aims to contribute valuable insights into their potential applications for real-world ocean acoustics simulations.

8:45

1aUW3. Ocean sound speed field reconstruction beyond tensor neural network. Lei Cheng (Zhejiang Univ., Zheda Rd., Hangzhou, Zhejiang 310058, China, lei_cheng@zju.edu.cn), Siyuan Li, Panqi Chen, Ting Zhang, and Jianlong Li (Zhejiang Univ., Hangzhou, Zhejiang, China)

Obtaining accurate ocean sound speed fields (SSFs) across a three-dimensional (3D) geographic region is vital for various underwater acoustic tasks. However, the scarcity of measurements due to the high cost of underwater sensors, combined with the high dimensionality of complex 3D SSF, makes the reconstruction problem highly ill-conditioned, thus demanding advanced models and methods. Our recent work has analyzed the reconstruction error and identified one promising way: finding a representation model that is both concise and expressive. Following this path, we proposed a tensor neural network (TNN) model, which leverages the conciseness of tensor models and the expressive power of deep learning. However, existing TNN-based approaches have two limitations: (1) they are unable to capture long-range correlations within the SSF and (2) they can only handle discrete-indexed tensor data. To overcome these limitations and fully unleash the power of deep tensor learning, we seamlessly introduce attention schemes into the existing TNN framework without compromising its interpretability. Additionally, we employ Gaussian process models to evolve the original parameterized tensor model into a new functional tensor model, enabling reconstruction with a continuous grid. Numerical results obtained from real-life datasets demonstrate the superior performance of our approach compared to state-of-the-art methods.

9:05

1aUW4. An overview of physics-based data science techniques in ocean acoustics. Mohsen Badiey (Elec. and Comput. Eng., Univ. of Delaware, 139 The Green, Rm. 140, Evans Hall, Newark, DE 19716, badiey@udel.edu), Jhon A. Castro-Correa, and Christian D. Escobar-Amado (Elec. and Comput. Eng., Univ. of Delaware, Newark, DE)

Scientific applications in ocean acoustics include inversion, signal detection, seabed classification, source localization, and environmental assessment. Traditionally, addressing these problems commonly rely on modeling, deterministic setups, or optimization methods. However, these solutions encounter challenges when faced with non-ideal data or experimental conditions, leading to ill-posed problems or assumptions that deviate from reality. Through a new approach in recent years the integration of data science and machine learning techniques has emerged as a promising solution across various research fields. Only recently have these methodologies been introduced to the field of ocean acoustics, offering efficient and alternative for addressing problems through data-driven approaches. This study presents a collection of techniques developed by the University of Delaware, focusing on acoustic and environmental assessment. The methodologies employed include statistical approaches like maximum entropy and data-driven methods, ranging from simpler strategies such as dictionary learning and image segmentation, to more sophisticated structures like convolutional neural networks and graph neural networks, the latter can exploit spatial information in the data. The results obtained from these diverse methods offer valuable insight, paving the way for innovative alternatives to physics-based problem-solving in ocean acoustics.

Contributed Papers

9:25

1aUW5. How will climate change affect the underwater sound environment? Modelling the potential effects of climate change on the oceanic speed of sound. Abigail Farkas (Stantec Consulting, 2100 Derry Rd. West, Ste. 400, Mississauga, ON L5N 0B3, Canada, abigail.farkas@stantec.com)

The purpose of this study is to create a year-by-year prediction model for surface-level oceanic speed of sound in different future climate scenarios and discuss how these results may be relevant for current and future work in underwater acoustics. It is already understood that sound propagation and speed in the oceans will be affected by future changes of oceanic pressure, temperature, salinity, and acidity. Certain trends of future sound speed have been analyzed in precedent studies; however, there are some information gaps and the availability of future oceanic sound speed models for use by the acoustical industry is lacking. This study intends to fill in some of these gaps and initiate an interdisciplinary discussion between both research and industry professionals in the field of acoustics.

9:40–9:55 Break

9:55

1aUW6. Simultaneous integration of acoustic and environmental data in a graph-based framework for underwater waveguide analysis. Jhon A. Castro-Correa (Elec. and Comput. Eng., Univ. of Delaware, 139 The Green, 3rd Fl., Newark, DE 19716, jcastro@udel.edu) and Mohsen Badiey (Elec. and Comput. Eng., Univ. of Delaware, Newark, DE)

Accurate assessment of underwater acoustic propagation relies heavily on understanding variability in the waveguide. Typically, prevailing methods in the field use environmental data exclusively to evaluate water column fluctuations. This study breaks new ground by integrating a graph-based

approach, merging insight from acoustics and environmental data to comprehensively analyze ocean fluctuations. The proposed framework leverages data obtained during the Shallow Water '06 Experiment (SW06) from a densely measured region in the experimental area. The data used in this work include acoustic broadband signals below 1 kHz from a 48-element L-shape array and environmental measurements from 59 thermistors across 16 mooring arrays, affected by the passing of internal waves. The spatial relationship in the data, facilitated by sensor locations, is harnessed through a graph built upon underlying features from the datasets, providing a nuanced understanding of environmental variability in the water column. The proposed framework is not confined to theoretical constructs; it is substantiated through rigorous model evaluation on the measured data. This validation process robustly demonstrates the generalization power of the proposed architecture, affirming its effectiveness in a highly fluctuating water column and showcasing its potential for advancing underwater acoustic research. [Work supported by ONR Ocean Acoustics program.]

10:10

1aUW7. Ocean sound speed field reconstruction with graph regularized deep matrix factorization. Hangfang Zhao (College of Information Sci. and Electron. Eng., Zhejiang Univ., Hangzhou, China), Zhenrong Xia (College of Information Sci. and Electron. Eng., Zhejiang Univ., Hangzhou, China, 22231149@zju.edu.cn), and Xuegang Shi (College of Information Sci. and Electron. Eng., Zhejiang Univ., Hangzhou, China)

To reconstruct the ocean sound speed field (SSF), a matrix in horizontal could be formed by gridding the sea area, where some sparse entries were filled by the observation sound speed profiles (SSPs). The reconstruction of the SSF can be modeled as a matrix completion problem, using limited and noisy observation values to reconstruct the entire matrix. Furthermore, the spatial correlation of the SSF could be modeled by graph space model, utilizing with the low-rank property of the SSF matrix, the problem of graph regularization low-rank matrix completion was developed. The observed average SSPs in six cross section were estimated from the four underwater acoustic tomography moorings, and nine local SSPs were obtained from pressure inverted echo sounders (PIESs) inversion. By combining the method of deep matrix factorization and the graph regularization, the Graph Regularized Deep Matrix Factorization (GRDMF) method was proposed. The simulation results showed that the GRDMF performs better than the spatial interpolation algorithm in SSF reconstruction. Considering the application scenarios of deep-sea acoustic tomography, an average constraint (AC) item was designed additionally, and then a customized algorithm named GRDMF-AC is proposed. Finally, data of the experiment at the South China Sea in 2021 is processed to verify GRDMF-AC, the results showed that the reconstructed SSF captures mesoscale eddy in the experiment.

Invited Paper

10:25

1aUW8. Exploiting passive acoustic recordings of seismic survey datasets. Alexander S. Douglass (Univ. of Washington, 2010 AL, 1231 Beal, Ann Arbor, MI 48109, asdougl@umich.edu) and Shima Abadi (Univ. of Washington, Seattle, WA)

Marine seismic reflection surveys provide a dense and abundant dataset covering a large region of geological interest in the ocean. These data provide an opportunity for acoustic analysis with varying environmental characteristics, both in the water column and the seabed. In a typical survey, an airgun array is fired tens to hundreds of thousands of times, and hundreds of receiver channels record the acoustic signal reflected from the seabed after each shot. Additionally, the high amplitude signal broadcast from airgun arrays used in these surveys may be measured passively by hydrophones located close to the survey. In this work, we consider the data measured by Ocean Observatories Initiative (OOI) hydrophones adjacent to two seismic surveys, MGL1905, and MGL2104. In each case, a 6600 in^3 airgun array is fired ~every 37.5 m along multiple survey lines extending 10s to 100s of kilometers. Each survey generates multiple terabytes of acoustic data, and many of the shots are also captured by OOI hydrophones. This work aims to combine these two datasets to evaluate the acoustic behavior of airgun shots over a variety of conditions and examine the relationship between various environmental factors and acoustic propagation behavior. [Work supported by ONR.]

Contributed Papers

10:45

1aUW9. Convolutional neural networks for signal identification and extraction and seabed classification. Wesley E. Olson (Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, wolson4@byu.edu), Tracianne B. Neilsen (Phys. and Astronomy, Brigham Young Univ., Provo, UT), David P. Knobles (Phys., Knobles Sci. and Anal., Austin, TX), Jason D. Sagers (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Preston S. Wilson (Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Sound underwater source (SUS) charges can be used for seabed characterization experiments. Many SUS were deployed during experiments in the New England MudPatch during 2017 and 2022. The goal of this work is to automatically detect and extract SUS signals in the dataset and then perform seabed classification on the extracted SUS signals. A binary classifier CNN is trained on simulated SUS charges and measured ambient noise to detect if a SUS signal is present in one minute pressure waveforms. The trained CNN model is then used to identify and extract the SUS signals from the measured data on the 52-channel PROTEUS L-array deployed by Applied Research Laboratories, University of Texas at Austin. This automated extraction process expedites the identification and time alignment of hundreds of SUS signals. The extracted signals will then be input to a ResNet-18 network trained on synthetic signals to perform seabed classification using a catalog of 34 seabeds. Lessons learned from the automated identification and extraction process will be presented as well as a statistical analysis of the seabed classification results. [Work supported by the Office of Naval Research, Grant N00014-22-12402.]

11:00

1aUW10. A mode separation method based on sparse Bayesian learning for explosive sources in a shallow water waveguide. Xuedong Zhang (Inst. of Acoust., Chinese Acad. of Sci., No. 4 Ring Rd., Haidian Dist., Beijing 100190, China, xuedong.zh@outlook.com), Qi He (Big Data Ctr., State Grid Corp. of China, Beijing, China), and Zaixiao Gong (Inst. of Acoust., Chinese Acad. of Sci., Beijing, China)

This paper presents a mode separation method based on sparse Bayesian learning (SBL) for explosive sources in a shallow water waveguide. In

previous work [Niu *et al*., JASA, 2021, 4366], the SBL dictionary was constructed by assuming a large number of horizontal wavenumbers and utilized an approximate mode-frequency dispersion relation for low frequencies. Then, modes were separated in the frequency domain by estimating the coefficients of the dictionary atoms. However, challenges inherent to explosive sources, such as bandwidth expansion and the bubble-pulse effect, result in a mismatch in the dictionary matrix built using the approximate mode-frequency dispersion relation for low frequencies, leading to unsuccessful mode separation. To address these issues, this paper builds the SBL dictionary matrix by utilizing the acoustic model (e.g., Kraken) to derive horizontal wavenumbers under various environmental hypotheses and combines it with the secondary bubble-pulse model. By estimating the coefficients of the dictionary atoms, both environmental parameter estimation and mode separation can be achieved simultaneously. Simulation and experimental data results demonstrate the validation of the proposed method.

Invited Paper

11:15

1aUW11. Discovery and learning of feature geometry in underwater acoustic sensing: Case studies and challenges in active sonar applications. Ananya Sen Gupta (Dept. of Elec. and Comput. Eng., Univ. of Iowa, 103 S Capitol St., Iowa City, IA 52242, ananya-sengupta@uiowa.edu), Ivars P. Kirsteins (NUWCDIVNPT, Newport, RI), Andrew J. Christensen (Elec. and Comput. Eng., Univ. of Iowa, Iowa City, IA), and Timothy Linhardt (College of Elec. and Comput. Eng., Univ. of Iowa, Iowa City, IA)

Interpretable artificial intelligence (AI) and related machine learning (ML) techniques are gaining popularity for underwater acoustic sensing applications. However, interpretability of machine learnt features poses a fundamental challenge to successful application of these powerful data science techniques to underwater acoustics. Autonomous sonar target recognition is especially interesting from the data science perspective as beyond target-specific features, a sonar ping response typically includes significant interference from the environment, e.g., clutter, multipath scattering, etc. Such environmental interference, resulting from complex, dynamic, unpredictable, and often unknown factors, manifest as identifiable structures in acoustic color or alternate multi-dimensional feature representations that can lead to machine learning and classification errors. In this talk, we will discuss these challenges commonly encountered in underwater acoustics using case studies from field experiments in active sonar target recognition. Some results will also include physics-driven simulations in this domain to provide robust ground truths. In particular, we will posit how braided feature geometry and its representation, as well geometry of overlap between features can render a sonar target feature informative, explainable, and discoverable.
[Work funded by the ONR Grant Nos. N000142112420 and N000142312503 and DoD Navy (NEEC) Grant No. N001742010016.]

Contributed Paper

11:35

1aUW12. SONAR target classification with complex-valued neural networks. Timothy Linhardt (College of Elec. and Comput. Eng., Univ. of Iowa, 103 South Capitol St., Iowa City, IA 52240, tlinhardt@uiowa.edu), Ananya Sen Gupta (Dept. of Elec. and Comput. Eng., Univ. of Iowa, Iowa City, IA), Matthew Bays (NSWC, Panama City, FL), and Ivars P. Kirsteins (NUWCDIVNPT, Newport, RI)

Popular acoustic signal processing techniques analyze acoustic color, which is a magnitude representation of a frequency spectrum. This paradigm allows for easy visualization of results and simpler models at the cost of throwing out phase information. Analysis and modeling of complex-valued data does have inherent difficulty from the nature of complex numbers. Optimization on the complex field requires alternate partial derivative definitions to circumvent consequences of the Cauchy–Riemann equations regarding holomorphic functions. To make use of phase information, we demonstrate classifier model optimization with complex-valued parameters on data with both magnitude and phase components and show how complex neural networks yield marked improvements over similarly shaped real-valued networks in both classification accuracy and generalization ability. We apply these techniques to small target SONAR with simulated Lamb wave resonance signals for hollow spheres, differentiating between different material classes. [Research funded by the DoD Navy (NEEC) Grant No. N001742010016.]

Session 1pAA

Architectural Acoustics, Structural Acoustics, and Vibration and Noise: Sound Transmission and Impact Noise in Buildings II

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

Invited Papers

1:00

1pAA1. Development of a new acoustic prediction tool by integration of life cycle assessment. Mohamad Bader Eddin (Appl. Sci., Univ. of QC at Chicoutimi, 412 Rue Jacques-Cartier E, Chicoutimi, QC G7H 1Z3, Canada, Mohamad.bader-eddin1@uqac.ca), Sylvain Ménard (Appl. Sci., Univ. of QC at Chicoutimi, Chicoutimi, QC, Canada), and Bertrand LARATTE (École nationale supérieure d'Arts et Métiers, Paris, France)

Recently, environmental awareness has been the key driver for using renewable materials that have low environmental impact and fulfill constructional requirements, such as timber. Despite the advantages of wood as a building material, it has a lower subjective quality of sound insulation. To fulfill the sound insulation requirements, it is, therefore, unavoidable to complement based-wooden assemblies with additional element(s). However, identifying the acoustic performance is costly and time-consuming. Therefore, developing an accurate prediction tool is vital. Since wood-based structures have been developed to consider the environmental aspects, the environmental performance of buildings should be integrated into the acoustic design. This paper aims to develop an acoustic design methodology for wooden structures using artificial neural network approach by integration of life cycle assessment (LCA). Various Lab-based measurements are used to develop the acoustic prediction tool. Then, a LCA study is conducted on the test assemblies. This paper initially found that wooded assemblies generally increase the environmental impacts to achieve better acoustic insulation. Moreover, different assemblies can meet the sound insulation requirements. Therefore, designers should cognize of environmental and acoustic trade-off by selecting assemblies that consider both aspects.

1:20

1pAA2. Evaluating the measurement of the velocity level difference for building constructions which combine Type A and Type B elements. Jeffrey Mahn (National Res. Council Canada, 1200 Montreal Rd., Ottawa, ON K1A 0R6, Canada, jeffrey.mahn@nrc-cnrc.gc.ca), Markus Mueller-Trapet (National Res. Council Canada, Ottawa, ON, Canada), Sabrina Skoda, and Iara B. Cunha (Construction Res. Ctr., National Res. Council, Ottawa, ON, Canada)

The standard ISO 12354 outlines a method of predicting the transmission of structure-borne noise in buildings based on data measured in a laboratory or in the field, measured in part according to the standard ISO 10848. The ISO 10848 standard describes procedures for the measurement of both the normalized flanking level difference and the velocity level difference of flanking paths through different building constructions and the junctions between them. Both the normalized flanking level difference and the velocity level difference were measured for a hybrid construction of a cross-lamented timber floor (Type A element) connected to lightweight timber walls (Type B elements) in the National Research Council Canada's four room flanking facility. The predicted normalized level differences are compared against the measured values to evaluate the prediction method for constructions that mix Type A and Type B elements.

1:40

1pAA3. Sound transmission loss through steel-framed building partitions. Michael Raley (PAC Int., 2000 4th Ave., Canby, OR 97013, mraley@pac-intl.com) and Benjamin M. Shafer (PABCO Gypsum, Tacoma, WA)

Load-bearing steel-stud walls pose a unique acoustical challenge because both the spacing and the mil thickness of the studs can significantly affect the acoustical performance of these walls. A single wall type in a load-bearing steel building may actually include many acoustically distinct walls as the stud spacing and mil thickness change from floor to floor in the building due to the different loads. This can be especially challenging on delegated design projects where the final selection of the steel studs is left to the framing contractor and may not be included in the project documents supplied to the acoustical consultant, if there even is one on the project. In prior ASA presentations, Ben Shafer has clearly documented the effects of steel stud spacing and mil thickness on acoustical performance. This presentation adds to Ben's prior work by investigating the effects of resilient sound isolation clips on wall performance. Clips are a unique application because the spacing of both the clips, and the hat channels does not change with changing stud spacing and because the low stiffness of the rubber isolators should reduce the effect of stud stiffness. This presentation will present the results of recent lab tests of walls with resilient sound isolation clips and compare the results to the data previously presented by Mr. Shafer.

2:00

1pAA4. Sound transmission loss of vertical building partitions designed to meet shear structural requirements. Benjamin M. Shafer (PABCO Gypsum, 3905 N 10th St., Tacoma, WA 98406, ben.shafer@quietrock.com)

Vertical partitions often include structural building elements designed to meet shear requirements. Shear partitions are designed to resist lateral load due to wind or seismic activity and are required in many regions throughout North America. Nightingale *et al*. completed a 70-assembly series quantifying the effect of shear OSB on the sound transmission loss (STL) of both wood- and steel-framed single-stud assemblies in the National Research Council Canada (NRCC) research report *NRCC-45018*. This research study builds on the NRCC research with a test of over 200 wood and steel assembly partitions. This research study includes single-, staggered-, and double-stud framing as well as various forms of shear treatment such as OSB Plywood installed on one or both sides of the partition, attachment with various nail attachment patterns, and steel strapping. The effect of these partition elements on the STL will be presented and discussed. Additionally, the effect of sound isolation treatment, such as additional mass, resilient channels, resilient sound isolation clips, and constrained-layer damping will be presented and discussed.

2:20–2:35 Break

Contributed Papers

2:35

1pAA5. Studs' impact on sound transmission in double-leaf partitions. Julia Idczak (AGH Univ. of Krakow, al. Mickiewicza 30, Cracow 30-059, Poland, idczak@agh.edu.pl)

Double-leaf partitions constructed by two parallel plasterboard panels installed on steel studs are widely used in building constructions. Although it is known that the studs significantly influence vibroacoustic behavior of the plates, there has not been extensive research on this topic. This study focuses on the impact of steel studs on the sound transmission in double-leaf constructions. In order to fully analyze the impact of the steel stud element, a comparison between experimental data from a double leaf partition with a mechanical connection using steel studs and a theoretical model that disregards any mechanical connection between the parallel plasterboard panels was conducted. Results presenting radiation efficiency, transmission loss, and velocity level differences in a grid of points equally distributed on the radiating side of a structure highlight the influence of steel studs in double-leaf constructions indicating frequency ranges, where the studs' impact is mostly significant.

2:50

1pAA6. A hybrid test method for measurement of airborne sound transmission loss of building partitions and elements. Omid Tamanna (School of Construction and the Environment, BC Inst. of Technol., 3700 Willingdon Ave., Burnaby, BC V5G 3H2, Canada, otamanna@bcit.ca), Nazanin Souri, Won S. Ohm, and Maureen Connelly (School of Construction and the Environment, BC Inst. of Technol., Burnaby, BC, Canada)

Single-number ratings, such as sound transmission class (STC), outdoor to indoor transmission class (OITC), and weighted sound reduction index (R_w), have been widely used to indicate the sound transmission properties of the building elements. The required transmission loss values over the frequency spectrum can be obtained from the two reverberation rooms test method (ASTM E90, ISO 10140), field measurement method (ASTM E966), or intensity method (ISO 15186). The limitations of the number of testing facilities that can provide the diffused field, minimum cut-off frequency, and other standard laboratory requirements, made the sound transmission loss measurement a costly and time-consuming procedure. In this study, alternative methods to measure the sound pressure and intensity have been investigated. The main objective of this research was to eliminate the complicated requirements of the diffused field and the room absorption measurement by averaging the direct sound amplitude at several angle of incidents and using the intensity probe to measure the sound intensity. In conclusion, the sound transmission loss, STC, and OITC values of double-glazed windows obtained from the hybrid method have been compared with predictions from the existing theory and the ASTM E90 standard test results.

3:05

1pAA7. A meta-analysis of sound isolation properties of various modular prefabricated wall systems. Jean-François Latour (Noise Control Div., Mecart, 110 Rotterdam St., St-Augustin-de-Desmaures, QC G3A 1T3, Canada, jflatour@mecart.com)

The current meta-analysis focuses on airborne sound isolation performances for modular prefabricated wall panels. As many properties may be varied in prefabricated wall assemblies, relying only on measurements either on prototypes in a lab or *in situ* tests would require a significant amount of resources. Understanding the precision and limitations of theoretical evaluation methods is, therefore, a must. A comparison of theoretical predictions with lab and field measurements (as per ASTM E90 or E336) is performed on multilayer systems up to a double wall panel system which incorporates four layers (two panels each composed of two layers). This allows a better appreciation of theoretical predictions of modular prefabricated wall systems.

3:20

1pAA8. Robust 3D localization of anomalies in reverberation time measurements using the sound field scanning method. Thomas Rittenschober (Seven Bel GmbH, Hafenstrasse 47-51, Linz 4020, Austria, thomas.rittenschober@seven-bel.com) and Rafael Karrer (Seven Bel GmbH, Linz, Austria)

Reverberation time measurements form the basis for room acoustic optimizations of existing building structures. During the verification of the achieved room acoustic improvements, anomalies may appear in the reverberation time signal which may be hard to spatially localize, especially in spaces with demanding acoustic requirements such as large, open workspaces, or concert halls. This contribution focuses on the application of the Sound Field Scanning technology to the fast spatial localization of room reflections. In this process, an omnidirectional sound source is positioned at an observation point in the room and periodically excited with band-limited pulses. At the same observation point, an acoustic camera system consisting of a rotating linear microphone array is oriented towards the preferred spatial direction. The emitted pulses and associated room reflections are captured on the measurement surface of the rotating microphone array. Acoustic images with high depth resolution are generated in parallel planes to the measurement surface. In complex situations, the task of spatially localizing anomalies in the reverberation time signal can be reduced to a few measurements from different viewing angles, thus, significantly accelerating the problem-solving process with high confidence. The method is exemplarily described through the room acoustic analysis of a University Lecture Hall.

3:35

1pAA9. Introducing the sound transmission loss suite at the British Columbia Institute of Technology. Won S. Ohm (School of Construction and the Environment, BC Inst. of Technol., 3700 Willingdon Ave., Burnaby, BC V5G 3H2, Canada, won_ohm@bcit.ca) and Omid Tamanna (School of Construction and the Environment, BC Inst. of Technol., Burnaby, BC, Canada)

A sound transmission loss suite is a facility for measuring the airborne sound insulation by building elements such as wall assemblies and partitions. It consists of two neighboring reverberation chambers (called the source and receiving rooms), where the only significant sound transmission path is presented by the test specimen, enclosed in the common wall separating the two chambers. In this talk, an overview of the sound transmission loss suite that was newly built and commissioned at the British Columbia Institute of Technology in late 2023 is given. The suite is comprised of two reverberation chambers with interior volumes of 200 m^3 (source room) and 125 m^3 (receiving room) and can test for frequencies from 63 to 20000 Hz, making it one of a kind in Western Canada. The talk touches upon the acoustical characteristics of the suite and the relevant test method for measuring transmission loss according to the ASTM E90 standard. [Work supported by Canada Foundation for Innovation—Project No. 36346.]

3:50

1pAA10. Commissioning of National Research Council Canada's four room flanking facility. Ivan Sabourin (National Res. Council Canada, 1200 Montreal Rd., Ottawa, ON K1A 0R6, Canada, ivan.sabourin@nrc-cnrc.gc.ca), Jeffrey Mahn, Markus Mueller-Trapet, Iara Batista da Cunha, and Sabrina Skoda (National Res. Council Canada, Ottawa, ON, Canada)

The National Research Council Canada has built a new four room flanking facility to expand its capability to test flanking noise transmission across junctions of typical multi-dwelling units. The cast-in-place concrete structure is configured to construct two rooms on top of two rooms. This facility allows various configurations to measure flanking noise: one continuous or two discontinuous floors (or walls) and a common façade (side wall). With the overhead crane, it is possible to install heavy monolithic elements such as precast concrete and cross-laminated timber panels such that are used in mid and high-rise buildings. There are structural breaks between the rooms to reduce unwanted structural transmission and the addition of shielding panels makes this facility capable of measuring flanking of high performing junctions/assemblies. The facility has been designed to meet the requirements of ISO 10848 for both airborne and impact noise transmission. Facility design details and commissioning data will be presented.

MONDAY AFTERNOON, 13 MAY 2024 — ROOM 210, 1:00 P.M. TO 4:15 P.M.

Session 1pBAa

Biomedical Acoustics: General Topics in Biomedical Acoustics: Imaging

John M. Cormack, Chair
Division of Cardiol., Dept. of Med., Univ. of Pittsburgh, Pittsburgh, PA 15261

Contributed Papers

1:00

1pBAa1. Native bubble nuclei for acoustic cavitation in 3D cell cultures. Ferdousi Sabera Rawnaque (Penn State Univ., 903 W Aaron Dr Apt F, State College, PA 16803, fmr5186@psu.edu) and Julianna C. Simon (Penn State Univ., University Park, PA)

The safety of biomedical ultrasound largely relies on controlling cavitation bubbles *in vivo*; however, bubble nuclei in biological tissues remain unexplored compared to water. No previous studies have observed where bubble nuclei form in tissues at a microscopic level or how the variation in tissue biomechanical properties can impact the presence of bubble nuclei for acoustic cavitation. In this study, we evaluated whether bubble nuclei form intracellularly or extracellularly in rat epithelial hepatoma (McA-RH7777) and musculoskeletal myoblast (L6) cell lines ($n = 5$ each), cultured in 3D using Matrigel™ scaffolds. A 3.68 MHz focused ultrasound transducer with f# = 1 was used to induce low-density cavitation using varying pulse lengths (10 to 100 μs) with pressures ranging up to $p+ = 29$ and $p- = 12$ MPa. The spatial location of the bubbles was monitored microscopically using high-speed photography at 20 000 fps. Despite significant radiation force on the cell scaffold, preliminary results show that acoustic cavitation bubbles ($r = 20$ μm approx.) preferentially form extracellularly near the cell membranes. This result suggests that the hydrophobicity in the amphipathic cell membrane may contribute to bubble nuclei formation. Future work includes investigating the distribution of bubble nuclei in healthy versus cancerous cell cultures. [Work supported by NSF CAREER 1943937.]

1:15

1pBAa2. Real-time assessment of focused ultrasound-induced bioeffects in elastic tissues. Jacob C. Elliott (Graduate Program in Acoust., Penn State Univ., Res. West, State College, PA 16801, jce29@psu.edu), Grace M. Wood, and Julianna C. Simon (Graduate Program in Acoust., Penn State Univ., University Park, PA)

Highly elastic tissues have proven resistant to fractionation via focused ultrasound (fUS); however, our previous work in rat tendon has demonstrated a small window of parameters conducive to mild mechanical disruption. For therapeutic applications, there is a need to assess the extent of fUS-induced mechanical bioeffects in real time in order to avoid over- or under-treatment. Here, elastic collagen hydrogels (TeloCol®-10), as well as healthy and collagenase-soaked *ex vivo* bovine tendons, were exposed to fUS at 1.1–3.68 MHz ($p+ \leq 127$ MPa, $p- \leq 35$ MPa) using 10-ms pulses repeated at 1 Hz. Cavitation signals were collected using simultaneous passive cavitation imaging (PCI) and passive cavitation detection (PCD) to monitor fUS treatment in real time. Preliminary data in polyacrylamide hydrogels and *ex vivo* bovine tendon show no consistent trends between

simultaneous PCI and PCD signals; this is potentially due to different orientations of the receiving transducers, which we will further investigate. However, neither PCI nor PCD trends were consistently linked to a mechanical bioeffect. Therefore, we are exploring the addition of Doppler ultrasound to PCI/PCD to help link the fUS exposure to the desired bioeffect. [Work supported by NIHR01EB032860]

1:30

1pBAa3. The role of fluid flow patterns in microbubble-mediated endothelial cell membrane permeabilization. Elahe Memari (Phys., Concordia Univ., 7141 Sherbrooke St. West, Montreal, QC H4B 1R6, Canada, Ememari91@gmail.com) and Brandon Helfield (Phys., Concordia Univ., Montreal, QC, Canada)

Blood flow dynamics vary throughout the circulatory system, influencing the pathophysiology of vascular endothelium. We investigated endothelial cell response to ultrasound-stimulated microbubbles across diverse anatomical sites by mimicking assorted blood flow patterns. First, we examined the effect of culture condition on cell sensitivity to sonication by culturing HUVECs either statically or under pulsatile flow (8 or 16 dyn/cm^2) for two days. Flow chambers were then co-perfused with microbubbles and propidium iodide under pulsatile flow (8 or 16 dyn/cm^2) and sonicated (1MHz, 20 cycles, 1ms PRI, 300kPa) using an acoustically coupled microscope. Additionally, in a subset of studies we investigated ultrasound-assisted endothelial permeability under both pulsatile and oscillatory flow patterns. Compared to static, cells cultured under pulsatile flow resulted in 1.2 to 2.0-fold increase in the percentage of permeabilized endothelial cells ($p < 0.0001$). Next, compared to sonication under laminar flow (15–30 ml/min), endothelial permeability was significantly augmented under pulsatile flow, ranging from 1.3 to 2.1-fold. Additionally, compared to laminar flow, oscillatory flow at ~16 ml/min with 0.5 Hz oscillation resulted in a 1.76-fold enhancement in cell perforation, yet yielded no observable permeability at slower flow rates (~8 ml/min). These findings highlight the influence of local blood flow dynamics on ultrasound-mediated cell permeabilization.

1:45

1pBAa4. Focused ultrasound and microbubble induced changes in the phenotype of breast cancer cell lines. Dure S. Khan (Ctr. for Neurosci. Studies, Queen's Univ., 18 Stuart St., Kingston, ON K7L 3N6, Canada, 20dsk1@queensu.ca), Rachel E. Rubino, Christopher J. Nicol (Div. of Cancer Biology & Genetics, Cancer Res. Inst., Queen's Univ., Kingston, ON, Canada), and Ryan Alkins (Ctr. for Neurosci. Studies, Queen's Univ., Kingston, ON, Canada)

Up to 10%–15% of patients with breast cancer (BC), particularly those with triple negative (TNBC) and HER2 positive subtypes, develop brain metastases. Despite aggressive treatment, the median survival for patients is 15 months. Treatment of brain metastases is challenging due to the blood–brain barrier (BBB) that limits the passage of molecules commonly used to treat primary BC into the brain parenchyma. Focused Ultrasound (FUS) and Microbubble (MB) are a non-invasive, image-guided therapeutic modality that can create a reversible, safe, and transient opening of the BBB. While this burgeoning area of research has progressed to clinical trials, the direct effects of FUS + MBs in the absence of therapeutic agents on tumor cells are poorly characterized. Therefore, this study aims to identify the FUS+MB induced changes in migration, invasion, and proliferation of representative brain-derived (BD) BC cell lines, namely, BD-MDA-MB-231 and BD-SKBR3 post-sonication in comparison to untreated cells. Results from these experiments will help guide future *in-vivo* studies evaluating the impact of FUS+MB on brain metastases to better understand its therapeutic implications and applications for patients.

2:00

1pBAa5. Coupling quantitative ultrasound using echo envelop statistics and shear wave propagation to provide new image contrast of mimicked liver lesions. Arnaud Héroux (Lab. of Biorheology and Medical Ultrason., Univ. of Montreal Hospital Res. Ctr., 900 Rue Saint-Denis, Montréal, QC H2X 0A9, Canada, arnaud.heroux@umontreal.ca), François Destrempes, and Guy Cloutier (Lab. of Biorheology and Medical Ultrason., Univ. of Montreal Hospital Res. Ctr., Montréal, QC, Canada)

In quantitative ultrasound, biomarkers are sensitive to the scatterers' positioning and number density, and therefore, shear wave (SW) propagation may modify their time-varying behaviors and provide new image contrast. Herein, we considered echo envelop statistics by using one parameter of the homodyned-K distribution (HKD): the diffuse-to-total signal power ratio $1/(\kappa+1)$. A liver lesion-mimicking phantom was simulated with an inclusion and a background of elastic moduli of 40 and 4 kPa, and scatterers' number densities per resolution cell of 15 and 5, respectively. Ten sets of spatially correlated scatterers were created, and a plane SW was simulated to reposition time and spatially varying scatterers according to the SW motion. Ultrasound images were simulated using MUST with added Gaussian noise to attain 20-dB SNR. Dynamic and static analyses were made over 30 frames with and without SW motion, respectively. Validation was based on the contrast of $1/(\kappa+1)$ between the inclusion and the background. HKD image contrast increased with SW motion with values from 0.12 ± 0.10 to 0.48 ± 0.16 (no units) ($p < 0.001$). Results suggest that the variation in scatterers' organization under SW propagation may provide new information over its static counterpart to enhance the contrast of liver lesions.

2:15

1pBAa6. A spectral Doppler ultrasound method for estimation of skeletal muscle velocity. Elizabeth L. Suitor (School of Eng. and Appl. Sci., Harvard Univ., 150 Western Ave., 4.101-10, Allston, MA 02134, esuitor@fas.harvard.edu), Yichu Jin, Sofia Cerasi, Robert D. Howe, and Conor J. Walsh (School of Eng. and Appl. Sci., Harvard Univ., Allston, MA)

Skeletal muscle velocity is a key indicator of neuromuscular function, and monitoring its changes plays important role in tracking the progression of musculoskeletal diseases, injuries, and fatigue. However, existing methods for non-invasive estimation of skeletal muscle velocity primarily use B-mode ultrasound, often with processing methods that are time-consuming or computationally expensive, with varying accuracy based on tissue structure. Here, we propose a spectral Doppler envelope estimation method designed for skeletal muscle measurements. When compared to the modified signal noise slope intersection (MSNSI) method, our method reduces the overall mean absolute error by 13.9% and the mean absolute zero error by 82.1%. We validated our method using a portable ultrasound system on a benchtop setup that mimics the acoustic properties, measurement angles, and velocity patterns of skeletal muscles. Ex vivo and *in vivo* muscle velocity estimates of parallel and pennate muscles will be compared to those obtained using the MSNSI method and manual tracking of B-Mode images. Our proposed method could enable automated estimation of skeletal muscle velocities during dynamic activities in unconstrained environments, providing new insight into neuromuscular function and movement biomechanics, with potential applications in monitoring fatigue, disease progression, or injury recovery. [Work supported by the National Science Foundation.]

2:30–2:45 Break

2:45

1pBAa7. Generalization of deep learning models for hepatic steatosis grading using B-mode ultrasound images. Pedro Vianna (Lab. of Biorheology and Medical Ultrason., Univ. of Montreal Hospital, 900, St.-Denis, Ste. R11.720, Montréal, QC H2X 0A9, Canada, pedro.vianna@umontreal.ca), Yue Qi (Res. Ctr., Univ. of Montreal Hospital, Montreal, QC, Canada), Michael Chassé (Univ. of Montreal Hospital, Montreal, QC, Canada), Guy Wolf (Mila - QC AI Inst., Montreal, QC, Canada), Eugene Belilovsky (Mila - QC AI Inst., Montreal, QC, Canada), An Tang (Univ. of Montreal Hospital, Montreal, QC, Canada), and Guy Cloutier (Lab. of Biorheology and Medical Ultrason., Univ. of Montreal Hospital, Montréal, QC, Canada)

Grayscale ultrasound remains a key modality for screening of hepatic steatosis due to its non-invasiveness and availability. While neural networks have shown promise in this field, their main drawback lies in their inability to generalize to diverse real-world settings. Variations in equipment, acquisition parameters, or population significantly affect model performance. Test-time adaptation, an unsupervised domain adaptation technique, overcomes these limitations by adjusting trained models during inference. Our retrospective study used two datasets collected in separate populations, with different scanners and protocols. We propose an adaptation method, using test-time batch normalization to selectively adjust BatchNorm layers based on test data for predicting steatosis grades. Comparing the non-adapted and adapted models, the mean absolute error ($\pm$ standard deviation) in grading four severities of steatosis decreased from 0.92 ± 0.21 to 0.64 ± 0.22 . Specifically, for detection of steatosis the area under the curve increased from 0.76 ± 0.05 to 0.95 ± 0.02 when using the adapted model. Adapted models show promising results in improving performance compared to base models when testing data differ significantly from training data. Results suggest that the proposed method effectively addresses domain shift in diagnosing fatty liver using ultrasound images, reducing risks associated with deploying trained models.

3:00

1pBAa8. Reconstructing the source condition for focused shear wave beams in soft elastic media. Branch T. Archer (Chandra Family Dept. of Elec. and Comput. Eng., Univ. of Texas at Austin, 4806 Park Ln., Austin, TX 78732, btaiii@yahoo.com), Yu-Hsuan Chao (Dept. of Bioengineering, Univ. of Pittsburgh, Pittsburgh, PA), John M. Cormack (Dept. of Medicine, Univ. of Pittsburgh, Pittsburgh, PA), Kang Kim (Dept. of Bioengineering, Univ. of Pittsburgh, Pittsburgh, PA), Kyle S. Spratt, and Mark F. Hamilton (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Shear waves are employed in medical ultrasound imaging because they reveal variations in viscoelastic properties of soft tissue. Focused shear waves produced by a longitudinally vibrating piston at several hundred hertz are investigated here experimentally in soft tissue phantoms and using an analytical model for shear wave beam generation and propagation. Experiments employed a spherically concave piston shaped to focus the shear wave beam at a depth of 4 cm in a soft tissue phantom. Analytical modeling of this data provided a good fit, but required treating the focal length and piston diameter as adjustable parameters. The largest source of error in the modeling is suspected to be that the source condition is assumed to exhibit zero displacement surrounding the piston, whereas the actual condition in the source plane is a traction-free surface surrounding the piston. Here, in order to elucidate the actual source condition, numerical back propagation of field measurements is employed. Simulations show that the shear source condition can be accurately recovered with information related only to the shear wave, i.e., without compressional wave information. Back propagation of measured beams is discussed in the context of more accurately modeling the source condition for future device optimization.

3:15

1pBAa9. Focused shear wave beam propagation through a 3D printed human rib cage. Yu-Hsuan Chao (Dept. of Bioengineering, Univ. of Pittsburgh, Pittsburgh, PA, YUC125@pitt.edu), John M. Cormack (Dept. of Medicine, Univ. of Pittsburgh, Pittsburgh, PA), Branch T. Archer (Dept. of Elec. and Comput. Eng., The Univ. of Texas at Austin, Austin, TX), Kyle S. Spratt, Mark F. Hamilton (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Kang Kim (Dept. of Bioengineering, Univ. of Pittsburgh, Pittsburgh, PA)

Transient elastography (TE) is used in clinical screenings for liver fibrosis. TE generates shear wave motion in the liver by vibration of a piston at the skin. The small, flat piston in current devices radiates shear waves that diverge from the liver, resulting in low SNR and high failure rates, especially in obese patients. Our group recently introduced focused shear wave beams for TE [Cormack *et al*., IEEE TBME (2024)], in which vibration of a concave piston generates shear waves that converge towards the focal region, thereby increasing SNR for liver stiffness estimation. However, because piston sizes needed for efficient shear wave focusing are larger than the typical intercostal space, the rib cage that lies between the skin and the liver may cause shear wave aberration and influence stiffness measurement. Here, we present measurements of broadband focused shear wave propagation in tissue-mimicking gel in which are embedded 3D printed human ribs. Both straight elliptical rods and anatomically realistic 3D printed ribs are used as aberrators. Measurements are compared to 3D simulations of shear wave beam propagation [Archer *et al*., JASA (2023)]. Effects of shear wave aberration by the rib cage depend on shear wavelength, piston-to-rib distance, and piston radius and curvature.

3:30

1pBAa10. Single and multiple scattering quantitative ultrasound methods in muscle. Haley Geithner (North Carolina State Univ., 361 The Greens Circle, Apt. 221, Raleigh, NC 27606, hgeithn@ncsu.edu), Marie Muller, and Katherine Saul (MAE, North Carolina State Univ., Raleigh, NC)

Increased intramuscular fat—or fatty infiltration (FI)—often develops as a consequence of various musculoskeletal and neuromuscular disorders. FI is an important factor for the assessment and intervention of surgical candidacy. Therefore, there is a need for an accessible, non-invasive quantitative, and objective evaluation of FI in muscle. We can exploit the complexity of ultrasound wave propagation in heterogeneous media to extract quantitative features of the fat distribution in muscle. From finite-difference time-domain simulations conducted with maps created from magnetic resonance imaging scans of healthy and injured shoulders, we acquire radiofrequency (RF) ultrasound data. Single and multiple scattering (SS and MS) components of the RF data are separated by using singular value decomposition and eigenvalue thresholding. Spectral and envelope quantitative ultrasound (QUS) parameters are computed from the SS component. MS-based QUS parameters are obtained by extracting the diffusion constant and tracking the SS intensity decay rate. We investigate the relationship between SS and MS-based QUS parameters and an increasing FI, because more FI will lead to more multiple scattering and modified ultrasonic signatures.

3:45

1pBAa11. Nonlinear imaging-based bubble cloud size estimates compared with histotripsy treatment zone: An *in vitro* study. Vishwas Trivedi (Dept. of Elec. Eng., Indian Inst. of Technol. Gandhinagar, MUSE Lab, Indian Inst. of Technol. Gandhinagar, Gandhinagar 382355, India, vishwas.t@iitgn.ac.in), Kenneth B. Bader (Univ. of Chicago, Chicago, IL), and Himanshu Shekhar (Dept. of Elec. Eng., Indian Inst. of Technol. Gandhinagar, Gandhinagar, Gujarat, India)

Volterra filtering combined with subharmonic imaging was previously reported to enhance the visualization of the histotripsy bubble cloud. In this study, we compared the bubble cloud size obtained by imaging to the histotripsy ablation region using receiver operating characteristic (ROC) analysis. Histotripsy was performed in a red blood cell (RBC)-doped agarose phantom using a 1 MHz transducer. Ultrasound imaging data was acquired using a C5-2V probe and chirp-coded excitation (bandwidth: 2–5 MHz). These ultrasound acquisitions were then processed by subharmonic (SH) matched filtering alone, SH with quadratic Volterra filtering, and SH with

cubic Volterra filtering. The therapy pulse and imaging sequence were interleaved (N = 100 frames), and a video camera was used to visualize the damaged region. A mean bubble cloud image was generated across all treatment sequences and co-registered with the camera image. A ROC curve was formed using binary classification of the mean bubble cloud versus the ablation zone, and the area under the ROC curve was determined. Filtering by quadratic and cubic Volterra filters reduced artifacts significantly [over 20 and 30 dB contrast-to-tissue enhancement ($p < 0.01$), respectively], along with achieving a high area under the ROC curve (0.97). These findings highlight the potential of Volterra filtering for histotripsy guidance.

4:00

1pBAa12. Analysis of gas evolution in the heart, liver, and kidney of turtles presenting with gas embolic pathology based on ultrasonography. Katherine M. Eltz (Biomedical Eng., The Univ. of North Carolina at Chapel Hill, 116 Manning Dr., Chapel Hill, NC 27599, kathmary@live.unc.edu), Jose-Luis Crespo (Res., Fundación Oceanogràfic de la Comunitat Valencia, Valencia, Spain), Arian Azarang (Biomedical Eng., The Univ. of North Carolina at Chapel Hill, Chapel Hill, NC), Emma González (Res., Fundación Oceanogràfic de la Comunitat Valencia, Valencia, Spain), Daniel Garcia (Res., Fundación Oceanogràfic de la Comunitat Valencia, Valencia, Spain), Andreas Fahlman (Kolmården Wildlife Park, Valencia, Spain), and Virginie Papadopoulou (Biomedical Eng., The Univ. of North Carolina at Chapel Hill, Chapel Hill, NC)

Human-caused disturbances of sea turtles can result in them presenting with gas embolic pathology which often leads to severe injury or death. While gas embolism has been previously observed in turtles using MRI and x-ray/CT, as well as ultrasound to a lesser degree, how the distribution of gas evolves in different organs over time, and its possible correlation to outcome, is poorly understood. We hypothesize that ultrasound imaging of the heart, kidney, and liver over time can help differentiate pathology resolution or worsening trajectory and may help refine veterinarians' treatment algorithm in this population. The liver, kidney, and heart of 100 by-caught turtles were imaged, and gas amount in each ultrasound scan was graded on a scale from 0 (no gas) to 5 (gas completely shadowing organ anatomy). Turtles scanned on the boat had higher grades in all organs compared to turtles first scanned at shore which was on average 163 minutes later. Average pixel brightness in the top half of cardiac scans increased with grade as expected, apart from grade 5 likely due shadowing. Ultrasound brightness could become a quantitative metric for veterinarians to determine which turtles need hyperbaric oxygen treatment and which can be released.

MONDAY AFTERNOON, 13 MAY 2024 ROOM 212, 1:00 P.M. TO 4:10 P.M.

Session 1pBAb

Biomedical Acoustics, Physical Acoustics, Signal Processing in Acoustics, and Engineering Acoustics: Ultrasound Brain and Super-Resolution Imaging II

Chengzhi Shi, Cochair
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Fabian Kiessling, Cochair
Exp. Mol. Imaging, RWTH Aachen Univ., Forckenbeckstrasse 55, n.a., Aachen 52074, Germany

Invited Papers

1:00

1pBAb1. Deep-brain imaging with 3D integrated photoacoustic tomography and ultrasound localization microscopy. Junjie Yao (Duke Univ., 100 Sci. Dr., Hudson Hall Annex 261, Durham, NC 27708, junjie.yao@duke.edu)

Photoacoustic computed tomography (PACT) is a proven technology for imaging hemodynamics in deep brain of small animal models. PACT is inherently compatible with ultrasound (US) imaging, providing complementary contrast mechanisms. While PACT can quantify the brain's oxygen saturation of hemoglobin (sO_2), US imaging can probe the blood flow based on the Doppler effect. Furthermore, by tracking gas-filled microbubbles, ultrasound localization microscopy (ULM) can map the blood flow velocity with sub-diffraction spatial resolution. In this work, we present a 3D deep-brain imaging system that seamlessly integrates PACT and ULM into a single device, 3D-PAULM. Using a low ultrasound frequency of 4 MHz, 3D-PAULM is capable of imaging the whole-brain hemodynamic functions with intact scalp and skull in a totally non-invasive manner. Using 3D-PAULM, we studied the mouse brain functions with ischemic stroke. Multi-spectral PACT, US B-mode imaging, microbubble-enhanced power Doppler (PD), and ULM were performed on

the same mouse brain with intrinsic image co-registration. From the multi-modality measurements, we future quantified blood perfusion, sO_2, vessel density, and flow velocity of the mouse brain, showing stroke-induced ischemia, hypoxia, and reduced blood flow. We expect that 3D-PAULM can find broad applications in studying deep brain functions on small animal models.

1:20

1pBAb2. Parametric study of blind-label acoustic subwavelength imaging. Jinuan Lin (Elec. and Comput. Eng., Univ. of Wisconsin-Madison, Madison, WI) and Chu Ma (Elec. and Comput. Eng., Univ. of Wisconsin-Madison, 1415 Eng. Dr., Rm. 3436, Madison, WI 53706, chu.ma@wisc.edu)

There is a long-existing tradeoff between the imaging resolution and the penetration depth in imaging systems caused by the diffraction limit. We developed a "blind label" approach to tackle this problem, which significantly improves the practicality of acoustic subwavelength imaging in biomedical ultrasound imaging, non-destructive testing, and other acoustic sensing and communication applications. The "blind labels" in our system refer to randomly distributed acoustic scatterers with deep-subwavelength sizes whose exact locations and trajectories are not necessary information in image reconstruction. Our imaging framework 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 and (2) computational reconstruction. In this talk, we will mainly report our quantitative investigation of the system parameters' impact on the performance of the blind-label subwavelength imaging system, providing guidance to future system setups in various applications.

1:40

1pBAb3. Wearable ultrasound technology. Sheng Xu (Nanoengineering, UC San Diego, 9500 Gilman Dr. Mail Code 0448, SME Bldg., Rm. 343J, La Jolla, CA 92093, shengxu@ucsd.edu)

The use of wearable electronic devices that can acquire vital signs from the human body noninvasively and continuously is a significant trend for healthcare. The combination of materials design and advanced microfabrication techniques enables the integration of various components and devices onto a wearable platform, resulting in functional systems with minimal limitations on the human body. Physiological signals from deep tissues are particularly valuable as they have a stronger and faster correlation with the internal events within the body compared to signals obtained from the surface of the skin. In this presentation, I will demonstrate a soft ultrasonic technology that can noninvasively and continuously acquire dynamic information about deep tissues and central organs. I will also showcase examples of this technology's use in recording blood pressure and flow waveforms in central vessels, monitoring cardiac chamber activities, and measuring core body temperatures. The soft ultrasonic technology presented represents a platform with vast potential for applications in consumer electronics, defense medicine, and clinical practices.

2:00

1pBAb4. Reconstruction methods for super-resolution imaging with PSF modulation. Jian-yu Lu (Bioengineering, The Univ. of Toledo, 2801 West Bancroft St., Toledo, OH 43606, jian-yu.lu@ieee.org)

Recently, a super-resolution imaging method called the PSF (point spread function) modulation method was developed (Lu, IEEE TUFFC 2024). In this method, the amplitude, phase, or both of the PSF of a linear shift-invariant (LSI) imaging system is modulated so that the modulated PSF has a higher spatial frequency than that of the original PSF to reconstruct super-resolution images. The modulator can be produced and manipulated remotely by methods such as radiation force or it can be a physical particle such as micro- or nanoparticle manipulated by an external force such as electrical and electromagnetic force. In principle, the super-resolution imaging method can be applied to any LSI imaging system, such as ultrasound, optical, photoacoustic, electromagnetic, underwater, nondestructive evaluation (NDE), and magnetic resonance imaging (MRI) system. These include pulse-echo ultrasound imaging, transmission imaging, wave source/field imaging, acoustical camera, and optical bright-field microscope. To optimize the quality of the images, methods for the reconstruction of pulse-echo and wave source/field super-resolution images are studied and the results will be presented. These methods include the uses of an analytic envelope of radio-frequency (RF) signals with and without windowing, "I" (in-phase) and "Q" (quadrature) signals, and the DC (direct current) component removal.

2:20–2:35 Break

Contributed Papers

2:35

1pBAb5. Improving photoacoustic imaging through the skull using deep learning: a numerical study. Matthew J. Olmstead (Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, mjolmstead@mac.com), Yu-tong Wang (Acoust., Penn State Univ., University Park, PA), Zixuan Tian (Elec. and Comput. Eng., Univ. of Illinois, Urbana Champaign, Urbana, IL), Hyungjoo Park (Acoust., Penn State Univ., University Park, PA), Aiguo Han (Biomedical Eng. and Mech., Virginia Polytechnic Inst. and State Univ., Blacksburg, VA), and Yun Jing (Acoust., Penn State Univ., State College, PA)

Photoacoustic computed tomography (PACT) has recently emerged as an attractive imaging modality for functional brain imaging due to its rich optical absorption contrast, high spatial and temporal resolutions, and relatively deep penetration. However, a major hurdle in using PACT for the human brain is distortion of the signal due to the skull, which negatively affects the quality of the images. In this project, we aimed to improve transcranial PACT using a U-Net architecture that can minimize distortion from the skull. This numerical study utilized a large collection of blood vessel images obtained from an online database and a computed tomography (CT) scan of an *ex vivo* human skull. The synthetic photoacoustic radiofrequency data were generated using the open-source wave solver *k*-Wave. Comparing the images generated by deep learning with the ground truth images, we achieved an average structural similarity index of 0.874 and an average peak signal-to-noise ratio of 17.92 dB.

2:50

1pBAb6. Plane wave approaches with dual-frequency arrays for superharmonic contrast imaging. Jing Yang (Medical Biophys., Univ. of Toronto, 2075 Bayview Ave., S640, Toronto, ON M4N 3M5, Canada, jingj.yang@mail.utoronto.ca), Emmanuel Cherin, Jianhua Yin (Physical Sci. Platform, Sunnybrook Res. Inst., Toronto, ON, Canada), Paul A. Dayton (Biomedical Eng., UNC Chapel Hill, Chapel Hill, NC), F. Stuart Foster, and Christine Demore (Medical Biophys., Univ. of Toronto, Toronto, ON, Canada)

Superharmonic imaging (SpHI) using dual-frequency probes enables high-contrast microvasculature imaging by taking advantage of higher order harmonics of the broadband nonlinear response from microbubble (MB) contrast agents. We previously introduced a DF probe with a low-frequency (LF, 2 MHz; 32 elements) array behind a high-frequency (HF, 21 MHz; 256 elements) array and demonstrated SpHI with conventional walking-aperture approaches which limit acquisition rates. In this work, ultrafast imaging is investigated to overcome this challenge. We demonstrate SpHI with plane waves and coherent compounding *in vitro* and *in vivo* while evaluating acquisition frame rates. LF plane waves were implemented on VevoF2 systems (FUJIFILM Visualsonics, Toronto) with beam steering enabled by element-specific delays (9 angles between $\pm 10°$, step: 2.5°). All SpHI images showed almost complete suppression of tissue clutter to the background noise level. A 2.5 dB contrast improvement was found *in vitro* with coherent compounding. Tumor perfusion and fine vascular structures were visualized *in vivo*. SpHI acquisition frame rate reached 3.5 kHz at 0° and 396 Hz with 9 angles, $\sim$40 times that of walking-aperture approaches. These results demonstrate plane wave imaging approaches can increase SpHI frame rates while maintaining a high image contrast for visualizing vasculature, enabling SpHI for fast flow imaging.

3:05

1pBAb7. Contrast-free microvessel imaging using null subtraction imaging combined with harmonic imaging. Zhengchang Kou, Rita Miller (Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, Urbana, IL), and Michael L. Oelze (Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, 405 N Mathews, Urbana, IL 61801, oelze@illinois.edu)

Ultrasound localization microscopy (ULM) has superior spatial resolution; however, it requires contrast agents and long data acquisition and processing time. Null subtraction imaging (NSI) is a nonlinear beamforming technique that improves the spatial resolution and reduces grating lobes with low computational cost. We combined pulse-inversion (PI) nonlinear imaging with NSI to increase resolution. Pulses with center frequency of 10.42 MHz and its inverted version were transmitted to obtain the second harmonic at 20.84 MHz (wavelength of $\sim$74 μm). The transducer pitch was 35% larger than the wavelength of the receive signal, which would introduce grating lobes in the field of view. A total of 1000, 9-angle (-8 to 8 degree in 2 degrees step) coherently compounded frames were acquired in 1 s. SVD filters were applied in raw RF data to filter out tissue signal. The dc offset was set to 0.1 in NSI imaging. Grating lobes, which were obvious in DAS images, were removed in the NSI images. Higher spatial resolution and contrast were observed from the NSI microvessel images. The spatial resolution of the NSI images using harmonic imaging approached one fourth of a wavelength with computation time increased by only 40% compared to DAS.

3:20

1pBAb8. Ultrasound imaging of mammalian cells using drug-induced acoustic reporter genes. John Kim (Mech. Eng., Univ. of Michigan, 3033 Lenox Rd. NE, # 23311, Atlanta, GA 30324, jkim3286@gatech.edu), Phoebe J. Welch (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA), Alessandro R. Howells, Xiaojun L. Lian (Biomedical Eng., The Penn State Univ., State College, PA), and Chengzhi Shi (Univ. of Michigan, Atlanta, GA)

The emerging area of genetically engineered ultrasound contrast agents, like gas vesicles, has the potential to broaden the applications of medical ultrasound imaging by enabling targeted and deep tissue imaging at the cellular level. Nevertheless, the existing gene construct/encoding relies on significant cell processing to ensure sufficient gas vesicles are formed within the cell to produce ultrasound contrast. Here, we describe a drug-inducible and drug-selectable acoustic reporter gene construct that can enable gas vesicle expression in mammalian cell lines, which we demonstrate in wild type HEK293T cells. Fluorescence microscopy was employed to validate the integration of the plasmid, while the creation of single-cell clones was achieved through the utilization of flow cytometry. The expression for gas vesicle was optically and ultrasonically verified, achieving 80% improved signal to noise ratio in cells expressing gas vesicles compared to negative controls. This technology introduces a novel paradigm for reporter genes, utilizing ultrasound to visually identify particular cell types both *in vitro* and *in vivo*, serving diverse purposes such as cellular reporting and applications in cell therapies.

3:35

1pBAb9. Active non-Hermitian complementary acoustic metamaterials for transcranial imaging. Yan Deng (Mech. Eng., The Univ. of Michigan, 911 North Univ. Ave., First Fl., Michigan League, Ann Arbor, MI 40189, dengyan@umich.edu) and Chengzhi Shi (Univ. of Michigan, Atlanta, GA)

High-frequency ultrasound has long been a safe and effective tool for medical imaging, diagnosis, and noninvasive treatments. However, the presence of the porous skull poses a challenge, impeding wave transmission and limiting noninvasive ultrasonic brain imaging due to a significant impedance mismatch and energy attenuation. In this study, we propose an innovative approach by introducing an active non-Hermitian complementary metamaterial (NHCMM) to enhance transcranial imaging. The NHCMM integrates piezoelectric elements, hydrogel materials, and a feedback control circuit. This actively modulates the effective acoustic properties of the metamaterial, ensuring precise tuning to match the impedance of the skull, thereby reducing energy loss and enhancing transmission. This design not only creates a transparent window for high-frequency ultrasound but also significantly improves brain imaging quality. The presented work establishes a foundational framework for non-invasive ultrasound-based brain imaging and high-precision ultrasonic therapies, preserving the structural integrity of the cranial barrier. We anticipate that this research holds immense potential for advancing the fields of brain imaging and ultrasonic treatments, paving the way for future innovations in medical diagnostics and therapeutic interventions.

3:50–4:10
Panel Discussion

Session 1pCA

Computational Acoustics, Physical Acoustics, and Underwater Acoustics: Diffusion Equation and Energy Flux Methods Across Acoustics

Jennifer Cooper, Cochair
Appl. Phys. Lab., Johns Hopkins Univ., 11100 Johns Hopkins Rd., Mailstop 8-220, Laurel, MD 20723

Michelle E. Swearingen, Cochair
Constr. Eng. Res. Lab., U.S. Army ERDC, P.O. Box 9005, Champaign, IL 61826

Invited Papers

1:00

1pCA1. Diffusion equation modeling in room-acoustic applications. Ning Xiang (Grad. Program in Architectural Acoust., Rensselaer Polytechnic Inst., 110 Eighth St., Troy, NY 12180, xiangn@rpi.edu) and Zühre Sü Gül (Bilkent Univ., Ankara, Turkey)

The diffusion equation application in room acoustic simulations can be traced back to 1969. However, much attention from room-acoustic community to the diffusion equation was given in the late 1990s and the early 2000s. In the literature, the acoustic diffusion equation may be considered as an approximation of the acoustic transport equation. The diffusion equation modeling has been considered suitable for applications in enclosed spaces with proportionate dimensions. The most intriguing features of the diffusion equation are its computational efficiency due to a sparse meshing condition and its capability of energy flux solutions due to Fick's law. Since the late 2000s, the diffusion equation modeling has evolved with boundary conditions formulated by the rigorous physical–mathematical theory [Jing and Xiang, JASA **123**, 145–153 (2008)] and its validity has also been systematically investigated [Xiang *et al.*, JASA **133**, 3975–3985 (2013)]. This paper reviews the vibrant development in the room-acoustics during this era and discusses wide range of applications from coupled-volume systems, historically significant worship spaces to recent investigations on sound absorption measurements in reverberation chambers. This paper also discusses its link to the acoustic transport equation and compares the modeling results of both. Different numerical approaches to solving the diffusion equations will also be reviewed.

1:20

1pCA2. Directional wave spectrum evolution modelling of ship noise. Michael G. Brown (Ocean Sci., Rosenstiel School, Univ. of Miami, 4600 Rickenbacker Cswy, Miami, FL 33149, mbrown@rsmas.miami.edu)

A new underwater acoustic propagation model that is well suited to modelling ship noise is described. The model describes the spatio-temporal evolution of the directional energy spectrum of a sound field. The governing equation can be thought of as an advection-diffusion equation for spectral energy in phase space (position, angle). Numerical simulations of sound fields excited by both isotropic compact sources and by highly anisotropic ship noise provide confidence that the model works well, with caveats that will be discussed. [Work supported by ONR.]

Contributed Paper

1:40

1pCA3. A novel computational modeling of blood capillary network effects on ultrasound-induced thermal therapy in tumors. Farshad Moradi Kashkooli (Toronto Metropolitan Univ., 350 Victoria St., Toronto, ON M5B 2K3, Canada, fmoradik@torontomu.ca), Graham Ferrier, Anshuman Jakhmola, Jahangir (Jahan) Tavakkoli, and Michael Kolios (Toronto Metropolitan Univ., Toronto, ON, Canada)

A novel computational model is developed to calculate how blood capillary networks impact the ultrasound-induced thermal therapy of tumors. In the existing Pennes' bioheat transfer (BHT) model, a blood perfusion term oversimplifies a potentially wide variety of temperature distribution influenced by the complex structure of the blood vascular network and blood flow within vasculature. Therefore, rather than use a blood perfusion term in a non-vascularized tumor domain (in BHT model), we solve generalized heat transfer (GHT) equations in a vascularized tumor domain. Temperature distribution is simulated by concurrently solving the Helmholtz equation for acoustic wave propagation, the convective blood flow in microvessels, and the GHT equations in both microvessels and tissue domains using finite element analysis. Ultrasound-induced heat maps are generated using blood perfusion values (in BHT) extracted from preclinical data in the literature as well as the equivalent blood flow rates (in GHT). While the spatial average and maximum temperatures agree closely in both models, differences in the BHT and GHT heat maps are observed. The BHT heat map is symmetric, homogeneous, and has a temperature maximum at the ultrasound's focal point, whereas GHT heat map is non-symmetric, heterogeneous, and has spatially shifted temperature maxima. These differences are attributed to cooling effects induced by convective blood flow in capillaries, capillary network structure, and blood flow direction in the tumor.

Session 1pEA

Engineering Acoustics, Physical Acoustics, and Structural Acoustics and Vibration: Applications of Acoustic Metamaterials

Farhad Farzbod, Cochair
Mech. Eng., Univ. of Mississippi, 1764 Univ. Circle, Rm. 203, University, MS 38677

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

Chair's Introduction—1:00

Invited Papers

1:05

1pEA1. Programmable topological insulators based on a reconfigurable electroacoustic material platform. Michael Leamy (Georgia Inst. of Technol., 771 Ferst Dr. N.W., Atlanta, GA 30332-0405, michael.leamy@me.gatech.edu)

Topological insulators (TIs), exhibiting topologically protected edge and interface waves, have recently emerged in phononic systems. Reconfigurabilty is essential for enabling TI-based applications. One potential means for achieving reconfigurability employs shunted piezoelectric (PZT) disks in which a unit cell's mechanical impedance is altered using negative capacitance circuits. Dynamic reconfigurability and programmability of such material platforms can then be obtained through simple on/off switching. In this vein, we propose and experimentally verify an electroacoustic TI which exhibits programmable topologically protected edge states useful for acoustic multiplexers, demultiplexers, and transistors. This reconfigurable structure is composed of an elastic hexagonal lattice whose unit cell contains two shunted PZT disks, each connected to a negative capacitance circuit by an on/off switch. Closing one or the other circuit results in the breaking of mirror symmetry and yields mechanical behavior analogous to the quantum valley Hall effect. By interfacing two topologically distinct materials, a domain wall is introduced exhibiting a localized interface state topologically protected from backscattering at defects and sharp edges. Through the use of programmable time-division, in which domain walls appear and disappear in time, we demonstrate multiplexing and demultiplexing. We also demonstrate an acoustic transistor using the same programmable platform, before closing with a discussion on future research directions.

1:25

1pEA2. Application of acoustic metamaterials to phase computing. Pierre A. Deymier (Mater. Sci. and Eng., New Frontiers of Sound (NewFoS) Ctr., Univ. of Arizona, 1235 E. James E. Rogers Way, Univ. of Arizona, Tucson, AZ 85721, deymier@arizona.edu), Keith Runge (New Frontiers of Sound (NewFoS) Ctr., Univ. of Arizona, Tucson, AZ), M Arif Hasan (Mech. Engineering/NewFoS, Wayne State Univ., Detroit, MI), Joshua A. Levine (Comput. Science/NewFoS, Univ. of Arizona, Tucson, AZ), and Michael Leamy (Mech. Engineering/NewFoS, Georgia Inst. of Technol., Atlanta, GA)

We review the notion of "phase bit" or "phi-bit" in externally driven nonlinear acoustic metamaterials. Phi-bits are classical analogues of quantum bits, which open pathways to promising and validated modes of initializing, operating, and measuring information. Acoustic metamaterials offer ways to compute information using phase that should compare favorably with state-of-the-art quantum systems without suffering from quantum fragility.

1:45

1pEA3. Tollmien–Schlichting wave manipulation by a multi-input multi-output phononic subsurface. Carson Willey (UES Inc./Air Force Res. Lab., Wright-Patterson AFB, OH), Caleb Barnes (AFRL, Wright Patterson AFB, OH), Vincent Chen (UES Inc./Air Force Res. Lab., Wright Patterson AFB, OH), Kevin Rosenberg (Spectral Energies, LLC, Wright Patterson AFB, OH), Alberto Medina (AFRL, Wright Patterson AFB, OH), and Abigail T. Juhl (AFRL, Wright Patterson Air Force Base, OH, abigail.juhl.1@us.af.mil)

Recently, there has been a reinvigorated effort to identify passive control mechanisms to reduce drag on vehicles moving through a fluid medium. In 2015, Hussein *et al*. engineered a phononic crystal subsurface (PSub) to interact with Tollmien–Schlichting waves of a channel flow through an interaction surface, which caused a localized reduction in the kinetic energy of the flow. Since then, arrayed PSubs, Helmholtz resonators, and other embedded structures have been studied for their ability to produce similar effects. In this talk, an implementation of a multi-input multi-output (MIMO) PSub is presented and its ability to delay the onset of turbulence is shown to depend on simultaneously satisfying single-input single-output (SISO) PSub phase requirements at both interaction surfaces of the MIMO PSub. The MIMO architecture provides a method to place an arbitrary phasing between the T-S wave induced forcing and the displacement response of the interaction surface and can be optimized for the flow conditions of the environment.

2:05

1pEA4. Meta-earplugs: Innovative concepts for alleviating the occlusion effect. Kévin Carillo (Mech. Eng., ETS (Ecole de technologie superieure), 1100 Rue Notre-Dame Ouest, Montréal, QC H3C 1K3, Canada, kevin.carillo@etsmtl.ca), Hugo Saint-Gaudens (Mech. Eng., ETS (Ecole de technologie superieure), Montreal, QC, Canada), Franck Sgard (IRSST, Montreal, QC, Canada), Olivier Dazel (Le Mans Université, Le Mans, France), and Olivier Doutres (Mech. Eng., ETS (Ecole de technologie superieure), Montreal, QC, Canada)

Passive earplugs are commonly used to reduce workers' exposure to excessively high noise levels. Yet, they are associated with various discomforts of diverse origins (e.g., acoustic, physical, functional, or psychological). The occlusion effect, characterized by an increased perception of physiological sounds transmitted through bone conduction to the cochlea, presents a challenge, leading to acoustic discomfort, especially at low frequencies (0.1 to 1 kHz) and for shallow or moderate insertion depths of the earplug. Building upon acoustic meta-material principles, this study presents "meta-earplug" concepts designed to alleviate the occlusion effect. The approach focuses on reducing the input impedance of the earplug medial surface either to the characteristic impedance of air, using broadband perfect absorption, or to the acoustic impedance of the open ear canal, resulting in a zero objective occlusion effect. For these purposes, the proposed meta-earplug concepts are made of Helmholtz resonators arranged in parallel or series. Transfer matrix models are used in an optimization process to refine the geometry of the meta-earplugs. Although meta-earplug concepts involving multiple Helmholtz resonators have been preliminarily assessed using an artificial ear, a meta-earplug featuring a single resonator has undergone testing with human participants. The evaluation encompasses both objective measurements and subjective assessments.

2:25–2:40 Break

Contributed Papers

2:40

1pEA5. Sound attenuation performance at high sound pressure level of micro-perforated panel sound absorber with embedded resistive screen. Zacharie Laly (Dept. of Mech. Eng., Sherbrooke Univ., 2500, boul. de l'Université, Sherbrooke, QC J1K 2R1, Canada, zacharie.laly@usherbrooke.ca) and Noureddine Atalla (Dept. of Mech. Eng., Sherbrooke Univ., Sherbrooke, QC, Canada)

In this paper, a composite sound absorber made of a micro-perforated panel (MPP), an air cavity with embedded resistive screen and a rigid back plate is investigated at high sound pressure level (SPL). The sound absorption coefficient predicted theoretically is compared with the experimental measurement at 150 dB and a good agreement is obtained. It is demonstrated that the incorporation of resistive screen within the air cavity improves significantly the sound absorption that presents a large frequency band and remains almost identical with respect to the SPL compared to a classical MPP absorber whose sound absorption peak varies with the SPL. The MPP absorber with the resistive screen glued on the MPP is studied and different resistive screens are compared at 150 dB and it is observed that the sound absorption coefficient increases with a large frequency band, which depends on the airflow resistivity of the screen. A sensitivity analysis shows that the resistance per unit area of the screen affects the acoustic properties of the absorber at low SPL while at high SPL, the acoustic properties are mainly controlled by the perforation ratio of the MPP and the acoustic orifice Mach number.

2:55

1pEA6. Use of metamaterials to reduce underwater noise generated by ship machinery. Mathis Vulliez (Dépt. de génie mécanique, Univ. de Sherbrooke, Sherbrooke, QC J1K 2R1, Canada, Mathis.Vulliez@USherbrooke.ca), Marc-André Guy (Dépt. de génie mécanique, Univ. de Sherbrooke, Québec, QC, Canada), Kamal Kesour, Jean-Christophe G. Marquis (Innovation Maritime, Rimouski, QC, Canada), Giuseppe Catapane, Giuseppe Petrone (Univ. of Naples Federico II, Naples, Italy), and Olivier Robin (Dépt. de génie mécanique, Univ. de Sherbrooke, Sherbrooke, QC, Canada)

Reducing underwater noise pollution from ship machinery is a significant challenge. Ship machinery usually operates at fixed speeds and emits tonal noise with large amplitudes at low frequencies. Conventional soundproofing materials are inadequate for absorbing tonal noise and require large thicknesses at low frequencies. Quarter-wavelength resonators effectively absorb sound at their fundamental frequency and odd harmonics. Still, the applicability of this solution is nevertheless limited by its length requirement, which becomes cumbersome at low frequencies and, thus, large wavelengths. This study explores different structured metamaterial designs based on labyrinth, coiled quarter-wavelength resonators, and hybrid configurations combining glass wool and coiled resonators. Analytical, numerical calculations, and experimental tests are carried out under normal plane wave incidence (using an impedance tube) and a diffuse acoustic field (in a small reverberant cabin). In particular, a numerical optimization based on a periodic unit cell model is used to optimize the hybrid configuration and analyze its behavior under variable plane wave incidence angles. Preliminary tests conducted in a water basin using a small, straightforward aluminum box equipped with some proposed designs indicate reductions in underwater noise levels. The proposed solutions offer limited-cost and compact solutions for mitigating machinery noise and, potentially, the preservation of marine ecosystems.

3:10

1pEA7. Improvement of the sound absorption performance of sandwich panel using inhomogeneous micro-perforations. Zacharie Laly (Dept. of Mech. Eng., Sherbrooke Univ., 2500, boul. de l'Université, Sherbrooke, QC J1K 2R1, Canada, zacharie.laly@usherbrooke.ca), Chris Mechefske (Dept. of Mech. and Mater. Eng., Queen's Univ., Kingston, ON, Canada), Sebastian Ghinet, Tenon Charly Kone (Aerosp., National Res. Council Canada, Ottawa, ON, Canada), Noureddine Atalla (Dept. of Mech. Eng., Sherbrooke Univ., Sherbrooke, QC, Canada), and Behnam Ashrafi (Aerosp. Manufacturing Technol. Ctr., National Res. Council Canada, Montreal, QC, Canada)

Sandwich structures are widely used in several fields because of their mechanical performance with low weight. In this paper, a sandwich panel made of a core and a top and a back panel is investigated using the finite element method. The core consists of 33 hexagonal cells and each cell is connected to a micro-perforation. To improve the sound absorption, inhomogeneous micro-perforations are considered within the top panel. It is shown that with homogenous micro-perforations made of the same diameter, the sound absorption coefficient presents only one resonant peak. When 2, 3, and 6 different sets of micro-perforation diameters are considered, the sound absorption coefficient presents, respectively, 2, 3, and 6 resonance peaks where the surface impedance is close to that of the air. When each micro-perforation diameter is different resulting in inhomogeneous distribution of micro-perforations within the top panel, the sound absorption frequency band is widened and the absorption coefficient value increases while without micro-perforations the absorption coefficient of the sandwich panel is zero over the entire frequency range. The studied structure can offer high mechanical stiffness and good sound absorption performance.

3:25

1pEA8. Acoustic metamaterial samples preparation for impedance tube measurements—The influence of different 3D printing techniques and mechanical post-processing on the simulation compliance. Bartlomiej Chojnacki (AGH Univ. of Krakow, Mickiewicza 30, Cracow 30-059, Poland, bchojnacki@agh.edu.pl), Aleksandra Chojak, Jan Pawlik, Wojciech Binek, and Julia Idczak (AGH Univ. of Krakow, Cracow, Poland)

Prototyping of acoustic metamaterials frequently involves the use of additive manufacturing techniques. It offers the best price and the option for complex structure production, which is beneficial in the R&D process. Previously, some differences in acoustic metamaterial sound absorption measurement dispersions regarding using 3D printing techniques for sample preparation were noted. The sound absorption sensitivity on different printing settings or materials is usually omitted. The paper will present the research on the impedance tube measurements of absorbing metamaterial manufactured with different 3D printing methods and post-processed with mechanical and chemical treatment. Two approaches to metamaterial construction were analyzed in the current study—the single-solid metamaterial and the structure divided into parts, where the influence of assembly methods was investigated. To achieve the best compatibility with FEM or TMM modeling results, different post-processing techniques were involved, such as drilling, grinding, and sealant use. The presentation will describe the valuable paths used in sample preparation to achieve better compatibility between the measurements and simulations. The need for a comprehensive part preparation description in metamaterials research and papers will be discussed. [Work funded and supported by the Polish Government, National Centre of Research and Development, agreement no. LIDER/11/0065/L-12/20/NCBR/2021.]

3:40

1pEA9. Design and optimization of meta-material ventilated sound barriers using Helmholtz resonator for building façades: An analytical and numerical investigation. Mohammad Tabatabaei Manesh (Architecture, Univ. of Washington, 3950 Univ. Way NE, Seattle, WA 98105, mhtaba@uw.edu) and Tomás I. Méndez Echenagucia (Architecture, Univ. of Washington, Seattle, WA)

Common building facade materials often struggle to balance between sound absorption and ventilation. As urbanization intensifies and the need for sustainable, occupant-friendly structures rises, there is a growing imperative to develop innovative solutions that address both acoustic and ventilation requirements concurrently. This research addresses this pressing need by proposing a meta-material ventilated sound barrier for building façades. The incorporation of Helmholtz resonators in the design aims to enhance acoustic absorption capabilities while maintaining optimal conditions for natural ventilation. The methodology of this research consists of two stages. First, design and optimize arrays of Helmholtz resonators using analytical models and calculate sound absorption and transmission loss for a broad range of sound frequencies and air flow ventilation. Second, validate the results with FEM analysis in COMSOL Multiphysics. This study anticipates that the meta-material ventilated sound barrier, enriched with Helmholtz resonators, will showcase superior acoustic absorption capabilities while concurrently facilitating optimal airflow for ventilation. The combined analytical and numerical approach is expected to provide comprehensive insights into the performance of the proposed design, laying the foundation for transformative advancements in the acoustic and ventilation aspects of building façade materials.

3:55

1pEA10. Long-range electrostatic forces in periodic structures: The development of adjustable wave filters. Farhad Farzbod (Mech. Eng., Univ. of Mississippi, 1764 Univ. Circle, Rm. 203, University, MS 38677, farzbod@olemiss.edu)

In this work, we examine the effects of electrostatic forces on wave propagation in periodic structures. The research explores how these structures, a type of metamaterial, can have their vibrational characteristics altered by interactions that are non-local. In this work, it is demonstrated how electrostatic forces influence dispersion curves and the resultant metamaterial properties. These findings have implications for developing tunable metamaterials, with potential applications in various fields, such as tunable wave-beaming devices and filters. This research contributes to a deeper understanding of the complex interactions in metamaterials and their practical applications.

Session 1pMU

Musical Acoustics: General Topics in Musical Acoustics I

Andrew A. Piacsek, Chair
Physics, Central Washington Univ., 400 E. University Way, Dept. of Physics, Ellensburg, WA 98926-7422

Contributed Papers

1:30

1pMU1. Statistical correlations between construction parameters and radiated sound in a set of professional-level violins. Sebastian Gonzalez (DEIB, Politecnico di Milano, via Bell'Aspa 3, C/O Lavoratorio Arvedi, Cremona, CR 26100, Italy, tsuresuregusa@gmail.com), Mary Jane Kwan (Sam Zygmuntowicz Violins, NY, NY), Fabio Antonacci (DEIB, Politecnico di Milano, Milano, Italy), and Sam Zygmuntowicz (Sam Zygmuntowicz Violins, NY, NY)

Violin making is perhaps one of the most mysterious crafts in the western tradition of instrument making. The current approach to crafting musical instruments relies heavily on intuition and accumulated knowledge, transmitted through generations in a traditional master-apprentice format. The extended time required for both instrument creation and the development of the skills and intuition essential for this craft makes the learning process highly time-consuming. The fundamental question in violin making is how construction and material parameters influence the acoustic response of the instrument. By analyzing data recorded for over 20 years of career, we are able to obtain strong correlations between material, design and acoustic characteristics for a set of instruments by the same maker. This is, to the best of our knowledge, the most complete dataset in the world. Applying least square fitting to the power emitted by the violins in different bands, we are able, for the first time, to find statistically significant correlations between construction parameters and acoustic response in a set of professional-level instruments.

1:45

1pMU2. Sound designing classical guitars through metamaterials. Rolf Bader (Inst. of Syst. Musicol., Univ. of Hamburg, Neue Rabenstr. 13, Hamburg 20354, Germany, r_bader@t-online.de) and Patrick Kontopidis (Inst. of Syst. Musicol., Univ. of Hamburg, Hamburg, Germany)

An approach is presented to alter the sound of a classical guitar using a metamaterial structure. By adding additional masses using magnets in a row between the bridge and the soundhole, a bandgap of around 300 Hz appears. Alternation of magnet masses and placement determine the location and strength of the bandgap. A similar approach was presented with a frame drum [R. Bader, J. L. Fischer, M. Münster, and P. Kontopidis, "Metamaterials in musical acoustics: A modified frame drum," JASA **145**(5), 3086–3094 (2019)]. There, the lower and upper limits of the bandgap are caused by the relations between the frequency wavelength and the magnet distances. With the guitar, the bandgap is caused by a shifting and attenuation of modes. Interestingly, the added masses cause a decrease in the guitar top plate mode frequencies, which is similar to membrane behavior and contrary to the expectancy of a plate. Such a guitar sound is unique due to the strong bandgap and cannot be achieved with typical alternations of changed fan bracing or top plate thickness. Magnets can easily be applied by guitarists within minutes and, therefore, qualify as a guitar sound design chosen by players.

2:00

1pMU3. The dependence of viola top plate mode frequencies and shape on string tension. Caroline P. Peyton (Whitman College, 345 Boyer Ave., Walla Walla, WA 99362, peytonc@whitman.edu) and Kurt R. Hoffman (Phys., Whitman College, Walla Walla, WA)

The normal mode vibrations of a viola's top plate impact the tone quality of the radiated sound. Luthiers pay some attention to the body mode frequencies during fabrication of the instrument, though specific tuning of modes does not seem to be a priority over other considerations. Interestingly, the normal mode vibrations of the instrument body are modified by the downward force of the bridge feet on the top plate due to the string tension. Different string materials result in different string tensions for proper tuning, so the final body mode frequencies are not strictly predictable. We will present ESPI studies of the top plate normal mode frequencies and mode shapes of a viola as the string tension is increased from zero (strings as bridge removed) to in-tune values for all four strings. The changes of resonant frequency and mode shapes will be discussed in terms of their sensitivity to string tension and the resultant tuning of the final instrument behavior. Knowing the sensitivity of the instrument body to string tension variation may help in understanding the importance of body resonance tuning during the fabrication process.

2:15

1pMU4. Timbral effects of col legno tratto techniques on bowed cello sounds. Montserrat Pàmies-Vilà (Dept. of Music Acoust. - Wiener Klangstil (IWK), Univ. of Music and Performing Arts Vienna, Anton-von-Webern-Platz 1, mdw - Inst. 22, Vienna 1030, Austria, pamies-vila@mdw.ac.at) and Charalampos Saitis (School of Electron. Eng. and Comput. Sci., Queen Mary Univ. of London, London, United Kingdom)

There are several playing techniques for bowed-string instruments that make use of the wooden stick of the bow. The stick is quite often used to strike the strings gently (col legno battuto) and less commonly to bow on them (col legno tratto). Col legno has existed since the 17th century, and it is often used in modern compositions. When the stick is drawn across the string (tratto), the contact between the scrubbing stick and the string introduces noise. The player may choose to combine both hair and stick, depending on the desired sound. To evaluate the timbral effects of col legno tratto on the cello sound, the current study compares three variations across ordinary and contemporary bowing techniques: using only the hair, using both hair and stick, and using only the stick. Motion capture and audio-video recordings with expert cello players show how the bow tilt varies greatly between the three cases. Suitable audio descriptors of timbre are evaluated, which may help to correlate the observed playing parameters and sound properties with the semantic attributes used by experts to describe the timbre of these techniques.

2:30

1pMU5. Experimental assessment of playability limits during bowed string attacks. Alessio Lampis (Dept. of Music Acoust. (IWK), mdw - Univ. of Music and Performing Arts Vienna, Anton-von-Webern-Platz 1, Vienna 1030, Austria, lampis@mdw.ac.at), Alexander Mayer, and Vasileios Chatziioannou (Dept. of Music Acoust. (IWK), mdw - Univ. of Music and Performing Arts Vienna, Vienna, Austria)

The playability limits in the bow force and bow acceleration parameter space (usually visualized as a Guettler diagram) define the conditions for establishing Helmholtz motion during bowed attacks in bowed-string instruments. Despite few theoretical and numerical studies, there is little empirical validation of these limits. The experimental scanning of a Guettler diagram is a tedious process, as it requires the use of a bowing machine to control the bowing parameters. This study proposes a method for collecting and analyzing data to produce measured Guettler diagrams. Experiments were conducted using a cello string in a monochord arrangement with fixed terminations. The string was excited using a robotic arm that allows control of the playing conditions by changing the bow force and bow acceleration. Each set of measurements contains more than 900 data points to obtain high-resolution Guettler diagrams. In addition, the measurements are repeated in reverse order and after dismounting and remounting the string. The results are compared with existing literature, with particular attention to theoretical limits. The experimental results suggest that the static and dynamic friction coefficients vary with bow force and acceleration, modifying the playability limits. [Work supported by the Austrian Science Fund (FWF) (P34852-N).]

2:45–3:00 Break

3:00

1pMU6. Towards reproducing bowed-string instrument transients using physical modeling. Ewa Matusiak (Dept. of Music Acoust. (IWK), mdw - Univ. of Music and Performing Arts Vienna, Anton-von-Webern-Platz 1/II, Wien 1030, Austria, matusiak@mdw.ac.at) and Vasileios Chatziioannou (Dept. of Music Acoust. (IWK), mdw - Univ. of Music and Performing Arts Vienna, Vienna, Austria)

Physical models of bowed-string instruments can forecast various aspects of bow-string interaction. However, accurately replicating transient waveforms of string oscillations remains a challenge. In this work, measured transient behavior of the bowed string is compared with predictions from a simulation model. Fixed boundary conditions are assumed and a finite-width bow, bow-hair compliance, torsional string motion, and an elasto-plastic friction model are incorporated. The latter links relative velocity and friction force through a differential equation, and therefore, exhibits hysteresis behavior. Simulated Guettler diagrams are then compared to measured ones obtained from a robot bowing a monochord. The general shape of the playability region is successfully predicted, but some details in the measured time-domain waveforms are not reproduced by the model. Employing inverse modeling, parameter values for the elasto-plastic friction model are derived, demonstrating the ability to generate reliable reconstructions of the measured transient signals. These results emphasize the need to account for variable friction coefficients in order to reproduce measured signals. This observation aligns with prior findings indicating that static and dynamic friction coefficients vary with bow force and bow acceleration. [Work funded in whole or in part by the Austrian Science Fund (FWF) (P34852-N).]

3:15

1pMU7. A comparison of various steel-string acoustic guitars' modal response with relation to typical playing styles. Mark Rau (Music Res., McGill Univ., 660 Lomita Court, Stanford, CA 94305, mrau@ccrma.stanford.edu) and Gary Scavone (Music Res., McGill Univ., Montreal, QC, Canada)

Steel-string acoustic guitars are built with large variations in geometry and materials, leading to different-sounding instruments. Musicians will have preferences for geometries or woods depending on certain musical styles or personal preferences regarding tonal characteristics. For example, dreadnought-style guitars with either mahogany or rosewood back and sides and a spruce top are overwhelmingly preferred for bluegrass music. This work presents the beginnings of a project to collect measurements from a vast and varied collection of guitars attempting to span the ranges of guitar woods and geometries. Vibration and acoustic measurements of the instruments are captured and modal analysis is performed to extract the modal frequencies, damping ratios, and amplitudes of the prominent modes. The modal characteristics among them are compared to better understand the most prominent differences with an attempt to learn why certain geometries or woods are preferred for specific genres of music and playing styles. The dataset is continuing to grow and currently includes measurements of twenty different guitars.

3:30

1pMU8. Room influence on the acoustical footprint of a violin. Anna Quiros (Tufts Univ., 419 Boston Ave., Medford, MA 02155, anna.quiros@tufts.edu) and Chris Rogers (Tufts Univ., Medford, MA)

Every violin has a unique acoustical footprint characterized by its sound transfer function (TF), and this study measures the influence of a room's acoustical properties on that footprint. Using an omnidirectional microphone (centered, perpendicular, 20 cm from the bridge) and a force measurement hammer, we collect data across six rooms (having 0.5 to 2 s average reverberation times). After vibrationally isolating the violin and muting its strings, we tap the bridge's upper left corner ten times and record 0.3 s audio files of each tap (48 kHz sampling rate). We estimate the TF by normalizing the sound pressure level data by the tap force measurement in frequency space. Analysis of TFs reveal variations due to violin angle. Between 250 and 1.5 k Hz, (frequency range containing the violin's characteristic A0, B1+, B1−, and transition hill modes), the average standard deviation within a 10-tap set on one day is 1 dB, and 2 dB across multiple days. Higher frequencies have more variation. Observations are not clearly tied to room size or reverberation time. The data suggest repeatable measurements of the characteristic violin modes in different acoustical environments are possible, though additional research across multiple days is necessary to confirm reproducibility.

3:45

1pMU9. A comparative study of the emotional characteristics of violin and erhu musical excerpts: Influence of playing techniques and instrument. Wenyi SONG (Dept. of Comput. Sci. and Eng., The Hong Kong Univ. of Sci. and Technol., Clear Water Bay, Kowloon, Hong Kong, wsongak@cse.ust.hk), Ziya ZHOU (Div. of Emerging Interdisciplinary Areas, Acad. of Interdisciplinary Studies, The Hong Kong Univ. of Sci. and Technol., Hong Kong, Hong Kong), Zeyu HUANG (Dept. of Comput. Sci. and Eng., The Hong Kong Univ. of Sci. and Technol., Hong Kong, Hong Kong), and Andrew B. Horner (Dept. of Comput. Sci. and Eng., The Hong Kong Univ. of Sci. and Technol., Hong Kong, Hong Kong)

This study investigates the emotional characteristics of the violin and erhu in multiple musical excerpts. Participants assessed 52 musical excerpts played by both instruments using the Self-Assessment Manikin (SAM) scale. Pairwise comparisons explored the impact of playing techniques (Vibrato, Portamento, and Trill) and instrument on emotional perception. Fifty-eight participants evaluated four versions of 13 musical pieces. Significant agreement was found between the emotion categories and Valence-Arousal (VA) ratings. Ambiguity between calmness and sadness emerged, with VA tending to classify ambiguous excerpts as sad. Playing techniques enhanced the energetic qualities of both instruments. The violin consistently evoked more positive and energetic perceptions compared to the erhu, which were further enhanced when playing techniques such as vibrato were employed. Although erhu with playing techniques tended to be more negative, it still elicited greater energy and positivity than erhu without techniques. Valence-Arousal means were used to determine the emotional qualities of the 13 musical pieces. A subsequent pairwise comparison involving 33 participants revealed that versions with playing techniques had stronger emotional impact than those without techniques. Furthermore, emotional impact was higher when the piece was originally composed for the instrument being played, which was manifest in higher Bradley–Terry–Luce (BTL) values.

Session 1pNSa

Noise, Physical Acoustics, and Engineering Acoustics: Jet & Rocket Noise

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

Contributed Papers

1:00

1pNSa1. Measurements of rocket landing sonic booms from three SpaceX Falcon-9 booster landings. Mark C. Anderson (Dept. of Phys. and Astronomy, Brigham Young Univ., ESC N201, Provo, UT 84602, mark.anderson@byu.net), Kaylee Nyborg, and Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

As commercial rocket companies reduce costs, a common strategy is to design reusable rockets. The recovery typically includes propulsively landing the first stage booster near the launch site or on an ocean platform. Because the booster returns at supersonic speeds, a sonic boom is produced. This paper analyzes measurements made of three SpaceX Falcon-9 booster landings at Vandenberg Space Force Base. Each measurement uses multiple microphones surrounding the launch and landing pads at distances varying between 300–25 000 m. The Falcon-9 sonic boom waveforms have three shocks, and all three are consistently measured at every single measurement location. Although the rise times increase with distance, the duration between the shocks shows a more complicated trend, with the farthest measurements sometimes having the same duration as the nearest measurements. Farther than 1 km, the sonic boom peak overpressure can exceed the peak launch noise overpressures. Farther than a few kilometers, the sound exposure level from the sonic booms can be comparable to the exposure during the entire 12 dB-down period for the launch noise. [Research funded in part by the Oak Ridge Institute for Science and Education and Vandenberg Space Force Base.]

1:15

1pNSa2. Variation of Falcon 9 noise at far-field recording sites. Noah Pulsipher (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, npuls@student.byu.edu), Levi T. Moats, Kent L. Gee, Grant W. Hart (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), Megan R. McCullah-Boozer (CSU Bakersfield, Bakersfield, CA), and Lucas K. Hall (Biology, California State Univ. Bakersfield, Bakersfield, CA)

The rapid launch cadence of SpaceX's Falcon 9 rocket provides the opportunity to study the relative consistency of far-field noise propagation. Acoustical measurements of several Falcon 9 launches have been made on and near Vandenberg Space Force Base at a far-field location 8.4–14 km from the pad. This paper compares collocated measurements from different Falcon 9 missions to begin to understand data variability as a function of launch and environmental conditions at far-field locations. The 8.4 km location has been measured over 10 times, whereas other locations span 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 by USACE.]

1:30

1pNSa3. Variation of Falcon 9 noise in the mid-field. Levi T. Moats (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, lmoats359@gmail.com), Noah Pulsipher, Kent L. Gee, Grant W. Hart (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), Megan R. McCullah-Boozer (CSU Bakersfield, Bakersfield, CA), and Lucas K. Hall (Biology, California State Univ. Bakersfield, Bakersfield, CA)

The rapid launch cadence of SpaceX's Falcon 9 rocket provides the opportunity to study the relative consistency of noise radiation. Acoustical measurements of several Falcon 9 launches have been made on and near Vandenberg Space Force Base within 1 km of the launch pad. This paper compares collocated mid-field measurements from different Falcon 9 missions to begin to understand data variability as a function of launch and environmental conditions. One location 395 m from the pad has been measured over 5 times. 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 by USACE.]

1:45

1pNSa4. Generating Corcos-coherent signal series using phase perturbation. S. Hales Swift (Sandia National Labs., P.O. Box 5800, MS 1082, Albuquerque, NM 87123-1082, shswift@sandia.gov) and Ihab F. El-Kady (Sandia National Labs., Albuquerque, NM)

A method for producing signals with coherence that decays exponentially as a function of the product of frequency and distance in the manner described by Corcos is proposed. This type of coherence constraint is used for simulating loads resulting from turbulent propagation or pressure signals resulting from wind noise. The present method is executed by stochastically perturbing the phase of the signal as a Gaussian random variable scaled by a simple function of frequency, distance, and a velocity-like decay parameter. While the present focus of this work is on producing signals that exhibit Corcos-type coherence decay, other varieties of coherence relationships including those associated with a diffuse field can also be produced with this approach. [*SNL is managed and operated by NTESS under DOE NNSA contract DE-NA0003525*.]

2:00

1pNSa5. An overview of aeroacoustics-related analyses for the Artemis-I mission. Makayle S. Kellison (Phys., Rollins College, Dept. of Phys., Rollins College - Box 2743, Winter Park, FL 32789, mkellison@rollins.edu), Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), Whitney L. Coyle (Phys., Rollins College, Winter Park, FL), Logan T. Mathews (Phys. and Astronomy, Brigham Young Univ., Provo, UT), Mark C. Anderson (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), and Grant W. Hart (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

To improve noise scaling laws for rockets and heated, supersonic jets, this paper discusses jet aeroacoustics-related scalings for the Space Launch System (SLS). Data from four measurement stations at the Artemis-I launch, located 1.4–1.8 km from the launchpad, are used in the analysis. First, SLS maximum overall sound pressure levels are compared against other rocket noise measurements at the same scaled distance using effective nozzle diameter. Second, Strouhal number frequency-scaling is used to compare the one-third-octave sound pressure and power level spectra with other jets and rockets. Third, the Oertel convective Mach number is used to interpret a maximum directivity angle of 60°–70°. Finally, SLS's acoustic efficiency is evaluated relative to that of other rockets.

2:15

1pNSa6. A study of rocket launch directivity and the potential correlation of various parameters. Matthew G. Yancey (Phys., Brigham Young Univ., Provo, UT 84602, mgy22@student.byu.edu), Ian R. Jackson (Phys., Brigham Young Univ., Provo, UT), Grant W. Hart (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), Logan T. Mathews (Phys. and Astronomy, Brigham Young Univ., Provo, UT), and Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

A rocket launch produces sound that is highly directional. This radiation is presumed to be due to Mach wave radiation, but the production mechanism is poorly understood. Previous research has shown that launch vehicle sound directivity peaks at an angle near 65° relative to the plume for most large rockets during early stages of the launch, yet limited attention has been given to the physical parameters involved in this sound production. This study explores the influence of some additional parameters that are not usually considered, such as ambient atmospheric pressure effects and the launch vehicle's Mach number on a rocket's directivity. Drawing parallels with similar studies on military jets, we aim to compare results and further our understanding of rocket acoustics, contributing to improved modeling precision.

2:30–2:45 Break

2:45

1pNSa7. Using empirical data to validate the role of computational fluid dynamics in various stages of aero-acoustic simulations. Sogand Okhovatian (Parklane Mech. Acoust., 1180 Speers Rd., Oakville, ON L6L 2X4, Canada, sogand@parklanemechanical.com) and Viken Koukounian (Parklane Mech. Acoust., Burlington, ON, Canada)

The purpose of utilizing higher level of understanding techniques is to improve the overall outcome of any process. As a full-service provider of complex engineering solutions to environmental noise problems, there is a need to house specialized knowledge to design and deliver bespoke solutions that are compatible with various constraints that implicate numerous subjects (acoustics, aerodynamics, structural, materials/chemical compatibility). The physics associated with seemingly simple products, such as an industrial acoustic silencer, is often complex. More specifically, its study should be described as aero-vibro-acoustical—whereby (1) airflow causes vibrations in the structure of the silencer, (2) the vibrations generate airborne and structureborne noise, and (3) components of the silencer (i.e., baffles) attenuate noise propagating through the duct. Motivated to expand our understanding of our products' performances, we are using Siemens software to circumvent exhaustive laboratory testing that is cost-prohibitive, and which is, generally, limited to common geometries and parameters. A systematic approach is necessary to validate correlations between simulated results with empirical data. This is accomplished by, first, correlating the aerodynamic performance of products using computational fluid dynamics (CFD) to predict pressure drop values and the distribution of forces on the structure, to then leverage additional solvers to assess the vibro-acoustical stage of the analysis.

3:00

1pNSa8. A photographic analysis of Mach wave radiation from a rocket plume. Grant W. Hart (Dept. of Phys. and Astronomy, Brigham Young Univ., N357 ESC, Provo, UT 84602, grant_hart@byu.edu), Kent L. Gee, Eric G. Hintz, Giovanna Nuccitelli (Dept. of Mech. Eng., Brigham Young Univ., Provo, UT), and Trevor Mahlmann (Cape Canaveral, FL)

At 7:30 AM on October 6, 2020 Space-X launched a Falcon-9 rocket from pad 39A at Kennedy Space Center. Photographer Trevor Mahlmann had positioned his camera in the location where the rocket would pass in front of the rising sun and took a series of images of that encounter. These images are visually stunning, but they are also scientifically useful. The high-intensity sound and shock waves originating in the plume are imaged by passing in front of the sun, particularly near the edge of the sun. This can be considered a background-oriented schlieren system. The sound from a supersonic rocket plume is thought to be due to Mach wave radiation, but the details are not yet well understood. We are analyzing these images to determine the position and radiation characteristics of the sound source in the plume. The results of this analysis will be presented.

3:15

1pNSa9. Development of a far-field prediction model for tactical jet noise spectra. Hunter J. Pratt (Phys. and Astronomy, Brigham Young Univ., C110 ESC, Provo, UT 84602, hpratt7@byu.edu), Logan T. Mathews (Phys. and Astronomy, Brigham Young Univ., Provo, UT), and Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Semi-empirical models for predicting far-field jet noise radiation are computationally efficient. One historical example is NASA SP-8072 (1971), which couples source sound power spectra with frequency-dependent directivity indices to obtain far-field radiated spectra for rockets. This paper applies the SP-8072 framework to develop spectra for the T-7A trainer aircraft. Data at 38 and 76 m arcs are used to obtain sound power spectra and frequency-dependent directivity indices for four different engine conditions ranging from 36% thrust through maximum afterburner. Application of a ground reflection correction model is discussed. The model is applied to obtain results at distances between 19 and 229 m and the performance of the model is evaluated using measured data. [Work supported by ONR Grant No. N00014-21-1-2069.]

3:30

1pNSa10. Acoustic source characterization of shock-associated noise in an installed, full-scale F404 engine. Logan T. Mathews (Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, lmathew3@byu.edu) and Kent L. Gee (Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Shock-associated noise in supersonic jets is an important factor in acoustic radiation, particularly in the upstream direction. A previous paper [L. T. Mathews *et al*., *J. Acoust. Soc. Am.* **154**, A325 (2023)] discussed a preliminary source characterization of shock-associated noise in a full-scale, installed tactical jet engine. Acoustical holography was used to reconstruct the acoustic behavior near the jet and high-amplitude features were produced that were consistent with shock-associated noise sources. This paper presents a more detailed characterization of shock-related noise in the jet using higher-fidelity methods, such as subarray processing and bandwidth extension. The spatial and spectral characteristics of the shock cell noise sources are discussed. A virtual reference decomposition method is used to examine the source and radiative characteristics of each detected shock noise source. The results are compared with shock noise characteristics in the literature. [Work supported by ONR Grant No. N00014-21-1-2069.]

3:45

1pNSa11. Multiple wavepacket source decomposition methods applied to an F404 engine noise source. Tyce W. Olaveson (Phys. and Astronomy, Brigham Young Univ., N284 Eyring Sci. Ctr. BYU, Provo, UT 84602, tyceolaveson@gmail.com) and Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Characterizing jet noise sources is an important part of noise reduction efforts. Many equivalent source models (ESMs) derived from inverse methods lack a compelling physical interpretation. Connecting the source model to flow properties is an important part of understanding noise radiation methods. Wavepackets are coherent, spatiotemporal structures observed in jet turbulence. Mathematical expressions describing wavepackets have been used in modal decompositions of the turbulent flow. Similar structures appear in frequency domain ESMs and these methods have been applied to create physically meaningful, reduced-order models. This paper applies a multiple wavepacket decomposition method developed by Harker [B. M. Harker, BYU Ph. D Dissertation (2017)] to acoustic data collected near a T-7A-installed F404 engine. Individual wavepackets are used to create total-field reconstructions and comparisons are made to the dual-lobe behavior observed in full-scale jets. Finally, sound power characteristics of each wavepacket are used to effectuate a sound power decomposition of the noise source. [Work supported by ONR Grant No. N00014-21-1-2069.]

MONDAY AFTERNOON, 13 MAY 2024 — ROOM 201, 1:00 P.M. TO 5:20 P.M.

Session 1pNSb

Noise, Psychological and Physiological Acoustics, and ASA Committee on Standards: Evaluation of Hearing Protection Devices with Impulse Noise and Acoustic Test Fixtures

Jeremie Voix, Cochair
École de technologie supérieure, Université du Québec, 1100 Notre-Dame Ouest, Montréal H3C 1K3, Canada

William J. Murphy, Cochair
Stephenson and Stephenson, Research and Consulting, 5706 State Route 132, Batavia, OH 45103

Chair's Introduction—1:00

Invited Papers

1:05

1pNSb1. Effect of shooting glasses on earmuff attenuation measured with an acoustic test fixture for gunfire impulses. Gregory Flamme (Stephenson and Stephenson Res. and Consulting, 5706 State Rte. 132, Batavia, OH 45103, gflamme@sasrac.com), William J. Murphy, Stephen M. Tasko (Stephenson and Stephenson Res. and Consulting, Batavia, OH), Donald S. Finan (Audiol. & Speech-Lang. Sci., Univ. of Northern Colorado, Greeley, CO), Deanna K. Meinke (Audiol. & Speech-Lang. Sci., Univ. of Northern Colorado, Greeley, CO), Kristy K. Deiters (Stephenson and Stephenson Res. and Consulting, Batavia, OH), James E. Lankford (Allied Health and Commun. Disord., Northern Illinois Univ., Dekalb, IL), and Michael Stewart (Dept. of Commun. Sci. and Disord., Central Michigan Univ., Mount Pleasant, MI)

The EPA requirement for hearing protector labels is based on the decrepit ANSI S3.19-1974 standard, which prohibits the use of eyewear during the REAT testing that leads to the NRR on the product label. Firearm users are recommended to wear protective eyewear while shooting. In this presentation, we review past work regarding the effects of eyewear on earmuff performance and present results from five samples of one product that includes both eyewear and an earmuff. Impulses ranging between 139 and 178 dB peak SPL were used. Impulse Insertion Loss was less than 10 dB below 400 Hz for earmuffs alone and negligible for the earmuff and glasses condition and attenuation in the high frequencies was reduced by approximately 15 to 25 dB. Impulse peak insertion loss values per ANSI S12.42 were reduced from 21 to 39 dB for the earmuff alone to 15 to 22 dB when eye protection was added. Hearing conservation programs need to account for the deleterious effect of protective eyewear on earmuffs, and additional studies with broad combinations of earmuffs and eyewear are needed to establish the range over which eyewear can be expected to compromise earmuff attenuation.

1:25

1pNSb2. Comparison of two acoustic test fixtures for measuring impulse level dependent attenuation. William J. Murphy (Stephenson and Stephenson, Res. and Consulting, 5706 State Rte. 132, Batavia, OH 45103, wmurphy@sasrac.com), Gregory Flamme (Stephenson and Stephenson, Res. and Consulting, Batavia, OH), Stephen M. Tasko (Stephenson and Stephenson, Res. and Consulting, Batavia, OH), Kristy K. Deiters (Stephenson and Stephenson, Res. and Consulting, Batavia, OH), Deanna K. Meinke (Audiol. & Speech-Lang. Sci., Univ. of Northern Colorado, Greeley, CO), Donald S. Finan (Audiol. & Speech-Lang. Sci., Univ. of Northern Colorado, Greeley, CO), James E. Lankford (Allied Health and Commun. Disord., Northern Illinois Univ., Dekalb, IL), and Michael Stewart (Dept. of Commun. Sci. and Disord., Central Michigan Univ., Mount Pleasant, MI)

In the American National Standards Institute (ANSI) S12.42-2010 standard for measuring insertion loss of hearing protection devices (HPDs) in impulsive noise, the range of peak sound pressure levels for the impulses is 132 to 168 dB peak sound pressure level (dB pSPL). The specification of the acoustic test fixtures (ATFs) included an IEC 60318-4 ear simulator fitted with a microphone. The microphones typically used are a 1/4" cartridge and have a dynamic range up to about 169 dB pSPL. For most small caliber firearms, the peak SPLs are less than 170 dB SPL at the location of the shooters ear (Schulz *et al*. 2013). For some military weapon systems that produce impulses greater than 170 dB pSPL, impulse insertion loss cannot be assessed without exceeding the microphone maximum dynamic range. Some condenser microphones have maximum dynamic ranges of 190 dB pSPL. This paper compares insertion losses for HPDs measured on two GRAS 45CB ATFs with microphones with maximum dynamic ranges of 169 and 193 dB pSPL. Waveform distortion, field probe to unoccluded transfer functions between the ATF and field probe will be examined for distortion and accuracy when estimating the impulse level dependent attenuation.

1:45

1pNSb3. Development of a biofidelic acoustic test fixture and evaluation in impulsive noise. Jed C. Wilbur (Creare LLC, 16 Great Hollow Rd., Hanover, NH 03755, jcw@creare.com), Brian Graybill, and Odile Clavier (Creare LLC, Hanover, NH)

Here, we describe the development and use of a novel biofiedlic head surrogate for evaluating head and hearing protection devices against impulsive, shock, and blast insults. The head surrogate, derived from biomedical imagery, includes biofidelic pinnea, ear canals, and skull. In addition to microphones positioned at the ear canal terminus, the surrogate includes accelerometers and a hydrophone to provide an estimate of sound propagation through the bone-conduction pathways. This allows use of the surrogate to estimate a device's ability to attenuate bone-conducted pressure waves. We discuss the tradeoffs between biofidelity, manufacturability, instrumentation, and cost. We will show example data from various test campaigns, including the estimating the attenuation of the bone-conduction pathway afforded by different helmet systems during a 178 dB SPL (peak) shock insult. We will conclude with a look towards the future, including potentially updating the underlying three dimensional models to align with modern biomedical models and databases.

2:05

1pNSb4. Protection exhibited by hearing protection devices using test fixtures under extended shock tube exposure ranges. Theodore F. Argo (Appl. Res. Assoc., Inc., 7921 Shaffer Pkwy, Littleton, CO 80127, targo@ara.com), Kiersten Reeser (Appl. Res. Assoc., Inc., Albuquerque, NM), Alexandria Podolski (Appl. Res. Assoc., Inc., Littleton, CO), Nicholas Brunstad, Santino Cozza (Appl. Res. Assoc., Inc., Albuquerque, NM), and Gregory T. Rule (Appl. Res. Assoc., Inc., Littleton, CO)

Hearing protection devices (HPDs) are used in a variety of environments to mitigate the risk of hearing injury, particularly in environments with pervasive impulsive noise. Evaluation of HPD performance for these conditions can be performed with acoustic test fixtures to ensure human subjects are not exposed to injurious levels of noise. Impulse Peak Insertion Loss (IPIL) and Impulse Spectral Insertion Loss (ISIL), as defined in ASA/ANSI S12.42, are used to quantify and compare performance of HPDs in exposure environments created by different impulsive sources. To simulate a range of exposures relevant across industries, multiple shock tubes were used to generate waveforms with A-durations ranging from 0.1 to 6.0 ms and peak pressures ranging from 130 to 183 dB. Protection provided by each device varied depending on the source conditions with passive and active devices producing fundamentally different responses depending on source characteristics. Ultimately, evaluation of devices under a variety of impulsive noise conditions allows for a more representative evaluation of HPD performance, bridging the gap between standard testing protocols and real-world acoustic environments, thus enabling better informed HPD selection.

2:25

1pNSb5. Designing an acoustical test fixture to evaluate the objective occlusion effect. Olivier Doutres (Mech. Eng., ETS (Ecole de technologie superieure), 1100, rue Notre-Dame Ouest, Montreal, QC H3C 1K3, Canada, olivier.doutres@etsmtl.ca), Yu Luan, Marc-Olivier Cyr-Desroches, Kévin Carillo (Mech. Eng., École de technologie supérieure, Montréal, QC, Canada), Robin Richert (Ecole Nationale Superieure d'Ingenieurs du Mans (ENSIM), Le Mans, France), and Franck Sgard (IRSST, Montreal, QC, Canada)

Earplugs are commonly used to prevent noise-induced hearing loss. However, their effectiveness is often hindered by the discomfort they cause, impacting consistent and correct use. An important acoustical discomfort, known as the occlusion effect, arises from an increased perception of bone-conducted physiological sounds (such as one's own voice, breathing, and chewing) when the ear canal is occluded. To objectively assess this discomfort, the study proposes the use of an acoustical test fixture (ATF) that avoids direct measurements on human participants. The ATF employs an anatomically realistic truncated outer ear, incorporating soft tissues, cartilage, and bone components to replicate the outer ear's bone conduction path, crucial for occlusion effect assessments. The study demonstrates that the proposed ATF can replicate key effects observed in objective ccclusion effect (OE) measurements on human participants, including significant OE at low frequencies diminishing with increasing frequency, reduction of OE with greater insertion depths, and distinctions among various earplug types—especially noticeable at deeper insertions. Furthermore, a computationally efficient finite element method-based virtual tester for the ATF is developed and validated.

2:45–3:00 Break

Contributed Papers

3:00

1pNSb6. Influence of reference transducer location on hearing protector attenuation metrics for impulse noise. William J. Murphy (Stephenson and Stephenson, Res. and Consulting, 5706 State Rte. 132, Batavia, OH 45103, wmurphy@sasrac.com), Stephen M. Tasko, Gregory Flamme, Kristy K. Deiters (Stephenson and Stephenson, Res. and Consulting, Batavia, OH), Deanna K. Meinke, Donald S. Finan (Audiol. & Speech-Lang. Sci., Univ. of Northern Colorado, Greeley, CO), James E. Lankford (Allied Health and Commun. Disord., Northern Illinois Univ., Dekalb, IL), and Michael Stewart (Dept. of Commun. Sci. and Disord., Central Michigan Univ., Mount Pleasant, MI)

The ANSI/ASA S12.42-2010 standard can be used to estimate the attenuation for impulsive noises between 130 and 170 decibels peak sound pressure level (dB pSPL). The ANSI/ASA S12.42 method uses a complex transfer function between the signal received in the ear of an acoustic test fixture (ATF) and a reference transducer. The reference transducer is at the same radial distance from the impulse source and at less than 30 degrees from the ATF angle. The reference transducer is specified as a pencil-type pressure (> 40.6 cm long). The maximum angle and the minimum pencil probe dimensions could be incompatible near the source, or may introduce artifacts or hazards. In this study, several hearing protectors were evaluated with two rifles (A-Bolt .300 Winchester Magnum and Colt AR-15 5.56 caliber), two ATFs (GRAS 45CB), and five transducers (four 1/4" microphones and one surface mount microphone). The ATFs were on opposite sides of the gun at distances for field levels between 180- and 140-dB pSPL. For each location and rifle, S12.42 metrics were calculated for all 20 combinations of ATF ear and reference transducer. Results indicated that S12.42 metrics were minimally affected by the locations of the reference transducers.

3:15

1pNSb7. Toward a standard for assessing civilian firearm suppressor noise reduction: Environmental and procedural considerations. Stephen M. Tasko (Stephenson and Stephenson, Res. and Consulting, 5706 State Rte. 132, Batavia, OH 45103, stasko@sasrac.com), William J. Murphy, Gregory Flamme, and Kristy K. Deiters (Stephenson and Stephenson, Res. and Consulting, Batavia, OH)

Currently, no consensus standard exists for measuring firearm suppressor noise reduction (FSNR). NATO developed a military standard to measure suppressor performance for an environment that does not generalize well to the recreational shooter. We propose a FSNR measurement standard using maximum accumulated A-weighted energy. Test environments included an outdoor space and a reverberant (indoor) facility with dimensions expected to provide similar results to the outdoor space. Two standard test barrels (0.308 and 0.223 caliber) mounted on a universal receiver system were used. For each environment and caliber, two ammunition types and three suppressors were tested, along with an unsuppressed condition. Ten discharges were recorded for each condition. Five field microphones were positioned at four locations around the universal receiver. Impulses in quasi-free field and reverberant environments are compared and the impact of combustion gases within the suppressor and the number of discharges to achieve stable estimates of noise reduction are considered. Similar FSNRs were observed in the environments. Clearing the suppressor of combustion gases had variable effects on impulse level, and relatively stable estimates of suppressor attenuation were achieved with seven test discharges.

3:30

1pNSb8. Military health system auditory blast injury prevention standard. Elizabeth B. Brokaw (MITRE, The MITRE Corp., 7515 Colshire Dr., McLean, VA 22102-7539, ebrokaw@mitre.org), Raj K. Gupta (DoD Blast Injury Res. Coordinating Office, Fort Detrick, MD), Rachel Spencer, and Lisa Lalis (MITRE, McLean, VA)

The Department of Defense (DoD) lacks a single medical standard for prevention of auditory blast injury, relying on the Service Branches to develop their own guidance. The DoD Executive Agent established the Blast Injury Prevention Standards Recommendation (BIPSR) process, led by the DoD Blast Injury Research Coordinating Office (BIRCO), to investigate the effects of blast injuries on warfighters. The BIPSR Process implementation of the Auditory Blast Injury Type illuminated characteristics, capabilities, and limitations of current capabilities to predict and prevent auditory injuries. The evaluation resulted in the recommendation of 8-h Equivalent Level (LAeq8hr) as the interim MHS Auditory Blast Injury Prevention Standard to serve as the cross-Service standard while the community performs additional research to fill critical knowledge gaps. BIRCO is working to formalize the Military Health System Auditory Blast Injury Prevention standard to support updates and coordination across DoD guidance including military design standards and health protection policies.

Invited Papers

3:45

1pNSb9. Measuring hearing protector attenuation of impulse noise on acoustic test fixtures using maximum A-weighted energy reduction. Stephen M. Tasko (Stephenson and Stephenson, Res. and Consulting, 5706 State Rte. 132, Batavia, OH 45103, stasko@sasrac.com), William J. Murphy, Gregory Flamme, Kristy K. Deiters (Stephenson and Stephenson, Res. and Consulting, Batavia, OH), Deanna K. Meinke, Donald S. Finan (Audiol. & Speech-Lang. Sci., Univ. of Northern Colorado, Greeley, CO), James E. Lankford (Allied Health and Commun. Disord., Northern Illinois Univ., Dekalb, IL), and Michael Stewart (Dept. of Commun. Sci. and Disord., Central Michigan Univ., Mount Pleasant, MI)

Hearing protection devices (HPDs) and firearm suppressors can help mitigate risk of hearing loss from small caliber firearms. ANSI S12.42 is the standard for measuring HPD impulse peak insertion loss (IPIL). No consensus standard exists for measuring firearm suppressor noise reduction, though the industry is working toward a standard using maximum accumulated A-weighted energy. A common attenuation metric for different impulse noise reduction technologies would inform the individual and combined impact of these technologies. The maximum A-weighted energy reduction for several HPDs was assessed using firearm impulses from two rifles (0.300 Winchester Magnum and 5.56x45) with two ATFs and five field microphones positioned on opposite sides of the gun at four locations (field levels between 180- and 140-dB pSPL). No adjustments were made to account for tissue/bone conduction. Maximum A-weighted energy reduction values for single protection ranged between 17 dB (earmuff with safety glasses) and 43 dB (preformed earplug). Similar to IPIL, maximum A-weighted energy reduction increased with increased impulse levels. Advantages, disadvantages, and technical issues associated with the procedure will be discussed.

4:05

1pNSb10. Evaluation of a multi-function in-ear device performance in the presence of impulse noise using acoustic test fixtures.
Chris Brooks (Creare LLC, Hanover, NH, cab@creare.com), Matt Swanson, and Odile Clavier (Creare LLC, Hanover, NH)

Real-time noise exposure monitoring has the potential to assess risk before damage occurs. We present on the development of an in-ear device that (1) provides hearing protection; (2) measures the noise in the ear canal to estimate exposure dose; and (3) monitors distortion product otoacoustic emissions (DPOAE) over time. To validate the device's ability to measure impulse noise under hearing protection, we measured the response in a field test. A single AR15 was shot multiple times, and four prototypes were used in multiple acoustic test fixtures (ATFs) at different relative distances from the exit of the gun. The ATFs included a G.R.A.S 45 CB in addition to a head simulator designed to detect both air- and bone-conducted sound. We evaluated the impulse peak insertion loss (IPIL) metric of the device inserted by itself or in combination with other hearing protection ("double protection" condition) to assess attenuation of impulsive noises. We also compared the device's microphone measurements to those obtained through the ATF microphones, and to the recordings of external microphones located at or near the ear of each ATF. Results demonstrate the value of testing in-ear devices in acoustic test fixtures during development phases.

4:25

1pNSb11. Design and evaluation of a noise dosimeter integrated into an ear-muff hearing protector. Christopher J. Smalt (MIT Lincoln Lab., 244 Wood St., Lexington, MA 02420, Christopher.Smalt@ll.mit.edu), Chris Brooks, Eric Yuan, and Odile Clavier (Creare LLC, Hanover, NH)

In impulse-noise environments, such as during weapons training, it can be difficult to characterize an individual's true cumulative exposure, and consequently their risk of hearing loss. Direct measurements of sound pressure in the ear canal underneath hearing protection are ideal for assessing the protected noise exposure because the effects of distance, orientation, protection, etc., are automatically included in the measurement. However, these types of measurements are not commonly collected outside the laboratory. In this research, we develop and evaluate a portable a noise dosimeter that is integrated into a commonly used muff-based hearing protector (3M^{TM} PELTOR^{TM} ComTac^{TM} V). The design of the device aims to simultaneously measure both the external and the under-the-muff sound pressures in a portable, wearable form-factor. First, we demonstrate that the modifications do not affect the hearing protector attenuation using an acoustic test fixture and a fit-check system across 10 refits ($p = 0.14$). Next, we show through testing with a shock-tube and acoustic test fixture that it is possible to obtain valid pressure measurements through this system. Finally, we report on initial field testing during weapons training. Future application of this technology can be used to develop damage risk criteria based on in-ear measurements that will provide for more accurate individualized risk assessment.

4:45

1pNSb12. Immersive auditory awareness: A smart earphones platform for education on noise-induced hearing risks. Alexis Pinsonnault-Skvarenina, Rachel Bouserhal, Valentin PINTAT (École de technologie supérieure, Université du Québec, Montréal, QC, Canada), and Jeremie Voix (École de technologie supérieure, Université du Québec, 1100 Notre-Dame Ouest, Montréal, QC H3C 1K3, Canada, jeremie.voix@etsmtl.ca)

The Audio Research Platform developed by the ÉTS-EERS Industrial Research Chair in In-Ear technologies is a digital audio processing wearable device that features a powerful Digital Signal Processor and a pair of wired earplugs featuring outer- and inner-ear microphones as well as miniaturize loudspeaker. ARP has been used since over the last 2 decades by CRITIAS to develop new algorithms for advanced hearing protection and communication in noise, as well as in-ear audio sensing. In this adaptation, the ARP enables transparent hearing of ambient sounds, simulates varying degrees of hearing loss, and includes a tinnitus simulator. ARP also functions as a personal music player, monitoring playback levels and displaying the "Age of your ears") a metric recently proposed by CRITIAS), predicting accelerated auditory aging from excessive music playback. In the playback mode, wearers experience pre-recorded noise, prompting vocal adjustments known as the "Lombard Effect." ARP analyzes the wearer's voice using the outer-ear microphone, illustrating level and pitch shifts in speech due to a noisy environment. This paper details ARP's technical aspects, positioning it as a powerful tool to inform the public about noise-induced hearing loss and promote awareness and prevention through enjoyable real-time demonstrations.

Contributed Paper

5:05

1pNSb13. Revising normative standards for personal noise dosimeters.
Peter Hanes (National Res. Council of Canada, 1200 Montreal Rd., Ottawa, ON K1A 0R6, Canada, peter.hanes@nrc-cnrc.gc.ca)

Personal noise dosimeters are designed to monitor noisy environments and are usually intended to attached to an individual person in order to estimate their exposure to noise over a given period of time. These devices are often used to indicate sound exposure as a percentage of a predetermined criterion. However, different jurisdictions employ different definitions of such quantities and different criteria for permissible exposure. Therefore, existing normative standards for measurement instruments differ in their specifications. For example, while the ASA/ANSI S1.25 standard *Specification for Personal Noise Dosimeters* makes provisions for use of three level-time exchange rates (3, 4, and 5 dB per doubling of exposure time), the international standard for personal sound exposure meters (IEC 61252) permits only an exchange rate of 3 dB. The International Electrotechnical Commission is revising IEC 61252 to modernize and harmonize requirements for the instruments to reflect the actual practice of measurements of noise exposure worldwide. The technical aims of the revision are to provide realistic specifications, consistent methods for testing all relevant characteristics of a model of instrument, and affordable methods for periodic testing of individual instruments. Potential consequences of the likely changes to IEC 61252 are presented and discussed.

Session 1pPAa

Physical Acoustics, Engineering Acoustics, Computational Acoustics, and Signal Processing in Acoustics: Cyber Threats and Acoustical Systems

Kavitha Chandra, Cochair
Electr. and Comput. Eng., College of Eng., Univ. Massachusetts Lowell, 1 Univ. Ave., Lowell, MA 01854

Charles Thompson, Cochair
Electr. and Comput Eng., UMASS Lowell, 1 Univ. Ave., Lowell, MA 01854

Chair's Introduction—1:00

Invited Papers

1:05

1pPAa1. Enhancing voice biometric security: Evaluating neural network and human capabilities in detecting cloned voices. Andrzej Czyzewski (Multimedia Systems, Gdansk Univ. of Technol., Narutowicza 11/12, Gdansk 80-233, Poland, andczyz@gmail.com)

This study assesses speaker verification efficacy in detecting cloned voices, particularly in safety-critical applications such as healthcare documentation and banking biometrics. It compares deeply trained neural networks like the Deep Speaker with human listeners in recognizing these cloned voices, underlining the severe implications of voice cloning in these sectors. Cloned voices in healthcare could endanger patient safety by altering medical records, causing inaccurate diagnoses and treatments. In banking, they threaten biometric security, increasing the risk of financial fraud and identity theft. The research reveals the neural network's superiority over human detection in pinpointing cloned voices, underscoring the urgent need for sophisticated AI-based security. The study stresses the importance of developing robust defenses against voice cloning attacks, which can have critical consequences in healthcare and fintech. This research is crucial for enhancing security in areas reliant on voice authentication, safeguarding confidential data, and preserving the integrity of vital services. The Polish National Center for Research and Development (NCBR) initially supported the project "BIOPUAP" (POIR.01.01.01-0092/19), which focused on digital banking. Subsequently, the project "ADMEDVOICE" (INFOSTRATEG4/0003/2022), also supported by the NCBR, conducted further research into voice cloning in the healthcare sector.

1:30

1pPAa2. Utilizing Near-Ultrasound Inaudible Trojan (NUIT) to compromise microphone security. Guenevere (Qian) Chen (Elec. and Comput. Eng., Univ. of Texas at San Antonio, 1 UTSA Circle, San Antonio, TX 78249-1644, guenevereqian.chen@utsa.edu)

Voice Control Systems (VCSs) provide a user-friendly interface for issuing voice commands to smart devices. However, the security of VCSs remains a challenge, as evident from two distinct attack classes: (i) inaudible attacks, feasible when the attacker is in proximity to the victim, and (ii) audible attacks, which can be executed remotely by embedding attack signals into audio. In talk introduces a novel class of attacks known as Near-Ultrasound Inaudible Trojan (NUIT). NUIT attacks combine the advantages of both classes, being both inaudible and capable of remote execution. Additionally, NUIT attacks can achieve end-to-end unnoticeability, a crucial aspect often overlooked in the literature. Notably, NUIT attacks leverage victim speakers to compromise victim microphones and associated VCSs, eliminating the need for specialized speakers. The feasibility of NUIT attacks is demonstrated, and an effective defense against them is proposed.

1:55

1pPAa3. Tubes among us: Analog attack on automatic speaker identification. Kassem Fawaz (Elec. and Comput. Eng., Univ. of Wisconsin-Madison, 1415 Eng. Dr., Madison, WI 53706, kfawaz@wisc.edu) and Shimaa Ahmed (Elec. and Comput. Eng., Univ. of Wisconsin-Madison, Madison, WI)

Recent years have seen a surge in the popularity of acoustics-enabled personal devices powered by machine learning. Yet, machine learning has proven to be vulnerable to adversarial examples. A large number of modern systems protect themselves against such attacks by targeting artificiality, i.e., they deploy mechanisms to detect the lack of human involvement in generating the adversarial examples. However, these defenses implicitly assume that humans are incapable of producing meaningful and targeted adversarial examples. In this paper, we show that this base assumption is wrong. In particular, we demonstrate that for tasks like speaker identification, a human is capable of producing analog adversarial examples directly with little cost and supervision: by simply speaking through a tube, an adversary reliably impersonates other speakers in the eyes of ML models for speaker identification. Our findings extend to a range of other acoustic-biometric tasks such as liveness detection, bringing into question their use in security-critical settings in real life, such as phone banking.

2:20

1pPAa4. SoK: Sensor wars: Attacks and defenses on acoustic sensors. Fidel Castro, Kavitha Chandra (Elec. and Comput. Eng., Univ. of Massachusetts Lowell, Lowell, MA), and Orlando Arias (Elec. and Comput. Eng., Univ. of Massachusetts Lowell, 1 University Ave., Ball Hall 407A, Lowell, MA 01854, orlando_arias@uml.edu)

In this talk, we will examine attacks that can be performed against sensors that have become prevalent in smart devices, Internet of Things, and Cyber-Physical Systems. These devices have become part of our everyday life, integrated, for example, in health monitoring solutions, autonomous vehicles, and home automation. A classification of current threat models on sensors is presented. Of particular interest are attacks that make use of interference to trick devices into performing unintended actions. This talk serves as a summary of knowledge on the state of the art in utilizing sensor data for improving the security of smart devices and systems with the goal of inspiring new interdisciplinary research between the acoustics and cybersecurity communities.

2:35–2:50 Break

2:50

1pPAa5. Nonlinear bone conducted hearing. Charles Thompson (Elec. and Comput. eng, UMASS Lowell, 1 Univ. Ave. Lowell, MA 01854, charles_thompson@uml.edu), Masoumeh Farhadi Nia, Emi Aoki, Flore Norceide, Vinh T. Tran, and Kavitha Chandra (Elec. and Comput. Eng., Univ. of Massachusetts Lowell, Lowell, MA)

Air conduction (AC) is the typical mode of hearing used by humans. AC process follows a defined path starting at the ear canal, middle ear to the cochlea, and finally to the sensory organs of hearing. The highest frequency we can hear using AC is approximately 20 kHz. Bone conduction (BC) may offer a high-bandwidth alternative to AC, where the highest frequency is 100 KHz. BC utilizes direct stimulation of the skull, which will then vibrate the cochlea and its structures. In this presentation, nonlinear mechanisms in bone-conducted hearing is examined. Nonlinear conduction that may be used to demodulate an ultrasonic signal to yield an audible signal supplying the cochlea is of particular interest.

3:05

1pPAa6. Listen to your key: Towards acoustics-based physical key inference. Soundarya Ramesh (CS, National Univ. of Singapore, 15 Computing Dr., Singapore 117418, Singapore, soundaryaramesh96@gmail.com), Rui Xiao (Zhejiang Univ., Hangzhou, China), Anindya Maiti (Univ. of Oklahoma, Norman, OK), Jong Taek Lee (Kyungpook National Univ., Daegu, Republic of Korea), Harini Ramprasad (CS, National Univ. of Singapore, Seattle, WA), Ananda Kumar (CS, National Univ. of Singapore, Singapore, Singapore), Murtuza Jadliwala (Univ. of Texas at San Antonio, San Antonio, TX), and Jun Han (Yonsei Univ., Seoul, Republic of Korea)

Physical locks are one of the most prevalent mechanisms for securing objects such as doors. While many of these locks are vulnerable to lock-picking, they are still widely used as lock-picking requires specific training with tailored instruments and easily raises suspicion. To overcome the limitations of lockpicking, we propose a novel attack vector that leverages the audio recording of the key insertion in order to infer the shape of victim's key, namely, bittings (or cut depths) which form the secret of a key. In particular, we show that computing the timing interval between audible click sounds that occur during key insertion enables inferring the bitting information, i.e., shape of the physical key. Such an audio-based attack has several advantages—unlike lock-picking, it minimizes the attacker's physical access to the lock, thus reducing the risks of them being apprehended. Second, as the attack only requires a microphone to launch the attack (e.g., a smartphone microphone), it significantly lowers the bar for the required expertise of the attacker. Despite the advantages, there are several challenges in extracting the required key-related signal from the audio. In this talk, we will discuss how we overcome the challenges and present initial results depicting the feasibility of audio-based key inference. This talk is based on two conference papers submitted to ACM HotMobile 2020 and USENIX Security 2021 on the same topic.

3:20

1pPAa7. Acoustic feature analysis of directional sound from parametric array speakers with application to mitigate attacks on voice-activated systems. Vinh T. Tran (Elec. and Comput. Eng., Univ. of Massachusetts Lowell, 1 University Ave. Lowell, MA 01854, vinh_tran2@student.uml.edu), Kathryn Quinn, Flore Norceide, Emi Aoki, Kavitha Chandra (Elec. and Comput. Eng., Univ. of Massachusetts Lowell, Lowell, MA), and Charles Thompson (Elec. and Comput. Eng, UMASS Lowell, Lowell, MA)

A number of studies have identified the vulnerabilities of voice-activated devices to attacks from amplitude-demodulated sound from directional speakers. In this research, a controlled set of speech data, both live and played back from directional speakers, is analyzed to identify the acoustic features integral to speaker recognition in smart devices. A category of features, including Mel-frequency cepstral coefficients derived from Mel filter banks, are examined for potential to differentiate and classify speech sources. Potential vulnerabilities in voice-activated systems are systematically identified by examining the acoustic characteristics of diverse voice samples. This analytical approach aims to unveil potential weaknesses that could be exploited, triggering responses from these systems. The findings of this study contribute to the comprehension of the security landscape surrounding voice-activated technologies, paving the way for enhanced cybersecurity measures within this domain.

3:35

1pPAa8. Acoustic streaming in the Cochlea resulting from periodic excitation. Aidan Keefe (Elec. and Comput. Eng., Univ. of Massachusetts Lowell, 1 Univ. Ave. Lowell, MA 01854, aidan_keefe@student.uml.edu) and Charles Thompson (Elec. and Comput. Eng., UMASS Lowell, Lowell, MA)

This paper presents a model describing acoustic streaming in the scala vestibuli and the scala tympani of the cochlea. The magnitude and spatial features of the acoustic streaming are a function of the vibration of the cochlear partition. Hence, spatial adjustment of the acoustic streaming field can be accomplished by suitable harmonic adjustment of the excitation. It is shown that the dynamic behavior of the fluid is a function of the values of two nondimensional parameters S and R. The Strouhal number S is the ratio of the typical length to the oscillatory particle displacement and the oscillatory Reynolds number R is the ratio of typical length to the oscillatory boundary layer thickness. The method of matched asymptotic expansions is used to obtain a solution that is valid in the limit as 1/S tends to zero. The solution is shown for the slow streaming $R/S = O(1)$ and moderate streaming $R/S^2 = O(1)$ cases.

Session 1pPAb

Physical Acoustics, Structural Acoustics and Vibration, Signal Processing in Acoustics, and Engineering Acoustics: Infrasound

Philip S. Blom, Cochair
Earth & Environ. Sci., Los Alamos National Lab., P.O. Box 1663, M/S F665, Los Alamos, NM 87545

David N. Green, Cochair
AWE Blacknest, Brimpton RG7 4RS, United Kingdom

Invited Papers

1:00

1pPAb1. Generalized least squares beamforming of infrasound data. Jordan W. Bishop (Los Alamos National Lab., P.O. Box 1663, Los Alamos, NM 87545, jwbishop@lanl.gov), Philip S. Blom, and Jeremy Webster (Los Alamos National Lab., Los Alamos, NM)

Generalized least squares (GLS) beamforming is a method for determining the direction of arrival and trace velocity of transient infrasound signals that may be otherwise obscured by persistent, correlated background noise, such as microbaroms. This method complements the adaptive F-detector by using an estimate of the noise background to form a generalized power ratio, which is used to estimate plane wave parameters (trace velocity and back-azimuth). Using a suite of fully synthetic signals, we first investigate the resolving power of the GLS estimator as a function of signal to noise ratio compared with a conventional, non-adaptive estimator. Fstatistic and plane wave parameter estimates will then be compared between GLS beamforming, conventional Bartlett beamforming, Capon beamforming, and the MUSIC algorithm. Recorded infrasonic signals from the Forensics Surface Experiment, where a persistent signal was observed from the south, will then be used to evaluate the GLS method. Initial analyses suggest that GLS beamforming results in a lower F-statistic during noise regions and a higher F-statistic value for transient signals compared to the Bartlett beam. Algorithmically determining an optimal window to characterize the background noise presents a significant challenge and different approaches will be discussed.

1:20

1pPAb2. Estimating infrasonic noise levels using a high resolution weather model. Jelle D. Assink (R&D Seismology and Acoust., KNMI, Utrechtseweg 297, Utrecht 3731GA, Netherlands, jelle.assink@knmi.nl), Madelon Smink (R&D Seismology and Acoust., KNMI, De Bilt, Netherlands), Fred Bosveld (R&D Observations & Data Technol., KNMI, De Bilt, Netherlands), and Läslo G. Evers (R&D Seismology and Acoust., KNMI, Utrecht, Netherlands)

The presence of turbulence and other wind-induced pressure fluctuations are considered a nuisance in infrasound monitoring. For this reason, wind noise filters are typically in place for suppression. In this study, we establish a relation between microbarometer observations and *in situ* turbulence measurements at the Cabauw Atmospheric Research Site in The Netherlands. Using this relation, we forecast turbulent pressure levels using forecasted levels of turbulence kinetic energy (TKE) that are computed as part of the high-resolution (2.5×2.5 km grid scale) HARMONIE weather model. The estimates are compared with pressure noise levels as a function of frequency. This approach has two foreseen applications: (1) the forecasted turbulence fields could possibly help in site selection for infrasound arrays and (2) microbarometer observations could possibly be of use in the further refining of sub-grid scale turbulence schemes in weather models.

1:40

1pPAb3. Exploring a planet with infrasound: Challenges in probing the subsurface and the atmosphere. Sven Peter Näsholm (Dept. of Informatics, Univ. of Oslo, Oslo, Norway), Quentin Brisaud (NORSAR, Gunnar Randers vei 15, Kjeller 2007, Norway, Quentin.Brissaud@norsar.no), Antoine Turquet, Tina Kaschwich, and Marouchka Froment (NORSAR, Kjeller, Norway)

Infrasound waves have shown great potential to retrieve a range of geophysical parameters across scales, such as atmospheric structures, surface and buried source characteristics, and seismic velocity structures. In particular, infrasound generated by seismic waves coupling to the atmosphere can provide unique insight into a planet's interior. However, utilizing infrasound data require efficient forward wave simulation techniques and an accurate description of waveform sensitivity to source and medium parameters. Even under idealized conditions, many of the inverse problems associated with infrasound-based probing are inherently ill-posed, requiring regularization or other approaches to yield reliable output. In this contribution, we highlight recent research results from our group and collaborators, tackling atmospheric probing at smaller and global scales, as well as innovative approaches to speed up infrasonic wave propagation modeling. We also call attention to the objectives and early results from a newly launched project focusing on the feasibility of exploring the interior of Venus using infrasound recorded from balloon platforms.

2:00

1pPAb4. The infrasonic choir: Decoding songs to inform decisions. Sarah McComas (US Army Res. and Development Ctr., 3909 Halls Ferry Rd., Vicksburg, MS 39180, sarah.mccomas@usace.army.mil)

The world around us is continuously evolving due to the actions of Mother Nature and anthropogenic activities, which impacts the decisions made on how we interact with the environment. Many of these changes are observable by listening to the infrasonic choir around us everyday. The voices of the infrasound choir include a wide variety of both natural (e.g., surf and volcanic activity) and man-made (e.g., explosions and infrastructure) sources. Understanding how to exploit the infrasonic song to make sense of the environment in near-real-time is the focus of ongoing research. An example of this is listening to infrastructure (bridges and dams) for insights into structural health, which can be utilized to prioritize limited resources after a natural disaster (hurricane or earthquake). This presentation explores operationalizing this capability, such as how to monitor source rich urban spaces, re-envisioning sensing in these environments with the development of lightweight/low-cost sensors and mobile arrays, and developing detection, classification, and localization methods for near-real-time processing. When successful, continuous monitoring of the infrasonic choir can provide a method for understanding the environment to inform decisions. Permission to publish was granted by the Director, Geotechnical and Structures Laboratory, U.S. Army Engineer Research and Development Center.

2:20

1pPAb5. Large-chamber reciprocity calibration for microbarometers. Chad M. Smith (Appl. Res. Lab., The Penn State Univ., State College, PA 16804, chad.smith@psu.edu), Thomas B. Gabrielson (Penn State Univ., State College, PA), and B. J. Merchant (Sandia National Labs., Albuquerque, NM)

The Penn State Applied Research Laboratory (PSU) and Sandia National Laboratories (SNL) have implemented a reciprocity-based primary calibration technique in SNL's infrasound calibration chamber. The goal in this work is primary calibration of microbarometers from 0.01 to 10.0 Hz. The National Center for Physical Acoustics at the University of Mississippi designed and built the chamber, which SNL has modified and currently operates. This chamber incorporates two moving coil loudspeakers capable of operating in receive and transmit mode. PSU developed the calibration technique and the required electronics to take advantage of these two reciprocal loudspeakers, while SNL installed this hardware and made the required test measurements. Because the chamber is large (1400 l), the chamber volume dominates the acoustical admittance thereby reducing the dependence of the calibration on the physical characteristics of the loudspeakers and reducing calibration uncertainty. However, laboratory tests have found increased uncertainty at very low frequencies ($<$ 0.05 Hz). This is due to reduced loudspeaker response and increasing noise within the chamber at these frequencies. This talk will discuss the large-chamber reciprocity method, important considerations for reducing uncertainty in this implementation, and limitations of this method.

2:40

1pPAb6. Infrasound source localization and correlation with seismic sources. Keehoon Kim (Geophysical Monitoring Program, Lawrence Livermore National Lab., 7000 East Ave., L-103, Livermore, CA 94550, kim84@llnl.gov)

Infrasound is widely used as a geophysical monitoring tool to detect and locate events of interest. Direct infrasound from atmospheric pressure disturbances can propagate long distances, allowing for reliable source localization. Infrasound can also be generated by underground sources which generally produce seismic waves in the solid Earth. Atmospheric pressures are coupled with ground motions induced by underground explosions or earthquakes and propagate in the atmosphere as infrasound. This ground-coupled infrasound can be used to locate the sources of atmospheric pressure disturbances and complement seismic observation to improve seismic source locations. It is well known that near-surface seismic sources are highly correlated with the infrasound source locations, but the correlation variance depending on source depths has not been explored extensively. In this study, we compare seismic and infrasound source locations of underground events detected by local and regional network in Utah, USA. Infrasound source locations are determined by back-azimuths and travel times on infrasound arrays and compared with seismic event catalogs. This systematic comparison for many seismic events will help us to understand statistical properties of seismic and acoustic source correlation.

3:00–3:15 Break

Contributed Papers

3:15

1pPAb7. Finite element modeling of infrasound resonance in an underground tunnel structure. Nathan Downey (Sandia National Labs., PO Box 5800, MS 0404, Albuquerque, NM 87185-0404, njdowne@sandia.gov)

Recent observations of acoustic/infrasound signals emanating from an underground mine in central Utah show a spectral signature consisting of several dominant harmonics whose amplitude decay rate increases with frequency. This spectral character is consistent with a model in which the observed signals are generated from acoustic resonance within the mine tunnels. I present models of these signals computed using a discontinuous-Galerkin finite element formulation. The mesh geometry of these models is based on the known geometry of the mine tunnels obtained using traditional survey methods. Explosion source locations in the mine are known from previous analysis of seismic signals and operator logs and are associated with observations of infrasound resonance at multiple far-field stations. I explore the sensitivity of the modeled signals to changes in the source spectrum, boundary conditions within in the mine and perturbations to tunnel geometry. I discuss the possible application of my results to volcanic monitoring, industrial monitoring of underground structures, and the possible determination of underground tunnel geometries using far-field observations of acoustic resonance.

3:30

1pPAb8. Abstract withdrawn.

3:45

1pPAb9. Infrasound analysis of the 2018-Dec-18 49 kiloton Bering Sea bolide: Misassociations and celerity-range models. Bethany Grant (AWE Blacknest, Reading RG7 4RS, United Kingdom, bgrant@blacknest.gov.uk), Alexandra Nippress (AWE Blacknest, Reading, United Kingdom), and David N. Green (AWE Blacknest, Brimpton, United Kingdom)

Automatic detections at the International Data Centre (IDC) of the Comprehensive Nuclear-Test-Ban Treaty Organisation (CTBTO) are often misassociated. This occasionally leads to single events being split into multiple events. Infrasound signals generated from a large bolide over the Bering Sea on 2018-Dec-18 were recorded on more than 15 of the CTBTO International Monitoring System (IMS) stations. The bolide was also reported by the Center for Near Earth Object Studies (http://cneos.jpl.nasa.gov/fireballs) with a calculated total impact energy value of 49 kt. Signals from this bolide were automatically detected and associated at the IDC, into two different events, with the origin time differing by approximately 7.5 minutes and location by ∼125 km. To investigate why detections from this bolide event were misassociated in the automatic bulletins, we analyze infrasound observations from the IMS infrasound arrays and compare our arrival time observations to the automatic detections from the IDC. We use the Bayesian Infrasonic Source Localisation method and employ different celerity-range models, to investigate signal association for this event. Through comparison with the associations based on IDC detections, we aim to improve our understanding of the causes of misassociations and identify potential improvements to future event association algorithms.

4:00

1pPAb10. Atmospheric effects on the propagation of sonic booms utilizing ray tracing methods. Jonathan Bouza (Phys., Farmingdale State College, 2350 NY-110, Farmingdale, NY 11735, bouzj13@farmingdale.edu) and Kimberly A. Riegel (Phys., Farmingdale State College, Farmingdale, NY)

The aviation industry's push towards a return of supersonic flight has some hurdles in the United States due to the regulation restricting overland supersonic use of civil aircraft without prior authorization. This regulation was put in place in response to the public's concerns over property damage from the boom front as well as the audible nuisance complaints in the path of the boom carpet. The physical behavior of the boom is understood well enough that NASA is in the testing stage of the X-59 QueSST to test the feasibility of overland flight. The boom characteristics are affected by the temperature, direction, and wind speed of the atmosphere as the signature propagates through the atmosphere. Our research plan is to implement atmospheric profiles from NCAR CFSv2 into a ray tracing Python code in order to model the behavior of the boom front in urban areas through different atmospheres. Several atmospheric conditions will be modeled and the resulting signatures around large structures will be compared.

4:15

1pPAb11. Explosive yield determination using infrasonic signal power estimates and propagation corrections from numerical modelling. David N. Green (AWE Blacknest, AWE Blacknest, Brimpton RG7 4RS, United Kingdom, dgreen@blacknest.gov.uk) and Roger M. Waxler (Univ. of MS, University, MS)

Infrasonic signal power estimated from stratospheric returns can be related to near-field acoustic power generated by an explosion. This requires propagation losses to be corrected for; one method for estimating such losses is to undertake numerical acoustic propagation modelling using state-of-the-art atmospheric specifications. Earlier work, focused on a small set of signals from well characterised explosive trials, identified that an incoherent modal sum was a promising method for calculating the source-to-receiver transmission loss. We extend the initial studies by applying this method to a wider set of explosively generated signals and include both tropospheric and stratospheric returns in the analysis. The larger dataset allows a preliminary assessment to be made of the uncertainties associated with yield determination using infrasonic power estimates.

4:30

1pPAb12. What else can we do with auxiliary parameters in ray tracing? How about back projection for localization? Philip S. Blom (Earth & Environ. Sci., Los Alamos National Lab., PO Box 1663, M/S F665, Los Alamos, NM 87545, pblom@lanl.gov)

Auxiliary parameters describing variations in ray path geometry with respect to initial launch angles have been leveraged in computing the Jacobian determinant needed to solve the transport equation as well as to build a Levenberg–Marquardt algorithm for identification of source-receiver paths (termed eigenrays) in a 3D inhomogeneous moving atmosphere. Building on these applications, recent investigations have demonstrated that these parameters can be computed along back projected ray paths from infrasonic detections to improve Bayesian localization capabilities. An overview of the auxiliary parameters as introduced in previous work will be provided along with a summary of current localization development. Example applications of the method will be presented and compared with existing Bayesian infrasonic localization methods using more general propagation models.

4:45

1pPAb13. The effect of the balloon on balloon-borne infrasound measurements. Oleg A. Godin (Phys. Dept., Naval Postgrad. School, Phys. Dept., Naval Postgrad. School, Monterey, CA 93943, oagodin@nps.edu)

Deploying acoustic sensors on free-flying, long-living balloons helps to reach the areas not accessible with the traditional ground-based sensors, reduce flow noise, and improve characterization of various infrasound sources. In particular, instrumented balloons can potentially increase the infrasonic detection range and the early warning lead time for natural hazards, such as tornadoes and avalanches. When assessing the capabilities of balloon-borne infrasonic sensors and interpreting the measurements, it is important to recognize that the balloon inevitably distorts both signals and the ambient infrasound field by scattering the incoming sound. Measurement distortions due to a nearby compact scatterer prove to be rather different from the well-understood effect of a rigid boundary on ground-based sensors. Using the recently developed theory of sound scattering by thin, prestressed elastic shells [O. A. Godin, J. Acoust. Soc. Am. **154**, 3223–3236 (2023)], this paper quantifies the effects of hot-air and helium balloons on acoustic pressure and oscillatory velocity. It is found that balloon-borne vector sensors are more susceptible to the distortions than pressure sensors, leading to large differences between the apparent and true source bearing and directionality. Possible approaches to compensate for the distortions and retrieve the free-field acoustic quantities will be discussed.

5:00

1pPAb14. Seismometers as infrasound sensors. Richard D. Costley (Geotechnical and Structures Lab., U.S. Army ERDC, 3909 Halls Ferry Rd., Vicksburg, MS 39180, richard.d.costley@usace.army.mil), Sarah McComas (Geotechnical and Structures Lab., U.S. Army ERDC, Vicksburg, MS), Christopher Simpson (Huffington Dept. of Earth Sci., Southern Methodist Univ., Dallas, TX), Chris Hayward (Huffington Dept. of Earth Sci., Southern Methodist Univ., Dallas, TX), and Mihan McKenna (Geotechnical and Structures Lab., U.S. Army ERDC, Vicksburg, MS)

An experiment was conducted in West-Central Mississippi in which five explosive charges were detonated. The TNT equivalent sizes of the charges ranged from 0.57 to 10.91 kg (1.25 to 24 lb). Among the arrays of sensors deployed, seismometers were deployed near microphones at distances of 0.5, 2.1, and 8.4 km from the source. The blast wave at these distances had decayed in amplitude to an acoustic wave. The coherence between the seismometer and microphone signals showed that the seismometer provided reasonable representation of the acoustic wave over limited frequency bands; however, these bands changed between sensor locations. In addition, two 3-component seismometers were deployed near each other 8.4 km from the source. These seismometers were of different types, one having a resonance frequency of 1 Hz and the other at 4.5 Hz. The signals from the horizontal components of these seismometers were analyzed to determine their

effectiveness as vector sensors. The results showed that the back-azimuth determined from the seismometers agreed reasonably well with ground truth for the first arrival of the acoustic wavefront; however, the results degraded as the trailing part of the wavefront passed. Permission to publish was granted by the Director, Geotechnical and Structures Laboratory.

5:15

1pPAb15. Infrasound propagation with deep neural operators. Christophe Millet (CEA, CEA, DAM, DIF, Bruyères-le-Châtel 91297, France, christophe.millet@cea.fr), Elodie Noele, and Fanny Lehmann (CEA, Arpajon, France)

Wave propagation modeling has played a crucial role in various applications of infrasound. However, despite significant progress in recent years, solving the acoustic wave equation numerically still presents practical challenges, particularly for randomly layered media. On the other hand, while machine learning has emerged as a promising alternative, training deep neural networks requires a tremendous amount of data, which can be challenging and expensive to obtain. In this work, we combine a projection-based reduced-order model (ROM) of the wave equation with a Fourier neural operator (FNO) to learn mappings between atmospheric specifications and ground-based waveforms. The ROM is utilized to generate a comprehensive database of waveforms, taking the ECMWF ensemble reanalysis as input. Unlike traditional neural networks that are restricted to the prediction of solutions in a predefined configuration, it is shown that the FNO captures the leading order propagation operator as well as important properties of the dispersion relation. It is also demonstrated that the FNO predicts subtle effects in the waveforms that can be unambiguously associated with small-scale heterogeneities, such as turbulence and gravity waves. The feasibility of this approach is illustrated using repetitive infrasound events over the last 20 years.

MONDAY AFTERNOON, 13 MAY 2024 ROOM 208, 1:00 P.M. TO 3:50 P.M.

Session 1pPP

Psychological and Physiological Acoustics: Toward More Inclusive Research Practices in P&P I

Monita Chatterjee, Cochair
Center for Hearing Res., Boys Town National Research Hospital, 555 N 30th St., Omaha, NE 68131

Peggy Nelson, Cochair
Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55455

Chair's Introduction—1:00

Invited Papers

1:05

1pPP1. Acknowledging the Social Bases of Speech Intelligibility: A Key to Improving Equity in Psychological and Physiological Acoustics. Benjamin Munson (Speech-Language-Hearing Sci., Univ. of Minnesota, 115 Shevlin Hall, Minneapolis, MN, munso005@umn.edu)

A major endeavor in psychological and physiological (P&P) acoustics is understanding the causes and consequences of hearing loss. One of the most common complaints of individuals with hearing loss is difficulty perceiving speech. Hence, many studies in P&P compare psychophysical or physiological measures of auditory perception with performance on speech perception tasks like sentence intelligibility. Sentence intelligibility tasks often use read speech samples produced by a small number of demographically unspecified talkers, with only limited exceptions (i.e., McCloy, Wright, & Souza, 2015; Wright & Souza, 2012). This talk reviews findings from the author's and others' research groups showing that sentence intelligibility varies as a function of talkers' social identities, including a talker's actual or perceived racial identity (Babel and Russell, 2015; McGowan, 2015; Tripp, Lyons, and Munson, 2022). These findings reinforce recent calls for P&P to revise the intelligibility measures used to characterize the consequences of hearing loss (i.e., Beechey, 2022a, 2022b). In particular, this talk argues for a collective research program developing entirely new speech intelligibility measures that reflect diverse ways of speaking, and the diverse functions of spoken language. [Work funded by NIH grant R21 DC018070.]

1:30

1pPP2. At the table or on the menu: Engaging the community in research. Yolanda F. Holt (Commun. Sci. and Disord., East Carolina Univ., 300 Moye Blvd. 3310-X HSB, MS 668, Greenville, NC 27834, holty@ecu.edu)

Community engaged or participatory research is a theoretical and practical framework of collaboration between a community and a research team to answer a question of mutual interest. The experimental factors from identification of the research question, identification of the group, and identification of the objects of measurement through the dissemination of the results are ideally a collaborative and iterative process between the community and the research team. This research method can be challenging to orchestrate. However, without ongoing conversations throughout the experimental process between the community of practice and the research team the data gathered may have limited practical use to the community, may misinterpret practices within the community or may inadvertently denigrate typical behavior in the community as dysfunctional or disordered. The presented work describes one approach to engage communities as active participants in the research process from problem identification through dissemination of the results. The methods used for engaging community participants in both a language and an acoustic phonetic study will be shared. Results will be discussed in terms of experimental outcomes, community perspective, and community benefit.

1:55

1pPP3. Incorporating minority perspectives into study design. Erin O'Neill (CATSS/GN Adv. Sci., Minneapolis, MN), Walter Yueh-Hsun Wu (Optometry, The Ohio State Univ., Columbus, OH), and Peggy Nelson (Ctr. for Appl./Trans. Sensory Sci./CATSS, Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, peggynelson@umn.edu)

The vast majority of auditory research in the field of hearing loss is conducted by scientists and clinicians with normal hearing sensitivity. As a result, well-intentioned study and survey designs often miss the mark in representing the disabled experience and quantifying outcomes that are meaningful to those with sensory loss. This underscores the importance of supporting the development and inclusion of scientists with sensory loss in academia. In this talk, we will give examples of studies from vision and hearing loss that were designed by disabled researchers. We will focus on how they differ from those designed by non-disabled scientists in the same field. These examples will highlight how the lived experience of scientists with sensory loss influences the types of research questions asked and metrics used to answer these questions. We will also share methods and data from studies that have successfully incorporated the perspectives of hearing- and vision-impaired individuals into their design process.

2:20–2:35 Break

2:35

1pPP4. Assessing perceptual performance of non-native English speakers. Sandra Gordon-Salant (Hearing and Speech Sci., Univ. of Maryland, 7251 Preinkert Dr., 0100 Lefrak Hall, College Park, MD 20742, sgsalant@umd.edu)

About 20% of the U.S. population, many of whom are foreign born, speaks a language other than English in the home. Non-native English speakers are known to have difficulty understanding spoken English, especially in noise. For this reason, they are often excluded from psychophysical perceptual experiments. However, the perceptual abilities of these individuals are important to characterize, particularly because solutions to their unique communication difficulties are important to establish. This presentation will review perceptual data from a series of experiments that examined speech recognition in degraded listening conditions and psychophysical performance on basic auditory processing tasks. Listeners in all experiments included young adults with normal hearing whose native language was either English or Spanish. Some of the experiments also included older adults with normal hearing or hearing impairment who learned English as a second language. The results show that non-native speakers of English have considerable difficulty on all complex speech recognition tasks, which is exacerbated with advanced age. Additionally, non-native speakers of English do not perform as well as native speakers of English on some basic psychoacoustic measures. These results have implications for evaluating performance of non-native speakers in the lab and in the clinical setting.

3:00

1pPP5. Inclusion of individuals with Down syndrome in auditory research. Lori Leibold (Hearing Res., Boys Town National Res. Hospital, 555 N 30th St., Omaha, NE 68131, lori.leibold@boystown.org), Heather Porter (Hearing Res., Boys Town National Res. Hospital, Omaha, NE), and Emily Buss (Univ. of North Carolina, Chapel Hill, NC)

This talk will provide an overview of the procedures, challenges, and key findings of Project INCLUDE at Boys Town National Research Hospital. Funded by a trans-NIH initiative to investigate health conditions that affect individuals living with Down syndrome, the goal of Project INCLUDE at Boys Town National Research Hospital is to characterize how speech, language, and hearing develop across the lifespan. Individuals living with Down syndrome have been inadequately represented in auditory research, which has likely contributed to health care disparities. The exclusion of individuals with Down syndrome in prior research is particularly concerning given the high prevalence of hearing loss and otologic disease observed in this population. Results from an ongoing experiment investigating masked speech recognition outcomes will be presented. Participants are school-age children and young adults with Down syndrome and age-matched participants who are neurotypical. Data collection includes clinical audiological measures and standardized assessments of speech, language, and executive function, facilitating comparisons with other cohorts affiliated with the NIH-wide initiative. Recommendations for recruitment, details on the formation of a community advisory board, and community-based testing initiatives will be discussed.

3:25

1pPP6. Auditory processing in neurodiverse children. Bonnie K. Lau (Univ. of Washington, 1715 NE Columbia Rd., Box 357988, Seattle, WA 98195, blau@uw.edu), Tanya St. John, Annette Estes, and Stephen Dager (Univ. of Washington, Seattle, WA)

Many neurodiverse individuals experience auditory processing differences including hyper- or hyposensitivity to sound, attraction or aversions to sound, and difficulty listening under noisy conditions. However, the origins of these auditory symptoms are not well understood. In this study, we tested 7-to-10-year-old autistic children and age and sex-matched neurotypical comparison participants. To simulate a realistic classroom situation where many people are often speaking simultaneously, we obtained neural and behavioral measures of speech perception in both quiet and noise conditions. Using electroencephalography, we recorded neural responses to naturalistic, continuous speech to assess the cortical encoding of the speech envelope. We also obtained behavioral multitalker speech perception thresholds and estimates of spatial release from masking, a binaural hearing phenomenon in which speech perception improves when distracting speakers are spatially separated from the target speaker. Our preliminary results from both neural and behavioral measures suggest that the autistic group shows worse speech perception in noise and less spatial release from masking than the neurotypical group. These findings suggest that autistic children may benefit from environments with reduced noise to facilitate speech perception. These findings also warrant further investigation into speech perception under real-world conditions and the neural mechanisms underlying sound processing in autistic children.

MONDAY AFTERNOON, 13 MAY 2024 ROOM 203, 1:00 P.M. TO 3:10 P.M.

Session 1pSA

Structural Acoustics and Vibration, Education in Acoustics and Physical Acoustics: Mistakes and Lessons Learned in Structural Acoustics and Vibrations

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

A. J. Lawrence, Cochair
Walker Dept. of Mech. Eng., Univ. of Texas at Austin, 204 E. Dean Keeton St., Stop C2200, Austin, TX 78712-1591

Chair's Introduction—1:00

Invited Papers

1:05

1pSA1. The collapse of the Tacoma Narrows bridge– Why? William Unruh (Dept. Phys. and Astronomy (UBC) and IQSE (A&M), Univ. of BC and Texas A&M Univ., 6224 Agricultural Rd., Vancouver, BC V6T1Z1, Canada, unruh@physics.ubc.ca)

In 1940, a new bridge over the Tacoma Narrows south of Seattle went into violent oscillation for over an hour and then collapsed, with the loss of one dog's life. The bridge satisfied all of the engineering standards but the first windy day doomed it. What happened? A variety of explanations have been advanced, but the cause turned out to be quite simple, though unexpected. (It was not resonance.) There are many similarities between it and the fundamental playing behaviour of wind instruments, a positive feedback mechanism coupling the bridge to the wind. (It was not resonance.) In 2006, Daniel Green and I, using a finite element program of G. Morgenthal, showed what triggered this positive feedback mechanism in the case of the bridge, and this talk will discuss the features of our explanation

1:25

1pSA2. Unexpected results from structural acoustic optimizations. James G. McDaniel (Dept. of Mech. Eng., Boston Univ., Boston, MA 02215, jgm@bu.edu)

Optimization algorithms for improving the sound and/or vibration characteristics of something, such as a consumer product, have become ubiquitous. However, using them may occasionally give unexpected and undesirable results. The story of the Herdsman from Aesop's Fables ends with the following relevant moral: "Be careful what you wish for; your wish may be granted." While optimization tools effectively minimize an objective function subject to constraints, they may do so in unexpected and undesirable ways. Optimizing with respect to one physical phenomenon may damage other physical phenomena. This presentation reviews optimization problems related to the steady-state response of a vibrating system due to a sinusoidal excitation. To reduce global vibration, the system parameters are varied to maximize the power absorbed by damping elements. The wish is granted. The optimization process maximizes power absorbed by increasing the power that enters the system from the excitation. The result is an unexpected increase in global vibration. This result, as well as similar results and remedies from the controls community, are discussed. [Work supported by ONR under Grant N00014-22-1-2785.]

1:45

1pSA3. Transforming discrete movements of a sonar array into smooth vibration of a plate: Constant force input versus constant displacement input. Brian E. Anderson (Dept. of Phys. & Astronomy, Brigham Young Univ., N245 ESC, Provo, UT 84602, bea@byu.edu), W. Jack Hughes (Appl. Res. Lab., Penn State Univ., State College, PA), and Stephen Hambric (Hambric Acoust., LLC, Asheville, NC)

Numerical modeling of systems is an efficient way to explore new ideas, but an understanding of modeling assumptions as they relate to the actual physics of the system is critical. As a graduate student, I inherited a potentially revolutionary project involving sonar arrays. The idea was that grating lobes in the sound radiation from arrays could be eliminated by placing a plate between the array and the water. Grating lobes result from unintended, aliased superposition of traveling waves when discrete transducers attempt to create a single traveling wave in the array's plane. The model initially assumed the transducers provided a constant displacement input to the plate's vibration and the plate, thus, transformed the discrete traveling wave into a smooth continuous wave, resulting in elimination of grating lobes. The problem was that the plate's impedance was higher than the transducers' impedance and, thus, a constant force input was a better approximation to use in the model. When accurate assumptions were used, the grating lobes returned. A new idea was conceived to make something useful out of the original idea, though it required yet undeveloped materials. Although unexpected and disappointing at the time, "negative discoveries" can sometimes lead to positive ones.

2:05

1pSA4. Doomed from the start, understanding the importance of boundary conditions in modal testing. Peter Kerrian (ATA Eng. Inc., 13290 Evening Creek Dr. S, San Diego, CA 92128, peter.kerrian@ata-e.com) and Teresa Kinney (National Aeronautics and Space Administration, Kennedy Space Ctr., FL)

Modal survey tests are commonly performed in the aerospace industry to validate the structural dynamics behavior of the finite element model used in the flight loads predictions for both hardware structural capability and expected crew loads. During the planning phase, great effort is spent on the selection of accelerometer locations to properly capture the modes and satisfy orthogonality conditions. The same due diligence is not always spent on the design of the test's setup and boundary conditions. Logistical and programmatic concerns may influence the technical approach taken, which, unfortunately, may result in test modes that are strongly coupled with boundary condition fixturing. The purpose of the paper is to present case studies of modal survey tests that were performed where the boundary condition influenced the test results. Additionally, best practices will be presented.

Contributed Papers

2:25

1pSA5. Modeling of construction vibration impacts in transit project applications. Nicholas Tam (Aercoustics Eng. Ltd., 1004 Middlegate Rd., Markham, ON L6C3A3, Canada, nicholast@aercoustics.com)

This presentation explores some of the challenges related to predicting the vibration impact from construction activities associated with new mass transit infrastructure. Pre-construction assessment methods are examined and compared against as-measured construction vibration monitoring data. Practical considerations and limitations of construction vibration monitoring are also discussed, citing examples, and insight from the ongoing projects, especially in the Greater Toronto Area.

2:40

1pSA6. Relationships and statistics between various vibration metrics, derived from a large set of measured train pass-bys. Vincent Jurdic (Arup Canada Inc., 1 Pl. Ville marie, Unit 3270, Montréal, QC H3B 3Y2, Canada, vincent.jurdic@arup.com) and Joseph Digerness (Arup US, Inc., New York, NY)

The impacts of groundborne vibration (GBV) and sound (GBS) induced by railway infrastructure are assessed across the world through different metrics. Many national and international standards can be found for assessing GBV, based either on acceleration (UK, Spain, etc.) or velocity metrics (Germany, USA, etc.). Various frequency and/or time-weightings can be

used, and different quantities (running RMS or highest levels) are also considered. Direct comparison between the various assessment criteria and measured metrics is, therefore, difficult. Although no international standard defines GBS criteria, many guidelines suggest such a criterion through a relationship between GBS levels and vibration velocity levels on room surfaces. Different quantities are often used, even for neighboring projects: London's Crossrail (now Elizabeth Line) and Northern Line extension impact assessments both used maximum vibration velocity levels but with different time weighting (slow and fast, respectively). Arup has amassed a large dataset of vibration measurements through its involvement in many railway schemes over the years. For each of the thousands of train pass-bys, measured at different distances, infrastructure types, operation, and ground conditions, the most common European and North American GBS and GBV metrics are derived and compared to each other to develop statistical relationships and associated uncertainties.

2:55

1pSA7. Learning from failure: An analysis of a flawed nonlinear acoustic metamaterial design. Michael B. Muhlestein (ERDC-CRREL, 72 Lyme Rd., Hanover, NH 03755, Michael.B.Muhlestein@usace.army.mil), Kyle G. Dunn (ERDC-CRREL, Hanover, NH), Gordon M. Ochi (U.S. Army ERDC, Champaign, IL), and Michelle E. Swearingen (Construction Eng. Res. Lab., U.S. Army ERDC, Champaign, IL)

Acoustic metamaterials (AMMs) are structured systems designed to exhibit specific, often exotic, effective material properties for acoustic wave motion. Most research on AMMs prioritize linear wave behavior, but a growing body of work has focused on nonlinear phenomena. While the primary approach to enhancing the nonlinear response of a system has been to place highly nonlinear inclusions in the system (such as bubbles, cracks, or buckling beams), an alternative approach is to control the linear material properties to enhance the relative strength of the nonlinearity. For example, one may reduce the shock formation distance of a system by reducing the effective mass density and wave speed even if the coefficient of nonlinearity is left unchanged. However, due to the nature of nonlinear waves, developing a practical design to accomplish this control can be challenging. We recently designed and built an experimental nonlinear AMM and found it to be fundamentally flawed. In this paper we discuss the thought process that led to the flawed design and present the lessons learned from an unsuccessful experiment.

MONDAY AFTERNOON, 13 MAY 2024 — ROOM 214, 1:00 P.M. TO 4:00 P.M.

Session 1pSC

Speech Communication: Speech Perception Poster Session I

Silvia Murgia, Chair

Speech and Hearing Sci., Univ. of Illinois - Urbana Champaign, 901 South Sixth St., Champaign, IL 61820

All posters will be on display and all authors will be at their posters from 1:00 p.m. to 4:00 p.m.

Contributed Papers

1pSC1. Closure duration versus f0 perturbation as a cue to underlying stops for the American English intervocalic flap. Janalyn A. Miklas (English, George Mason Univ., 4400 University Dr., Fairfax, VA 22030, jmiklas@gmu.edu) and Matthew C. Kelley (English, George Mason Univ., Fairfax, VA)

Researchers found that there are five main cues for distinguishing the voicing of these intervocalic plosives, preceding vowel duration, following vowel duration, closure duration, semantic context, and f0 perturbation, also known as consonant intrinsic f0. In American English, when /t/ and /d/ occur intervocalically, they are realized as a voiced alveolar flap [ɾ] (e.g., [ɹa͡ɪɾɪŋ] *writing*/*riding*, [liɾɚ] *liter*/*leader*). The presence of these cues indicates incomplete neutralization of these forms. To address a current gap in the literature, the present study focused on American English native speakers' perception and use of CF0 and closure duration as independent and combined informative cues for deciding voicing contrast of /t/ and /d/ when neutralized by the flap. Participants engaged in a self-administered online forced-judgement task of a pseudoword in 30 combinations of CF0 and flap closure duration to indicate the underlying representation (/hɑtɑ/ or /hɑdɑ/) of the surface production [hɑɾɑ] they perceived. This study observed that participants are more likely to select /hɑtɑ/ when the CF0 is higher and /hɑdɑ/ elsewhere. These modest findings suggest that CF0 is a useful cue for voicing distinction. The results suggest that models of spoken word recognition and speech perception ought to include CF0 as a cue.

1pSC2. The interplay of speech categorization and meta-phonological skills on reading in children. Ethan Kutlu (Linguist., Univ. of Iowa, 459 Phillips Hall, Iowa City, IA 52242, ethankutlu@gmail.com), Jamie Klein-Packard, and Bob McMurray (Psychol. and Brain Sci., Univ. of Iowa, Iowa City, IA)

Speech perception was thought to emerge early developmentally and to be stable at the onset of reading instruction. Additionally, meta-phonological skills (e.g., phoneme awareness) were regarded as a primary predictor of early reading. A simple model would then predict that early speech perception shapes later meta-phonological skills, which lead to reading development. However, the link between speech perception and meta-phonological

skills was never established. Additionally, we now know that speech perception develops through early adolescence (McMurray *et al.*, 2018; Hazan, 2000), raising the possibility that speech is not stable at reading onset, and may be impacted by reading instruction. We examined children's (age 6–11, $n = 259$) speech categorization with the Visual Analogue Scaling Task, and language, meta-phonology, and reading abilities using an SEM on a battery of language assessments. We found that school-aged children's speech categorization improves with age ($p < 0.001$), but this was mediated by overall reading ability ($p = 0.029$) and not language ($p > 0.1$) suggesting reading development may promote better speech perception. Importantly, meta-phonological skills and speech perception were not strongly correlated, but each uniquely explained variation in reading. This suggests that these two constructs play unique roles in reading development. We will discuss implications for speech perception and reading studies.

1pSC3. The phonetically balanced kindergarten assessment: An alternative evaluation method utilizing machine learning technology. Ayden M. Cauchi (The Hospital for Sick Children, 2647 Fonthill Dr., Oakville, ON L6J 6Y8, Canada, ayden@modimages.com), Jaina Negandhi, Micheal Cornacchia, and Karen A. Gordon (The Hospital for Sick Children, Toronto, ON, Canada)

This study investigates the use case of machine learning (ML) technologies to score the Phonetically Balanced Kindergarten (PBK) word test. The PBK, developed to test pediatric speech perception, is also used to monitor outcomes of cochlear implantation (CI) in children. Many factors contribute to high PBK score variability in CI users, but effects of scoring by human listeners are unclear. In this study, an alternate ML scoring method was developed. The Ursa classifier (Speechmatics, 2023) was used to recognize 100 PBK words spoken by 12 adults with normal hearing, speech, and language. The spoken words ($n = 1200$) were manipulated in six conditions (first and last phoneme deletion; high frequency filtering at 1, 2, 4, and 6 kHz), yielding 8400 stimuli. The classifier correctly recognized most of the unaltered words (true positive rate of 0.90; peak distribution density at 96%) and provided a null or incorrect response to altered stimuli as expected; most frequently for the first phoneme deletion and 1 kHz filtered conditions. Inter-class correlations showed comparable results between Ursa and 7 normal hearing human scorers (Cohens Kappa: 0.39). Thus, ML tools show promise in the scoring of the PBK; increasing the accuracy, efficiency, and reliability of this clinical assessment.

1pSC4. The effects of delay on task-based interactive conversation. Benjamin Masters (Systems Design Eng., Univ. of Waterloo, 200 University Ave. W, Waterloo, ON N2L 3G1, Canada, bpmasters@uwaterloo.ca) and Ewen MacDonald (Systems Design Eng., Univ. of Waterloo, Waterloo, ON, Canada)

Previous studies have found that, in the presence of noise, the distribution of floor transfer offsets (FTO) broadens and shifts to the right, and it has been suggested that this is due to increased listening effort. The present study investigates if speakers are sensitive to the timing of the start of a partner's turn during interactive conversation. By manipulating the delay on the channel between two talkers on a turn-by-turn basis, the floor transfer offset (FTO) distribution can be modified to simulate what has been observed in more challenging conversational settings. Talkers were seated in different rooms but could communicate via headset microphones and headphones with gains fixed such that the levels would simulate the two talkers sitting in the same room. Conversations where pairs of talkers solved the DiapixUK spot-the-difference task were conducted in both quiet and noise, both with and without added delay. Measures of speech production (e.g., speech level, articulation rate, etc.) and conversation behavior (FTO, turn length, etc.) are compared across the conditions.

1pSC5. Sound insights: The potential of hearables for early detection and continuous monitoring of Alzheimer's disease. Miriam Boutros (Elec. Eng., École de Technologie Supérieure, 1100 Notre-Dame St. W, Montreal, QC H3C 1K3, Canada, miriam.boutros.1@ens.etsmtl.ca), Christopher Niemczak (Geisel School of Medicine, Dartmouth College, Hanover, NH), and Rachel Bouserhal (Elec. Eng., École de Technologie Supérieure, Montréal, QC, Canada)

Early identification using non-invasive biological markers is crucial for detecting and treating Alzheimer's disease (AD). Hearables are intra-aural wearable devices equipped with in-ear microphones capable of capturing inner body signals, including heartbeat, breathing, and blinking. These non-verbal audio events give valuable insight into the individual's physiological state, offering a unique method to detect and monitor subtle changes in the body, which holds the potential for early disease detection. Central auditory (CA) tests, such as understanding speech in background noise, have shown promise in tracking AD's effects on the central nervous system. They also provide sensitive, quantitative, and repeatable assessments over time. Therefore, this study will assess physiological signals captured by a hearable during central auditory assessment. Thirty-five participants with AD and mild cognitive impairment will be recruited, in addition to age-matched control participants. Hearables will be used to capture physiological data and perform CA tests such as hearing in noise test, triple digits test, dichotic digits test, and dichotic sentence identification test. Moreover, a picture description task, where participants will have to produce coherent speech that holds potential for tracking the decline in cognitive abilities, will be administered. The data collected will be used to build an open-access database serving as the foundation of future research on early detection and monitoring of AD using hearables.

1pSC6. Hearing gender in variation: The role of gendered expectations in perceptions of sociophonetic covariation. Stella Takvoryan (Univ. of Kentucky, 1550 Trent Blvd, 1612, Lexington, KY 40515, stakvoryan@uky.edu)

Social information mediates phoneme identification, which has been explored robustly with the phonemes [ʃ] and [s] (e.g., Strand and Johnson, 1996; Bouavichith *et al.*, 2019). However, existing research has not focused on understanding how visual information can impact perceptions of covariation. This study expands upon existing work (Laycock and McGowan, in press) by examining how gendered visual information impacts listeners' perceptions of sociophonetic covariation. Participants ($n = 40$) saw one of three photographs: a stereotypically feminine face, a less feminine face, or a blank profile picture. Participants heard 175 versions of the carrier phrase "the pale shack/sack" and were asked to identify the last word, which was manipulated in a seven-step continuum. Pitch and voice onset time varied with fricative center of gravity during the target word. Listeners in the more-feminine condition identified *sack* earlier in the continuum across pitch variations but later in the continuum when the carrier phrase was lower pitch; listeners in the less-feminine condition heard *sack* later than those in the more-feminine condition but earlier than listeners in the profile condition. This study has implications for the role of listeners' expectations in perceiving sociophonetic covariation and usage of social constructs like "femininity" in perception methodologies.

1pSC7. Political affiliation affects spoken language processing. Enengy Schutt (Speech-Language-Hearing Sci., Univ. of Minnesota - Twin Cities, 115 Shevlin Hall, 164 Pillsbury Dr., SE, Minneapolis, MN 55455, schut342@umn.edu), Machaela Campbell (Psych., North Carolina Agricultural and Tech. State Univ., Greensboro, NC), Luis Rivera (Psych., Wabash College, Crawfordsville, IN), Sarah R. Bellavance (Commun. Sci. and Disord., New York Univ., New York, NY), and Susannah V. Levi (Commun. Sci. and Disord., New York Univ., New York, NY)

Listeners typically perform better on intelligibility tasks with native compared to nonnative speakers (e.g., Tsurutani, 2012). Listeners from multilingual, urban locations perform better on intelligibility tasks than those from non-multilingual, rural locations (Kutlu *et al.* 2022). A recent study from our lab found that listeners with more liberal political affiliation also perform better, but the study did not control for a possible relationship between more liberal political affiliation and living in an urban environment.

To further investigate, 54 listeners from an urban location and 27 listeners from rural locations were recruited. Listeners heard sentences from two native (US, India) and two nonnative speakers of English (Korean, French) and typed their responses. Listeners then completed a political-affiliation questionnaire. A linear mixed-effects model with fixed effects for group, speaker, and political-affiliation score, and all interactions were fit to the data. The model revealed a significant interaction between group and political affiliation score. For the urban listeners, more liberal listeners performed better overall than more conservative listeners. For the rural listeners, political affiliation did not reach significance. Taken together, this study demonstrates that political affiliation predicts performance even when exposure (urban versus rural) is controlled.

1pSC8. Integration of semantic and coarticulation cues during spoken language comprehension in adults. Scarlet Wan Yee Li (Dept. of Linguist., Univ. of Ottawa, Hamelin Hall, Rm. 401, 70 Laurier Ave. East, Ottawa, ON K1N 6N5, Canada, wli240@uottawa.ca), Margarethe McDonald (Dept. of Speech-Language-Hearing: Sci. and Disord., Univ. of Kansas, Lawrence, KS), and Tania Zamuner (Dept. of Linguist., Univ. of Ottawa, Ottawa, ON, Canada)

Recent work examining cue integration across levels of linguistic representation has found that listeners can dynamically integrate some of the lower-level and higher-level cues during spoken language comprehension. However, it is still not well understood how the mechanism of cue integration works. This study investigated how adults ($n = 52$) process preceding higher-level semantic cues and later low-level coarticulation cues during spoken language comprehension using an eye-tracking paradigm. Participants were tested on sentences that contained a prime (semantically related or semantically unrelated to the target) and a target which had varying coarticulation cues (matching versus mismatching splicing cues). Participants were presented with two pictures (target and competitor) on a screen. Analyses looked at the proportion of looking to the target during the prime and target time windows. Results demonstrate that adults flexibly use both the preceding semantic cues and later coarticulatory cues once they are available. Our findings also indicate that adults flexibly weighed both the preceding higher-level and later lower-level cues, such that the processing of low-level coarticulatory cue varied depending on the semantic context. We have added an unstudied level of cue (semantic context) to the set of cues that our cognitive system can integrate during language comprehension.

1pSC9. Age-related decline in hearing and emotional prosody processing: A multi-feature oddball study. Yang Zhang (Speech-Language-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, zhang470@umn.edu), Erica Kuntz (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN), and Chieh Kao (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN)

This study investigated how age and hearing loss contributed to the challenges in perceiving emotional prosody in speech. A multi-feature oddball paradigm was employed to assess mismatch negativity (MMN) and P3a responses to three deviant prosodies (happy, angry and sad) against the standard neutral prosody in spoken words. The participants were 22 adults in the age range of 18 to 70 with various degrees of hearing loss as assessed with pure tone audiometry. Linear mixed effects models revealed heightened MMN response to angry voice and strongest P3a response to happy voice among the three deviants. There was a significant positive correlation between age and MMN amplitude for the angry prosody. Significant correlations were also observed between hearing loss and MMN amplitude in response to both angry and happy stimuli. In addition, negative correlations were found between P3a amplitude and hearing loss as well as aging for happy stimuli. However, aging and hearing loss did not significantly affect the processing of sadness in this paradigm. These findings showed category-sensitive age-related decline in emotional prosody processing. The intricate correlations patterns among age, hearing loss, MMN, and P3a responses called for further exploration of broader implications in cognitive aging.

1pSC10. Listening effort for L2 accents: Adults' adaptation to Mandarin accented english. 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), Mel Mallard, and Kristin J. Van Engen (Psychol. and Brain Sci., Washington Univ. in St. Louis, St. Louis, MO)

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 this study, we investigated the relationship between three prosodic measures (relative word duration variance, pitch stability, and pitch range) and changes in listening effort for L1 and L2 speakers over the course of an experiment. Using growth curve analysis, we found significant interactions between each of the three measures and trial number, indicating that prosodic factors influence speech adaptation. However, these interactions were in opposite directions for the L1 and L2 speakers. Thus, adult monolingual English listeners rely on prosodic information to adapt to a particular speaker's voice, but use that information differently based on speaker identity.

1pSC11. Real-time word recognition in heritage speakers: Evidence from the visual world paradigm. Ethan Kutlu (Linguist., Univ. of Iowa, 459 Phillips Hall, Iowa City, IA 52242, ethankutlu@gmail.com), Jacob Boudreau (Linguist., Univ. of Iowa, Iowa City, IA), Samantha Chiu, Jamie Klein-Packard, and Bob McMurray (Psychol. and Brain Sci., Univ. of Iowa, Iowa City, IA)

Up to 20% of children in the US come from multilingual households (Census, 2019). However, current language development models are still mostly based on monolingual input. Heritage speaker (HS) children pose a challenge for current understandings of monolingual language development because their first language (the heritage language, HL) becomes their less dominant language as they age (often when schooling begins), and their second language (the majority language, ML) eventually becomes dominant, despite a delayed start. Here, lexical processing—a key hub in the language system—is targeted as a model system in which to ask how HSs become more automatic language processors in the ML. To do so, the proposed study focuses on spoken word recognition, which is a critical bottleneck in early language development, as the ability to efficiently recognize continuous speech paves the way for higher level language abilities. Successful recognition requires listeners to suppress competitors, such that only the target word remains. We present eye-tracking data from school-aged HSs who have become more dominant in English ($n = 30$). Our findings show that HSs, despite a delayed start in English, reach target words efficiently. We discuss the strategies employed by HS children to resolve competition in their ML.

1pSC12. Heritage speaker children's speech categorization patterns. Ethan Kutlu (Linguist., Univ. of Iowa, 459 Phillips Hall, Iowa City, IA 52242, ethankutlu@gmail.com), Nick Theuerkauf (Linguist., Univ. of Iowa, Iowa City, IA), Samantha Chiu, Jamie Klein-Packard, and Bob McMurray (Psychol. and Brain Sci., Univ. of Iowa, Iowa City, IA)

Bi-/multilingual experiences vary widely due to differing degrees of input frequency, dominancy in languages, and societal preferences for the language(s) used. Heritage bilingualism has received notable attention over the last decade. Heritage bilinguals are those who often use or are exposed to one language at home (the heritage language) that differs from that used outside of the home (the majority language). For a long time, bilingual children's unbalanced input was targeted as a source of deficiency in categorizing speech. However, prior studies used tasks that were not equipped to address variability in speech categorization across listeners (see Apfelbaum *et al.*, 2022). We use the visual analogue scaling (VAS) task which can capture variability in speech categorization across listeners with more sensitivity and which can reveal new dimensions of differences such as the trial-by-trial variability in responding (Kapnoula *et al.*, 2017; Kutlu *et al.*, 2022).

In an ongoing study, we are testing school-aged bilingual children with various heritage language backgrounds (n = 30, 6–11 years). Preliminary findings suggest that heritage speaker children are not deficient in speech categorization, and in fact, show more gradient categorization patterns, suggesting a functional adaptation to increased phonetic variation in their language environment (Kutlu *et al.*, 2022).

1pSC13. The role of gradient speech categorization in accent perception. Ethan Kutlu (Linguist., Univ. of Iowa, 459 Phillips Hall, Iowa City, IA 52242, ethankutlu@gmail.com), Emerson Peters (Linguist., Univ. of Iowa, Iowa City, IA), Samantha Chiu, and Bob McMurray (Psychol. and Brain Sci., Univ. of Iowa, Iowa City, IA)

A key question in speech perception research is how listeners categorize highly varied signals into discrete units. The long-held assumption was that listeners discard variation and focus on the category itself (categorical perception, Liberman *et al.*, 1957). However, listeners often encounter highly varied speech signals (e.g., processing unfamiliar accents). In such cases, discarding variation can be detrimental for listeners and can lead to more processing difficulty. Here, in an ongoing online study (n = 80), we measure English-speaking adult listeners' speech categorization patterns with a continuous measure (the Visual Analogue Scaling Task: Kutlu *et al.*, 2022; Apfelbaum *et al.*, 2022) to see whether they are more susceptible to phonetic variation in their language environments. We index adults' language exposure with an extensive social network survey which quantifies the extent to which they are exposed to varied accents on a regular basis and what those accents are. We then asked participants to recognize spoken sentences from a diverse set of unfamiliar accents. We predict that adults who are more gradient in their speech categorization are more accurate in transcribing sentences with a diverse set of accents.

1pSC14. Prior knowledge benefits older adults' tonal consolidation through talker generalization. Kangdi Liu (The Hong Kong Univ. of Sci. and Technol., Hong Kong, Hong Kong), Yan Feng (Nanjing Univ. of Sci. and Technol., Nanjing, China), and Zhen Qin (The Hong Kong Univ. of Sci. and Technol., Hong Kong, Hong Kong, hmzqin@ust.hk)

Recent research indicated that post-training sleep facilitates memory consolidation and talker generalization in young adults' tone learning. For instance, a nap study showed the newly learned tonal contrast that relates to prior knowledge consolidated more rapidly than that does not after daytime naps. Notably, older adults' declarative memory consolidation is impaired with age-related changes in sleep architecture. The current study examines whether prior (contour-tone) knowledge benefits Mandarin-speaking older adults' Cantonese tone learning. Mandarin employs pitch contours (falling versus rising) to cue tones in signaling word identity. Besides pitch contours, Cantonese also utilizes pitch heights (higher versus lower) in level tones to encode word meanings. In the pilot phase (60 Mandarin-speaking older adults will be recruited with data collection ongoing), participants were trained to learn Cantonese contour-level and level-level contrasts in the evening or the morning. A novel word-object identification task was used in the training, followed by three tone identification tasks: a posttest immediately after training, a 12-h delayed posttest (a trained talker), and another delayed posttest (a novel talker). Aligned with the nap study, preliminary results suggested overnight sleep might enhance seniors' tone-related memory by promoting generalization across talkers in learning contour-level contrast owing to their prior contour-tone knowledge.

1pSC15. Perceptual training facilitates Mandarin tone production for preschoolers with cochlear implants: Evidence from acoustic analysis. Hao Zhang (Ctr. for Clinical Neurolinguistics, School of Foreign Lang. and Lit., Shandong Univ., 5 Hongjialou, Jinan City, Shandong Province, China, Jinan 250100, China, hao.zhang0099@sdu.edu.cn), Lele Xu, Wen Ma (Ctr. for Clinical Neurolinguistics, School of Foreign Lang. and Lit., Shandong Univ., Jinan, China), Hongwei Ding (Shanghai Jiao Tong Univ., Shanghai, China), and Yang Zhang (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN)

This study primarily aimed to investigate the benefits of an established perceptual training in the production of lexical tones for Mandarin-speaking preschoolers with cochlear implants (CIs). Thirty-two pediatric CI recipients were tested in this study. Half of the child participants received five sessions of high variability phonetic training (HVPT) within a period of three weeks, whereas the other half served as control who did not receive the formal training. Production of Mandarin tones was recorded before the provision (pretest) and after (posttest) the completion of the training protocol, which was coded and analyzed acoustically with automatic pitch tracking implemented in the software of *Praat*. Results showed significantly enhanced concave characteristic of dynamic pitch contours for the trained children's Mandarin tones produced in posttest relative to pretest using growth curve analysis. Moreover, a tonal ellipse analysis indicated significantly improved tone differentiability and tone hit rate from pretest to posttest following training. By contrast, no significant changes were observed in the control children between the two test sessions. The findings represent initial acoustic evidence of HVPT-induced benefits in lexical tone production for CI users, which supports the application of this perceptual training protocol to aural rehabilitation practice.

1pSC16. Developmental changes in phonological and semantic competition during spoken word recognition. Tania Zamuner (Linguist., Univ. of Ottawa, Ottawa, ON, Canada), Margarethe McDonald (Speech-Language-Hearing, Univ. of Kansas, Lawrence, KS), and Samaa Salama (Linguist., Univ. of Ottawa, 100 Laurier Ave E, Ottawa, ON K1N 6N7, Canada, ssala053@uottawa.ca)

Spoken word-recognition is a foundational skill in child language development; the organization and activation of phonological and semantic competitors develops which in turn impacts word recognition. One of the factors impacting spoken word recognition is the ability to resolve lexical competition. The current study first examines how the time-course of lexical activation changes through development, and second, whether children attend more to targets when competitors are visually present or not. To date, 25 children between ages 3–7 who began acquiring French from birth participated in a visual world eye-tracking paradigm. Children were presented with 4 images, each representing a different French word. Images were either presented with both a phonological (e.g., baleine) and semantic (e.g., cerise) competitor of the audio stimuli (e.g., banana) or the audio stimuli had no relation to the competitor images (e.g., épée). Primary analysis reveals that older children look more to the target than younger children [$F(2,22) = 3.8$, $p = .04$] and that the presence of competitors does not impact target recognition [$F(1,18) = 0.03$, $p = .87$]. Such findings imply that children improve in their abilities to efficiently recognize words with age but are not affected by the visual presence of lexical competitors.

1pSC17. The effect of HVPT on the categorical perception and production of mandarin tones: Behavioral and electrophysiological measures on learners with non-tonal backgrounds. Bing Cheng (English Dept. & Lang. and Cognit. Neurosci. Lab, School of Foreign Studies, Xi'an Jiao Univ., No. 28 Xianning Rd. (W), Xi'an, Shaanxi, 710049, China, bch@mail.xjtu.edu.cn), Xi Xiang (English Dept. & Lang. and Cognit. Neurosci. Lab, School of Foreign Studies, Xi'an Jiao Univ., Xi'an, China), and Yu Zou (English Dept. & Lang. and Cognit. Neurosci. Lab, School of Foreign Studies, Xi'an Jiao Univ., Xi'an, Shaanxi, China)

Mid-rising (Tone 2) and low-dipping (Tone 3) are considered the most challenging Mandarin Chinese tones for Chinese-as-a-foreign-language (CFL) learners. This study aimed to investigate the impact of high variability phonetic training (HVPT) with infant-directed speech (IDS) features on the categorical perception and production of Tone 2 and Tone 3. A total of 21 CFL learners with non-tonal backgrounds participated in the experiment. Both behavioral data and event-related brain potentials were recorded to assess the effectiveness of the training. The results revealed that CFL learners significantly improved their perception and production of Tone 2 and Tone 3 after the HVPT training. Specifically, CFL learners demonstrated an enhanced ability to identify and discriminate between Tone 2 and Tone 3 after the training. There was also a significant increase in pronunciation accuracy ($p < 0.05$), as assessed by three native Mandarin speakers on recordings of 20 monosyllabic words produced by all participants. Moreover, CFL learners exhibited increased MMN amplitudes at the left electrodes, suggesting enhanced sensitivity to tone categorical perception after the

training. These findings demonstrate that HVPT with IDS features can be a promising method for effective Mandarin tone learning among CFL learners.

1pSC18. Target-masker onset asynchrony modulates linguistic release from masking. Anne J. Olmstead (Commun. Sci. and Disord., The Penn State Univ., 404D Ford Bldg., University Park, PA 16802, ajo150@psu.edu), Navin Viswanathan (Commun. Sci. and Disord., The Penn State Univ., University Park, PA), Andrea Burgos Mercado (Commun. Sci. and Disord., The Penn State Univ., University Park, PA), and Grace Caplan (Commun. Sci. and Disord., The Penn State Univ., University Park, PA)

Speech-in-speech recognition is a challenge that listeners often encounter. Performance in such situations is improved if the competing speech is in a language different from the target—a phenomenon called Linguistic Release from Masking (LRM). LRM occurs due to a combination of energetic masking that arises from the physical overlap between the target signal and the masker and informational masking that arises from cognitive, attentional, and other factors. The contributions of informational masking to LRM is poorly understood. In the current study, we manipulated informational masking by varying the target-masker onset asynchrony while holding energetic masking constant. Typical, monolingual English listeners transcribed BKB sentences with either English or Mandarin two-talker babble in the background. The target sentences started either simultaneously with the masker, 500 ms after the masker, or 1000 ms after the masker. Listening accuracy was higher for the Mandarin than for the English masker, replicating typical LRM. However, the size of LRM was moderated by onset asynchrony with larger effects for longer lag times. This finding suggests that the detrimental effect of English masker accrues with lag. Implications of this finding for the role of informational masking in speech-in-speech listening will be discussed.

1pSC19. Listening effort mitigates rollover effects on speech-in-noise perception. Chengjie G. Huang (Dept. of Hearing and Speech Sci., Univ. of Maryland, 7251 Preinkert Dr., College Park, MD 20742, chuang88@umd.edu), Natalie A. Field (Dept. of Hearing and Speech Sci., Univ. of Maryland, College Park, MD), Marie-Elise Latorre (Program in Cognit. Sci., McGill Univ., Montreal, QC, Canada), Rebecca M. Farrar (School of Eng., Smith College, Northampton, MA), Samira Anderson, and Matthew J. Goupell (Dept. of Hearing and Speech Sci., Univ. of Maryland, College Park, MD)

Increasing the sound intensity may lead to worse speech understanding, especially in noise. This is known as the "Rollover" phenomenon. There is mounting evidence that listening effort plays an important role in challenging listening conditions and can be directly quantified with objective measures such as pupil dilation. However, there is limited understanding of how listening effort relates to rollover in speech understanding. We hypothesized that listening effort plays an essential role in mitigating rollover effects to differential extents across age and hearing status. We recruited across the adult lifespan ($N = 50$, 20–83 years) with different hearing statuses in acoustic listeners and cochlear implant users to perform a speech discrimination task. Minimal word pairs were presented both in quiet and in 0 dB SNR babble noise, ranging from 35–85 dB SPL. Pupil area was tracked simultaneously with behavioral responses during the task. We found that normal-hearing listeners are fully able to utilize effort contributions to minimize rollover effects between in quiet and in noise conditions, with diminishing benefit as a function of age and increased hearing loss. The results of this project could broadly influence how to design future hearing devices and interventions that maximize hearing abilities for those affected by hearing loss.

1pSC20. Listening effort under different rates of speech. Minhong Jeong (Korea Adv. Inst. of Sci. and Technol., 291, Daehak-ro, Yuseong-gu, Daejeon 34141, Republic of Korea, jeongmh@kaist.ac.kr), Haeun Oh (Korea Adv. Inst. of Sci. and Technol., Daejeon, Republic of Korea), Jaehan Park (KT Corp., Seoul, Republic of Korea), and Jieun Song (Korea Adv. Inst. of Sci. and Technol., Dajeon, Republic of Korea)

Listening effort, the cognitive resources allocated to understand spoken language, is a critical issue in communication under adverse listening environments. The present study investigated how listening effort would be affected by differing speech tempos. The experiment involved 23 native Korean adults with normal hearing, who were presented with linguistically complex Korean sentences at five different speeds (35%∼200% of original time). The cognitive load associated with listening under various speech rate conditions was quantified employing a dual-task paradigm, where the increased listening effort would impair the performance on an additional cognitive task. Participants listened to the sentences while performing an n-back task and then repeated back the sentence they heard. The results found a significant effect of speech rate on the n-back task, with lower accuracies in the faster speech conditions. The additional task also increased the overall cognitive load of sentence recognition, leading to various speech errors, such as the omission of key content words, unnecessary additions, and substitutions with similar words. This demonstrates that increased cognitive load during listening impairs speech processing at multiple levels (e.g., semantic processing, retention of words in working memory), emphasizing the need to adjust speech tempos for better understanding and comfort in listeners.

1pSC21. Exploring auditory recognition: Native Japanese listeners' perception of frequent and infrequent English words in noise with varied phonological neighborhood densities. 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)

Native Japanese listeners identified English words embedded in noise (SNR + 6dB) produced by native American English and Japanese speakers. The identical word list used in our previous study (Nozawa and Wayland 2023) was employed. Listeners responded by typing the words they heard. The findings revealed no discernible impacts of phonological neighborhood density, possibly due to the small vocabulary size limiting competition from other words. However, frequent words exhibited greater precision in recognition compared to infrequent words. Despite the introduction of noise, the overall accuracy of responses did not experience a substantial impact. Nevertheless, specific words like "class" and "club," initially nearly perfect in accuracy without noise, dropped to 0% accuracy. Most errors included phoneme substitution rather than phoneme addition or deletion, and in /CVC/ words, the onset consonant was identified substantially better than the vowel and the coda consonant. Initial consonant clusters, such as /br/, /kl/, and /pl/, were frequently perceived as /l/ or /r/, leading to the initial stops or fricatives being missed. Errors encompassed the substitution of phonemes with others that are not typically perceptually assimilated into the same native language (L1) categories, which would not be revealed through predefined multiple-choice identification or discrimination tasks.

1pSC22. Contribution of positive affect in infant directed speech: what do amplitude modulations patterns suggest? Samin Moradi (School of Commun. Sci. & Disord., McGill Univ., 1975 Boul De Maisonneuve O., Apt. 303, Montreal, QC H3H 1K4, Canada, samin.moradi@mail.mcgill.ca) and Linda Polka (School of Commun. Sci. & Disord., McGill Univ., Montreal, QC, Canada)

Speech perception relies heavily on cortical entrainment of amplitude modulations in speech, which occur at different rates. Following a study by Leong *et al.* (2017) showing slower modulations have higher power than faster modulations in infant directed speech (IDS), we hypothesized that positive emotions in IDS might drive this pattern. Using the same analyses (spectral amplitude modulation phase hierarchy method and phase synchronization index (PSI) (Leong and Goswami, 2015)), we compared the power of isolated modulation rates (synchronous with neural oscillations) and the synchrony between them in IDS and adult directed speech (ADS), using English stimuli from Many Babies Consortium (Frank *et al.*, 2020). We repeated the same analyses comparing happy and neutral ADS using stimuli of four native English speakers (Pell *et al.*, 2009). Our analysis did not uncover significant power differences between IDS and ADS. However, it revealed happy ADS has higher power at slower rates, with the reversed pattern for neutral ADS ($p < 0.001$). PSI was higher for two faster rates in neutral ADS ($p < 0.0001$), as reported by Leong *et al.* comparing IDS versus ADS. These findings reveal novel acoustic features of vocal emotions that might be important in attracting infant attention to IDS.

1pSC23. Limits of short-term perceptual training for enhancing Seoul-Korean listeners' use of English lexical stress in spoken word recognition. Annie C. Tremblay (The Univ. of Texas at El Paso, 500 West Univ. Ave., Liberal Arts 137, El Paso, TX 79912, actremblay@utep.edu), Hyoju Kim (Univ. of Iowa, Iowa City, IA), Sahyang Kim (Hongik Univ., Seoul, Democratic People's Republic of Korea), and Taehong Cho (Hanyang Univ., Seoul, Democratic People's Republic of Korea)

This study investigates whether short-term perceptual training can enhance Seoul-Korean listeners' use of English lexical stress in spoken word recognition. Unlike English, Seoul Korean does not have lexical stress (or lexical pitch accents/tones). Seoul-Korean speakers at a high-intermediate English proficiency completed a visual-world eye-tracking experiment adapted from Connell *et al*. (2018) (pre-/post-test). The experiment tested whether pitch in the target stimulus (accented versus unaccented first syllable) and vowel quality in the lexical competitor (reduced versus full first vowel) modulated fixations to the target word (e.g., *PARrot; ARson*) over the competitor word (e.g., *paRADE* or *PARish; arCHIVE* or *ARcade*). In the training (eight 30-min sessions over eight days), participants heard English lexical-stress minimal pairs uttered by four talkers (high variability) or one talker (low variability), categorized them as noun (first-syllable stress) or verb (second-syllable stress), and received accuracy feedback. The results showed that neither training increased target-over-competitor fixation proportions. Crucially, the same training had been found to improve Seoul-Korean listeners' recall of English words differing in lexical stress (Tremblay *et al*., 2022) and their weighting of acoustic cues to English lexical stress (Tremblay *et al*., 2023). These results suggest that short-term perceptual training has a limited effect on target-over-competitor word activation.

1pSC24. Fundamental dimensions of real-time spoken word recognition in cochlear implant users. Sarah E. Colby (Psychol. and Brain Sci., Univ. of Iowa, Psychol. Brain Sci. Bldg., 340 Iowa Ave. Rm. G60, Iowa City, IA 52242, sarah-colby@uiowa.edu), Francis X. Smith (Commun. Sci. and Disord., Univ. of Iowa, Iowa City, IA), Marissa Huffman, Charlotte Jeppsen, John B. Muegge, Ethan Kutlu, and Bob McMurray (Psychol. and Brain Sci., Univ. of Iowa, Iowa City, IA)

Spoken word recognition is a sophisticated cognitive process that maps incoming speech to meaning. For young, normal-hearing adults, word recognition is served by a competition process which plays out incrementally as speech unfolds. Candidates that match the input are activated and compete as mismatching candidates are suppressed. It remains unclear how this process changes in other listeners. Previous results from small-scale studies suggest three dimensions: 1) the speed of activating words (Activation Rate), affected by development and aging; 2) the degree of ultimate competitor suppression (Sustained Activation), affected by hearing loss; and 3) the delay of competition entirely (Wait-and-See), affected by severe hearing loss. To investigate these dimensions, a large, heterogenous group of cochlear implant users ($N = 101$) completed a Visual World Paradigm task. A principal component analysis supported these three dimensions. Each dimension was predicted by different demographic or auditory factors (onset of deafness, auditory fidelity, age), and each predicted outcomes over and above auditory fidelity. This suggests that these are orthogonal dimensions along which listeners vary, not all-or-nothing strategies. This work identifies the degrees of freedom that can extend theories of word recognition to account for a range of experiences and contexts.

1pSC25. The effect of talker intelligibility on adaptation to an unfamiliar accent. Kaitlyn L. Matthews (Psychol. and Brain Sci., Washington Univ. in Saint Louis, 1 Bookings Dr., Apt 604, St Louis, MO 63130, k.l.matthews@wustl.edu) and Kristin J. Van Engen (Psychol. and Brain Sci., Washington Univ. in St. Louis, St. Louis, MO)

Listening effort, as indexed by task-evoked pupil response, increases as the intelligibility of second language (L2) speech decreases, and this processing cost is mitigated by experience with talkers of the same accent (Porretta and Tucker, 2019). Prior research has shown that listeners can adapt to unfamiliar L2-accented speech and generalize the adaptation to different L2-accented talkers with the same L1 (Bradlow and Bent, 2008). The current study investigates what type of L2-accented speech exposure (high or low intelligibility) best reduces later listening effort to a novel talker of the same accent. We hypothesize that exposure to highly intelligible L2 speech supports adaptation to the features of an accent because it allows listeners to map accented tokens to the talker's intended lexical representations. Alternatively, adaptation to a less intelligible talker may require exposure to other low-intelligibility speech, which is characterized by features that deviate more significantly from L1 norms. To test this, participants were exposed to simple sentences spoken in English by either L1 talkers, highly intelligible L2 talkers, or less intelligible L2 talkers. At test, all participants listened to different sentences spoken by a previously unheard L2 talker while listening effort was measured via pupillometry. Data collection is ongoing.

1pSC26. Perception and recognition of English /s/ and /ʃ/ with varying acoustic-auditory contrast. Molly Babel (Linguist., Univ. of BC, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, molly.babel@ubc.ca), Roger Y. Lo (Linguist., Univ. of BC, Vancouver, BC, Canada), Charlotte Vaughn (Linguist., Univ. of Maryland, Eugene, OR), and Michael McAuliffe (Amazon, Bellevue, WA)

Seminal work (Newman *et al*., JASA, 2001) found that listeners' responses to talkers with more variable fricative productions were slower, though listeners' ability to categorize the varied fricatives was robust. The current study selects North American English-speaking talkers with /s/ and /ʃ/ productions that varying in the magnitude of the acoustic-auditory contrast. With selected speech samples, listeners were asked to categorize (1) the isolated fricative (C-only), (2) the fricative-vowel sequence (CV), or (3) to complete a speeded-shadowing task where listeners were auditorily presented with the full words and asked to identify the words by repeating them as quickly and accurately as possible. Data were analyzed with Bayesian methods. The fricatives from talkers with greater contrast were identified more accurately and more quickly, with a greater effect size for the C-only condition and /s/ productions. This suggests listeners leverage information from the formant transitions to differentiate these fricatives. The speeded-shadowing results indicate the participants are faster at identifying the words with less acoustic-auditory contrast. This is the opposite of the expected pattern. Coupling C-only, CV, and word-level responses paints a more accurate picture of how talker differences in auditory-acoustic contrast affect categorization and intelligibility.

1pSC27. Relationship between intelligibility, naturalness, and listening effort of source-separated speech. Behdad Dousti (Commun. Sci. and Disord., Univ. of Cincinnati, Cincinnati, OH), Richard Goldhor (Speech Technol. and Appl. Res. Corp., Lexington, MA 02421, rgoldhor@protonmail.com), Sarah Dugan (Rehabilitation, Exercise, and Nutrition Sci., Univ. of Cincinnati, Dayton, OH), Joel MacAuslan (Speech Technol. and Appl. Res. Corp., Lexington, MA), and Suzanne Boyce (Commun. Sci. and Disord., Univ. of Cincinnati, Cincinnati, OH)

Various signal processing techniques have been proposed for separating a speech signal from an acoustic environment that includes other sound sources. One important measure of success of a source separation method is the intelligibility of the extracted speech signal, determined as the fraction of intended words correctly recognized. However, even for extracted signals with similar intelligibility, listeners may experience different reactions to the result. Speech samples extracted by different techniques may differ in the degree to which they sound pleasant or natural to a listener, and/or the degree of cognitive effort required to understand them. We present results of an experiment in which listeners were presented with context-free spondees recorded in a noisy environment and subsequently processed in various ways to enhance the speech and suppress the noise. Listeners transcribed the speech and also judged the "naturalness" and "listening effort" of the speech. Intentionally, stimulus intelligibility varied widely—as did naturalness and effort. We present and discuss the measured relationship among intelligibility, perceived naturalness, and reported listening effort.

1pSC28. Potential influences of autistic traits and perceptual acuity on lexically guided perceptual learning. Shawn N. Cummings (Univ. of Connecticut, 2 Alethia Dr., Unit 1085, Storrs, CT 06269-1085, shawn.cummings@uconn.edu), Brooke Duda, Wesley Medeiros, and Rachel M. Theodore (Univ. of Connecticut, Storrs, CT)

Listeners overcome rampant variability in speech input, at least in part, by adapting to talker-specific speech patterns. However, most work demonstrating this type of perceptual learning has focused on group-level effects in modal populations. This approach masks potentially meaningful differences in sensory perception, social functioning, and language processing among individual listeners and/or populations. These differences—present among all listeners but particularly associated with autism—may well be expected to influence adaptation, but their role remains unclear. Previous investigations have reported *absent adaptation* among diagnosed autistic listeners in addition to *increased adaptation* among listeners with more autistic traits. The present investigation aims to clarify these potentially contradictory findings and the relationships between autistic traits, perceptual acuity, and adaptation. Listeners will hear ambiguous spectral energy between /s/ and /ʃ/ in lexical contexts designed to elicit adaptation and then categorize tokens from an *ashi–asi* continuum to assess learning. Autistic traits and pitch perception will also be assessed. We predict a non-linear interaction between the autism quotient and adaptation which cannot be attributed to perceptual acuity alone. These results will help explain how differences in sensory perception and social functioning associated with autism interact with lexically guided perceptual learning.

1pSC29. The effect of age of acquisition on listening effort: Pupillometry and subjective measures. Sita Carraturo (Psychol. & Brain Sci., Washington Univ. in Saint Louis, 1 Brookings Dr., Saint Louis, MO 63130, sita@wustl.edu) and Kristin J. Van Engen (Psychol. and Brain Sci., Washington Univ. in St. Louis, St. Louis, MO)

Bilinguals typically perform worse on speech-perception-in-noise relative to monolinguals, and bilinguals who learned the language later typically perform worse than those who learned it earlier. This decrement in performance has largely been attributed to the fact that both bilingualism and later acquisition result in comparatively less exposure to a language than monolingualism and earlier acquisition do. Though this effect of age of acquisition has been well-documented for speech recognition accuracy, less is known about how it affects listening effort. This study quantifies listening effort using three measures (pupil dilation, subjective effort ratings, and subjective fatigue ratings) for four listening conditions (quiet, easy, moderate, and hard). We collect these measures from four groups of participants: monolinguals, simultaneous bilinguals, early sequential bilinguals, and late bilinguals. Data collection is ongoing for a target sample size of 128 (32 per group). Based on related work, we expect greater listening effort for listeners with later ages of acquisition (e.g., late bilinguals relative to simultaneous bilinguals), and for that effect to increase with worsening acoustic quality (i.e., the hard relative to the easy condition). Preliminary results ($N = 56$) support these hypotheses. These data provide foundational insights into the relationship between bilingualism and listening effort.

1pSC30. Islands in the stream: Talker interference is conditioned on phonetic continuity. Rachel M. Theodore (Univ. of Connecticut, 850 Bolton Rd., Unit #1085, Storrs, CT 06269, rachel.theodore@uconn.edu) and Sahil Luthra (Carnegie Mellon Univ., Pittsburgh, PA)

Listeners endure a processing penalty in multi-talker compared to single-talker environments, with both speech perception and auditory attentional mechanisms implicated as its locus. Here, we argue that it is useful to consider how speech processing is influenced not only by changes in talker but also by changes in phonetic content (e.g., word changes) across trials. Thus, the current work tests the hypothesis that mixed-talker input incurs a processing cost due to trial-level, talker-contingent disruptions in auditory attention. In six experiments, listeners completed a speeded word identification task involving a single-talker block and a mixed-talker block. In the mixed-talker block, we manipulated whether word identity and talker identity were the same or different across consecutive items. As expected, responses to targets in mixed-talker blocks were always slower than responses to targets in single-talker blocks. However, RTs to targets in mixed-talker blocks were only disrupted by a change in talker when word identity was held constant across items. In addition, though the presence of a carrier led to faster RTs in mixed-talker blocks, this held even when the carrier was in a different voice from the to-be-identified word. These results have important implications for understanding the mechanisms through which multi-talker processing costs emerge.

1pSC31. Effects of time compressed speech and background noise on memory recall. Min Young Lee (Psych., Binghamton Univ., Binghamton, NY) and Sung-Joo Lim (Psych., Binghamton Univ., 4400 Vestal Parkway E, Psych. (Sci. 4), Binghamton, NY 13902, sungjoo@binghamton.edu)

Listeners demonstrate remarkable flexibility in processing speech under varying conditions, rapidly adapting to acoustically distorted signals like time-compressed speech or speech in noise. Existing research has explored the immediate effects of time-compression and background noise on speech intelligibility, but the impact of these distortions on long-term memory recall for speech content is less well understood. With the growing use of online education platforms where listeners often prefer to use accelerated playback, understanding the effects of time-compression and noise on long-term memory from speech is crucial. Here, we investigated how speech distortions from time-compression influence explicit memory, and how this interacts with background noise. Participants ($N = 35$) each listened to six recorded lectures at varying time-compression rates (1.0x, 1.5x, 2.0x) and background noise levels (quiet versus babble noise at + 10 dB SNR). After each lecture, participants answered 10 multiple-choice questions probing explicit content recall. Listeners exhibited a significant decline in content recall with increased time compression, while background noise during the lecture did not impact listeners' content recall. Our findings demonstrate that time-compressed speech negatively affects long-term memory recall, which may arise from the additional cognitive demands necessary to process time-compressed speech, even after perceptual adaptation.

1pSC32. Abstract withdrawn.

1pSC33. Abstract withdrawn.

1pSC34. Lexical encoding of second language tones in English learners of Mandarin. Kuo-Chan Sun (Univ. of AB, Pembina Hall 3-20, Univ. of Alberta, Edmonton, AB T6G 2H8, Canada, kuochan@ualberta.ca)

The study investigates *how* native and non-native differences in tone perception influence lexical encoding of second language (L2) tones. Previous work shows that tone language listeners perceived tones as phonemic categories while non-tone language listeners relied more on psychoacoustic cues such as pitch height to discriminate stimulus tones. However, little is known about the influence of such differences in perception on L2 tonal encoding. In the present study, two experiments were conducted with nineteen English learners of Mandarin and 20 Mandarin native speakers. Experiment 1 was an ABX task. Results showed that while native speakers' overall performance was superior to L2 listeners', both groups poorly discriminated tone pairs with shared tone contours (i.e., T2-T3). Experiment 2 was a medium-lag repetition priming task. In the repetition condition, significant facilitations were observed in both language groups. In the minimal-tone-pair condition, despite T2-T3 contrasts posing greater challenge for both groups to accurately distinguish than other tonal contrasts as shown in Experiment 1, positive priming was observed only in the L2 group. The findings from the two experiments suggest that the L2 listeners, although quite proficient in Mandarin, have yet to achieve native-like competence in regard to lexical tones.

1pSC35. Impact of noise, dysphonia, and cognitive functions on speech perception in children. Silvia Murgia (Speech and Hearing Sci., Univ. of Illinois - Urbana Champaign, 901 South Sixth St., Champaign, IL 61820, smurgia2@illinois.edu), Bisma N. Choudhry (Speech and Hearing Sci., Univ. of Illinois - Urbana Champaign, Champaign, IL), Mary M. Flaherty (Speech and Hearing Sci., Univ. of Illinois Urbana-Champaign, Champaign, IL), and Pasquale Bottalico (Dept. of Speech and Hearing Sci., Univ. of Illinois Urbana-Champaign, Champaign, IL)

This study explores the detrimental effects of classroom noise and teacher dysphonia on children's speech understanding and teacher voice quality. It also investigates their impact on cognitive functions such as attention and working memory, as well as the subjective and objective measures of listening effort. Participants were 26 children aged 8–12. Results indicate that word recognition was primarily affected by the signal-to-noise ratio (SNR), with lower SNRs leading to decreased accuracy. However, listening comprehension scores were significantly influenced by both SNR and voice quality, especially when the lowest SNR combined with a dysphonic voice resulted in a significant decrease in accuracy. Higher working memory was associated with better comprehension performance. Subjective listening effort was higher when noise and dysphonia were present, with greater selective attention linked to lower perceived effort. Response times also showed that children took more time to respond in lower SNR conditions and when the speech was dysphonic, although higher selective attention led to shorter response times. In conclusion, noise primarily impacted word recognition, while dysphonia exacerbated listening comprehension. However, both factors increased perceived listening effort. These findings suggest that assessing word recognition alone may underestimate the impact of poor voice quality in noisy environments.

MONDAY AFTERNOON, 13 MAY 2024 ROOM 213, 1:00 P.M. TO 2:50 P.M.

Session 1pSP

Signal Processing in Acoustics: Signal Processing in Acoustics Potpourri I

Trevor Jerome, Cochair
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Manton J. Guers, Cochair
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Chair's Introduction—1:00

Contributed Papers

1:05

1pSP1. Investigating the efficacy of autoencoders and other machine learning methods for studying dynamic oceanic processes in long-range acoustic propagation environments. Ananya Sen Gupta (Dept. of Elec. and Comput. Eng., Univ. of Iowa, 103 S Capitol St., Iowa City, IA 52242, ananya-sengupta@uiowa.edu), Andrew J. Christensen (Elec. and Comput. Eng., Univ. of Iowa, Iowa City, IA), Timothy Linhardt (College of Elec. and Comput. Eng., Univ. of Iowa, Iowa City, IA), Ivars Kirsteins (NUWC, Newport, RI), and Kay L. Gemba (Phys. Dept., NPS, Monterey, CA)

Long-range acoustic propagation is a topic of great interest in applications like acoustic thermometry and underwater navigation. These applications utilize the measured arrival times and structure obtained from the transmitted probe pulses. However, they are highly dependent on the stability and repeatability of the ocean channel. In real oceans, random medium effects like internal waves [1] can induce considerable fluctuations and distortions to the received probe pulses. Our objective here is to investigate the use of machine learning (ML) methods such as autoencoders and other deep learning architectures to see if they can unravel and give insight into the dynamics of the ocean processes generating the fluctuations. In particular, we will investigate geometric concepts from braid, loops, and knot theory that can capture the changing shapes of smoothly deforming features that represent these processes. For the analysis, we will use transmitted frequency maximum length sequence (MLS) signal probe pulses from the 75 Hz Kauai Beacon source received at the International Monitoring Station near Wake Island at a nominal distance of 3500 km. We show that ML analysis can provide some useful insights. J. Xu, "Effects of internal waves on low frequency, long range, acoustic propagation in the deep ocean," MIT Ph. D. dissertation (2007).

1:20

1pSP2. Feature extraction of small underwater sonar targets using neighborhood discriminant analysis. Andrew J. Christensen (Elec. and Comput. Eng., Univ. of Iowa, 3100 Seamans Ctr., Iowa City, IA 52242, andrew-christensen@uiowa.edu), Ananya Sen Gupta (Dept. of Elec. and Comput. Eng., Univ. of Iowa, Iowa City, IA), and Ivars Kirsteins (NUWC, Newport, RI)

Linear discriminant analysis (LDA) stands as a widely used supervised feature extraction technique that maps data onto a low-dimensional subspace such that the between-class scatter is maximized while within-class scatter is minimized. Despite its utility, LDA faces many challenges, particularly when the within-class scatter matrix becomes singular due to small sample size. Other dimensionality reduction techniques, such as Neighbourhood Component Analysis (NCA), have been proposed as alternatives to LDA. NCA learns a linear transformation that maximizes the likelihood that datapoints of the same class are clustered together in the lower-dimensional space. However, the optimization of NCA relies on a non-convex cost function, making it prone to local minima. To address the challenges faced by both LDA and NCA, we propose a novel dimensionality reduction method named neighborhood discriminant analysis (NDA). Like NCA, NDA learns a linear transformation that aims to cluster datapoints based on class label. However, NDA is framed as an eigendecomposition problem, eliminating the need for non-convex optimization. We demonstrate the new approach on real small target sonar data. [Work funded by the ONR grant numbers N000142112420 and N000142312503, and DoD Navy (NEEC) Grant No. N001742010016.]

1:35

1pSP3. Blind passive signal detection via dictionary learning in unknown multipath time-spreading distortion underwater channels. Rami Rashid (Elec. Comput. Eng. Dept., New Jersey Inst. of Technol., Newark, NJ), Ali Abdi (Elec. Comput. Eng. Dept., New Jersey Inst. of Technol., 323 Dr. Martin Luther King Jr. Blvd., Newark, NJ, ali.abdi@njit.edu), and Zoi-Heleni Michalopoulou (Dept. of Mathematical Sci., New Jersey Inst. of Technol., Newark, NJ)

Blind passive signal detection is a challenging endeavor since typically there is no prior knowledge of the transmitted signal. This becomes further complicated in time-spreading distortion (TSD) underwater channels [R. Rashid, E. Zhang, A. Abdi, and Z. H. Michalopoulou, "Theoretical and experimental multi-sensor signal detection in time spreading distortion underwater channels," Proc. Oceans (2022)], where due to the presence of multiple propagation paths, the unknown signal is convolved with an unknown channel impulse response. In this paper, we introduce and develop a blind passive signal detection method, using dictionary learning [M. Sadeghi, M. Babaie-Zadeh, and C. Jutten, "Dictionary learning for sparse representation: A novel approach," IEEE Signal Process. Lett. (2013)]. In our blind passive signal detection method, we use the received data to estimate the unknown signal, and also to separate it from the unknown channel impulse response. We have conducted simulations and underwater experiments, to generate receiver operating characteristic curves, to study the performance of the proposed method. The high detection probabilities of the blind method, compared to the replica correlation integration method—that needs to know the transmitted signal for matched filtering—demonstrate the usefulness of the method for passive detection of unknown signals in unknown TSD channels.

1:50

1pSP4. Performance bounds for source localization using tetrahedral microphones. Steven J. Todd (The Penn State Univ., State College, PA, svt5691@psu.edu), Daniel C. Brown (Penn State Univ., State College, PA), and Michael Roan (ME, Penn State, State College, PA)

The Cramér-Rao Bound (CRB) quantifies the minimum variance on a given unbiased estimation parameter. It can be used to derive a lower bound on source localization error for arrays, of which there are several probabilistic models in the literature. However, existing work only considers source localization using arrays of omnidirectional sensors. This presentation describes a model of the CRB as a minimum variance estimation of localization using multiple tetrahedral microphones which have cardioid microphones. Tetrahedral microphones are commonly used to record soundfields for virtual audio reproduction. The proposed method uses the cross spectral density of signals in order to calculate the CRB. This allows for more signal parameters to be included in the model. The model of the CRB was validated with an experiment in an anechoic room with a single sound source. Experimental results show the CRB forms a lower bound at low signal-to-noise ratios.

2:05

1pSP5. GHOST target suppression for distributed multi-array localization using temporal and spatial correlations. Mingu Kang (Underwater Acoust. and Signal Processing, Sejong Univ., 209, Neungdong-ro, Gwangjin-gu, Seoul 05006, Republic of Korea, rkd9412@sju.ac.kr), Youngmin Choo (Underwater Acoust. and signal processing, Sejong Univ., Seoul, Republic of Korea), Keunhwa Lee (Underwater Acoust. and signal processing, Sejong Univ., Seoul, Republic of Korea), and wooyoung hong (Underwater Acoust. and signal processing, Sejong Univ., Seoul, Republic of Korea)

We propose a ghost target suppression method for distributed multi-array localization. Triangulation, included in bearings-only target localization method for the well-separated array, is highly dependent on bearing measurements. Its performance is degraded by ghost targets from the ambiguity of bearing measurements. The proposed method is based on the assumption that received signals from the same target have similar frequency components within short-term time and at different locations. We utilize the consistency of acoustic signals in Frequency-Azimuth plots by measuring the frequency component similarities of the target signal over time at a single array (temporal information) and across different arrays (spatial information). The ghost target suppression is evaluated using *in situ* data from Swellex-96 S5 Event and results reveal the target track along time with much less ghost targets. [Work supported by Korea Research Institute for defense Technology planning and advancement(KRIT)—Grant funded by the Korea government(DAPA(Defense Acquisition Program Administration))(No. KRIT-CT-23-026, Integrated Underwater Surveillance Research Center for Adapting Future Technologies, 2024).]

2:20

1pSP6. On the accuracy of acoustic source localization with GPS-enabled environmental acoustics recorders and high-precision receiver placement. Zadia A. Huges (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, zadiaah@student.byu.edu), Levi T. Moats (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), Megan R. McCullah-Boozer (CSU Bakersfield, Bakersfield, CA), and Lucas K. Hall (Biology, California State Univ. Bakersfield, Bakersfield, CA)

The technique of using microphone arrays with cross correlation to obtain acoustic source locations has many applications. Imprecise array element time synchronization and placement yield errors in source localization efforts. This paper describes experiments to determine the accuracy of localization measurements made using multiple Wildlife Acoustics SM4TS recorders that are positioned using both the provided GPS receiver and a differential GPS unit that is accurate to within 1 cm. Results from various array geometries, element spacings, and signal types are presented. [Work supported by USACE.]

2:35

1pSP7. Comparison of sparse arrays that maximize detection and optimize beampatterns. Zaynah Kalaoun (Elec.,Comput. and Biomedical Eng., Univ. of Rhode Island, 75 Briar Ln., Kingston, RI 02881, zaynah_kalaoun@uri.edu), Kaushallya Adhikari, and Steven Kay (Elec.,Comput. and Biomedical Eng., Univ. of Rhode Island, Kingston, RI)

In a standard uniform linear array (ULA), the sensors are uniformly spaced on a grid with half wavelength intersensor spacing. The beampattern of an array shows the gain applied by the conventional beamformer to a plane wave arriving at the array from various possible directions. The main lobe width, peak side lobe height, and the side lobe area are important beampattern parameters. Many sparse arrays have been widely studied because of their ability to provide one or more of the same beampattern parameters as a full array using fewer sensors. Examples of such arrays are coprime [Vaidyanathand and Pal 2011] and nested [Pal and Vaidyanathan 2010] arrays. Recently, an exact way to design sparse sampling schemes to maximize detection performance of a known signal in first-order autoregressive noise was introduced [Adhikari and Kay 2023]. In this research, we compare the beampatterns and detection performances of these detection maximizing arrays with the sparse arrays that are designed to optimize beampattern metrics. We analyze beampatterns and detection performances under various conditions using simulated and real underwater acoustic data.

Session 1pUW

Underwater Acoustics, Acoustical Oceanography, Signal Processing in Acoustics, and Computational Acoustics: Data Science in Ocean Acoustics II

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Invited Papers

1:00

1pUW1. Ethical considerations and regulatory frameworks in ocean acoustic data science. Solomon O. Ologe (Mech., Univ. Polytechnic of Catalonia, Calle de Colom, 11, Tarrassa Campus, Barcelona 08222, Spain, ologe.solomon@upc.edu)

The integration of data science methodologies into ocean acoustic research has ushered in unprecedented capabilities for monitoring and understanding marine environments. However, this technological advancement also brings forth a myriad of ethical considerations and challenges concerning the responsible collection, use, and dissemination of acoustic data. This article delves into the ethical implications of deploying acoustic monitoring systems in marine ecosystems, addressing concerns related to potential impacts on marine life, biodiversity conservation, and indigenous communities. Furthermore, it examines the existing regulatory frameworks governing ocean acoustic data science, highlighting gaps, inconsistencies, and opportunities for enhancing governance and accountability. Through a comprehensive analysis of ethical dilemmas and regulatory constraints, this article underscores the imperative for adopting ethical best practices and robust regulatory mechanisms to ensure the ethical conduct and sustainability of ocean acoustic data science endeavors.

1:20

1pUW2. Variability and influence of fisheries acoustic echogram annotations on machine learning applications. Wu-Jung Lee (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, wjlee@apl.washington.edu), Valentina Staneva (Univ. of Washington, Seattle, WA), and Caesar Tuguinay (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

High-frequency echosounders are the workhorse in fisheries and marine ecological surveys. Due to the inherent complexity of biological aggregations and ambiguity in interpreting echoes from species of similar size and anatomical compositions, echogram annotation typically requires combining spectral information referencing scattering physics, biological ground-truth from nearby net-trawls, and empirical school morphology of specific fish species. Here, we investigate the variability of echogram annotations and its influence on machine learning applications using data from the biennial Pacific hake acoustic-trawl survey. Compared to many other fish species, hake tend to possess less defined school boundaries with variable acoustic features and often form mixed-species aggregations in the mesopelagic. Nonnegative matrix factorization and hierarchical clustering of volume backscattering strength (Sv) distributions across the 18, 38, and 120 kHz channels revealed a spectrum of annotation region types that reflect differences in morphological and acoustic features as well as differences in annotator style. This variability likely contributes to the observed variable segmentation behavior of deep learning models trained using this dataset. These results highlight the importance of considering the diversity of echogram annotation, its connection to scattering physics and the underlying aggregation composition, and the incorporation of such information in developing machine learning models.

1:40

1pUW3. Automated zooplankton detection from *in situ* imagery for forward scattering predictions. Benjamin D. Grassian (Biology, AOPE, Woods Hole Oceanographic Inst., 45 Water St., Falmouth, MA 02543, bgrassian@gmail.com), Andone C. Lavery (AOPE, Woods Hole Oceanographic Inst., Woods Hole, MA), Heidi Sosik, Megan Ferguson (Biology, Woods Hole Oceanographic Inst., Falmouth, MA), Sidney Batchelder (Information Services, Woods Hole Oceanographic Inst., Falmouth, MA), and Elizabeth T. Crockford (Biology, Woods Hole Oceanographic Inst., Falmouth, MA)

Ocean midwaters—the vast region between the sunlit surface layers and seafloor—comprise the largest habitat on Earth but are among the least understood marine environments. This project aims to combine concurrently collected imaging and acoustic measurements in the epi and mesopelagic environment to interpret zooplankton scattering from mixed assemblages and determine *in situ* zooplankton distributions within their local environments. We have trained a machine learning model for the automated detection of 13 zooplankton functional groups from a 1000m rated towed shadowgraph imaging system. The zooplankton detection model currently achieves 80% F1 scores on our validation image set and was trained using adversarial methods. We have derived biometric measurements from the zooplankton image data necessary to generate forward scattering predictions. We compare the distributions of scattering layers detected acoustically with zooplankton distributions from the imagery. We will compare forward scattering predictions derived from sparse geometric representations of the zooplankton and full 3D model volumes using the distance transform of the zooplankton images. This work will further enable the use of optical techniques for midwater surveys and the interpretation of acoustic scattering returns from mixed zooplankton populations.

1:55

1pUW4. Potential of K-means clustering for preliminary labeling of acoustic data samples. Emily C. Bacon (Phys. and Astronomy, Brigham Young Univ., N286 ESC, Provo, UT 84602, e2cook98@byu.edu), Tracianne B. Neilsen (Phys. and Astronomy, Brigham Young Univ., Provo, UT), Paul Leary, Kay L. Gemba, and Kevin B. Smith (Phys. Dept., Naval Postgrad. School, Monterey, CA)

Labeling data for use in supervised learning algorithms can be a long and arduous process, especially for large datasets. This work explores if an unsupervised k-means clustering algorithm can act as a tool to efficiently obtain preliminary labels for large acoustic datasets. This approach is tested on vector sensor data from Monterey Accelerated Research System (MARS). To determine the optimal number of clusters in the k-means algorithm, the silhouette method is used. The clustering of different types of input data will be compared. Specifically, time average spectra from the pressure sensor, the acoustic particle velocity, and intensity will be considered. In addition, the results of clustering data from different times of day and months over a two-year period will be compared. Results of the clustering will be presented along with an assessment of the potential for using a k-means clustering as a preliminary labeling tool for large acoustic sets. [Undergraduate research funded by the College of Physical and Mathematical Sciences as Brigham Young University.]

2:10

1pUW5. Using a sequence deep learning model to increase the acoustic context of a killer whale detector. Fabio Frazao (Comput. Sci., Dalhousie Univ., 6050 Univ. Ave., Halifax, NS B3H 1W5, Canada, fsfrazao@dal.ca), Oliver S. Kirsebom (Comput. Sci., Dalhousie Univ., Halifax, NS, Canada), April Houweling (Simon Fraser Univ., Burnaby, BC, Canada), Jennifer Wladichuk (Univ. of Victoria/JASCO Appl. Sci., Victoria, BC, Canada), Jasper Kanes (Sci. Services, Ocean Networks Canada, Victoria, BC, Canada), Ruth Joy (School of Environ. Sci., Simon Fraser Univ., Burnaby, BC, Canada), and Mike Dowd (Mathematics & Statistics, Dalhousie Univ., Halifax, NS, Canada)

Although Killer whales (*Orcinus orca*) produce many stereotypical vocalizations, their sounds can be difficult to identify in isolation. Experts often rely on acoustic context to accurately identify these animals acoustically. Automated detectors and classifiers, on the other hand, frequently rely on short clips that capture individual vocalizations, not leveraging information regarding previous sounds. We developed deep learning models that used 1-minute inputs containing from 0 to 50 calls, with the average clip having 18. We tested three artificial neural network architectures that used recurrent layers to take the sequence of acoustic events into account. As a baseline, we used a convolutional neural network that only took 3-s clips at a time, without considering sequences of events. Here, we present preliminary evaluations on a dataset containing 360 min with Southern Resident killer whale activity in the Salish Sea, and an equal amount of data without killer whale sounds. The best model used a combination of temporal convolutional layers and gated recurrent units to achieve a recall of 0.95 at the maximum precision of 0.98. The models will be applied to near real-time monitoring efforts and will be open-sourced in the future.

2:25

1pUW6. Physics-anchored masked autoencoder for efficient sonogram simulation. Jongkwon Choi (Dept. of Ocean Systems Eng., Sejong Univ., 209, Neungdong-ro, Gwangjin-gu, Seoul 05006, Republic of Korea, jkchoi.sju@gmail.com), Geunhwan Kim (Dept. of Ocean Systems Eng., Sejong Univ., Seoul, Republic of Korea), Youngmin Choo (Dept. of Ocean Systems Eng., Sejong Univ., Seoul, Republic of Korea), wooyoung Hong (Dept. of Ocean Systems Eng., Sejong Univ., Seoul, Republic of Korea), and Keunhwa Lee (Dept. of Ocean Systems Eng., Sejong Univ., Seoul, Republic of Korea)

We present a technique for efficiently creating sonogram such as LOFAR and DEMON, by combining physical acoustic modeling and data-driven method of masked autoencoder. This technique involves two stages. First, in the given the ocean environment, the physical model accurately calculates a restricted portion of entire sonogram image. Next, the data-driven model based on masked autoencoder creates an image of remaining region. By employing an iterative decoding structure and controlling the weight of the physical loss term, the reconstruction accuracy of masked autoencoder is improved. The results are compared with those of a pure physical model, a hybrid model with original masked autoencoder, and a hybrid model with traditional PCA technique. We will discuss the practicality of the proposed technique in terms of accuracy and calculation performance, as well as the applicablity of real data using Deepship dataset. [Work supported by the Korea Research Institute for Defense Technology Planning and Advancement (KRIT) grant funded by the Korea government (Defense Acquisition Program Administration (DAPA)) (No. KRIT-CT-22-052, Physics-guided Intelligent Sonar Signal Detection Research Laboratory).]

Invited Paper

2:40

1pUW7. An asymptotically exact estimate of the median noise eigenvalue of sample covariance matrices. Yongjie Zhuang (Stony Brook Univ., Light Eng., Stony Brook, NY 11790, yongjie.zhuang@stonybrook.edu), David C. Anchieta, John R. Buck (Elec. and Comput. Eng., UMass Dartmouth, North Dartmouth, MA), and Andrew C. Singer (Stony Brook Univ., Stony Brook, NY)

The Dominant Mode Rejection beamformer [Abraham & Owsley (1990)] averages the noise subspace eigenvalues of the sample covariance matrix (SCM) to estimate the background noise power. This noise power estimate from snapshot deficient SCMs can be unreliable for large arrays in time-varying environments with limited snapshots. Median filtering offers robustness to outliers and subspace mismatch for these challenging problems. Recent numerical experiments [Campos Anchieta & Buck (2022)] identified a simple regression relating the median sample eigenvalue to the true background power for snapshot deficient SCMs. However, the complicated expression of the Marchenko-Pastur (MP) distribution for the noise eigenvalues of the SCM thwarted prior attempts to find a closed form estimator for the MP distribution median. This talk exploits a coordinate transform to obtain a closed-form median of the MP distribution that is exact for the leading term in the power series expansion of the distribution. The new median estimator is more accurate than the previous numerical regression when the ratio of sensors to snapshots is 2 or more. Given the central role of SCM eigenvalues in principal component analysis and constant false alarm rate detectors, the new median expression should find application in other data science algorithms for underwater acoustics. [Work supported by ONR Code 321US.]

Contributed Paper

3:00

1pUW8. Fast deconvolved beamforming For arbitrary arrays based on beam-domain sparse Bayesian learning. Jianli Huang (State Key Lab. of Acoust., Inst. of Acoust., Chinese Acad. of Sci., No. 21 North 4th Ring Rd., Haidian Dist., Beijing 100190, China, huangjianli@mail.ioa.ac.cn), Yu Wang, Zaixiao Gong, Jun Wang, and Haibin Wang (State Key Lab. of Acoust., Inst. of Acoust., Chinese Acad. of Sci., Beijing, China)

Aiming at the problem that the conventional deconvolved beamforming methods cannot be directly applied to the specific array with a shift-variant point spread function and also have considerable computational workload, a deconvolved beamforming method for arbitrary arrays based on beam-domain sparse Bayesian learning(SBL) is proposed. First, generalized convolution model for arbitrary arrays in beam-domain is derived. Conventional beamforming is used to obtain several complex output beams. Then, to improve the accuracy of direction of arrival(DOA) estimation, the off-grid SBL method which adopts a coarse grid and takes the sampled positions in the coarse grid as the adjustable parameters is applied to achieve deconvolution of complex output beams. Controlling the number of output beams from the conventional beamforming can accelerate the off-grid SBL method while maintaining reasonable accuracy. The simulation results show that the proposed method provides enhanced recovery performance and higher DOA estimation accuracy comparable to the traditional SBL beamforming method in element domain at the same grid interval. Especially for short and dense arrays, it can achieve a decrease in computational complexity by one to two orders of magnitude with the same accuracy.

3:15–3:30 Break

Invited Paper

3:30

1pUW9. Automatic detection and 2D localization using a network of unsynchronized passive acoustic sensors in a dispersive waveguide. Mark Goldwater (Woods Hole Oceanographic Inst., 266 Woods Hole Rd., MS #55, Woods Hole, MA 02543-1050, mgoldwater@whoi.edu), Daniel P. Zitterbart (Woods Hole Oceanographic Inst., Woods Hole, MA), and Julien Bonnel (Woods Hole Oceanographic Inst., Woods Hole, MA)

The time-frequency positions of modal dispersion curves in shallow-water low-frequency impulsive signals are strongly dependent on source-receiver range, making them suitable for range-based localization. Here, we apply a temporal convolutional network (TCN) to spectrograms estimated from individual sensors in an array of unsynchronized hydrophones to simultaneously detect dispersive signals and produce source-range estimates. The TCN is trained on simulated signals generated over a spatial grid and various environmental parameters using the adiabatic approximation for normal modes. Assuming that the number of unique sources is unknown, range measurements from the same source across different sensors are simultaneously associated and used in localization. To accomplish this automatically, the proposed method considers all unique combinations of range measurements from every collection of k sensors. For every range measurement combination, if the location estimate generated using each subcombination of k-1 measurements is within a certain threshold of the remaining measurement, the whole collection is labeled as group-k consistent. All such groups of measurements are represented as neighboring nodes in a graph, and strongly connected components are used to calculate the final source location estimates. The whole detection/localization method is validated using both simulated and experimental marine data. [Work supported by NDSEG and ONR].

Contributed Papers

3:50

1pUW10. Mode-informed complex-valued neural processes for matched field processing. Yining Liu (Beijing Inst. of Technol., No. 5 South St. Zhongguancun, Haidian Dist., Beijing 100081, China, lyning@bit.edu.cn), Runze Hu, Desheng Chen, and Lijun Xu (Beijing Inst. of Technol., Beijing, China)

Matched field processing (MFP) is a key technique for passive underwater source localization, estimating the source position by matching array measurements with acoustic model replicas. Its effectiveness relies on matching environmental parameters with the actual oceanic environment, but performance declines with environmental mismatches and lower signal-to-noise ratios. This paper proposes a novel approach that integrates neural networks (NNs) and complex Gaussian processes with modal depth functions for acoustic field reconstruction that is more accurate and efficient compared to Gaussian process regression. A meta-learning strategy is used to optimize parameters of the NN. The reconstructed data are denoised and interpolated, generating densely populated acoustic fields at virtual arrays, which are then used as data in MFP. Replicas are also computed at the virtual receivers. This mode-informed complex-valued neural processes enhance MFP performance, particularly in low SNR and mismatched environments. It captures the propagation properties of underwater acoustic fields more comprehensively, showing superior localization performance in both simulated and real-world data from the SWellEx-96 Event S5 environment.

4:05

1pUW11. Online machine learning-based channel estimation for underwater acoustic communications. Yonglin Zhang (Inst. of Acoust., Chinese Acad. of Sci., No. 21 North 4th Ring Rd., Haidian Dist., Beijing 100190, China, zhangyonglin@mail.ioa.ac.cn), Yupeng Tai, Diya Wang, Haibin Wang, Jun Wang, Lixin Wu (Inst. of Acoust., Chinese Acad. of Sci., Beijing, China), and Fabrice Meriaudeau (Institut de Chimie Moléculaire, Unité Mixte de Recherche, Ctr. National de la Recherche Scientifique 6302,Université Bourgogne Franche-Comté, Dijon, France)

In recent years, machine learning-assisted underwater acoustic (UWA) communication technology has been proved its promising performance in simulations and ideal training and testing environments. However, given the unique characteristics of real-world UWA communication, offline training-based methods face two major challenges: (1) The severe temporal and spatial variations result in a significant gap between offline training and actual deployment conditions. (2) The availability of high-quality labeled samples is extremely limited, making it difficult to construct an appropriate offline training sample set. To address this, we propose two online learning methods for UWA channel estimation and apply them to an OFDM communication system. The first method employs an unsupervised learning approach, where pilot signals are split and input-output pairs for model construction are formed. Thus, the received real-time data can be applied for the online training directly. The second method adjusts the loss function during the machine learning training phase to embed the learned channel within it. It allows online-sampled data to be trained sequentially by leveraging prior knowledge of transmitted information as the actual learning target. Based on the experimental results of measured underwater acoustic channels, it is demonstrated that the proposed online learning methods can approximate the near-optimal solution.

4:20

1pUW12. Enhancing ship noise reduction: a deep-learning based noise extraction model. Hailun Chu (State Key Lab. of Acoust., Inst. of Acoust., Chinese Acad. of Sci., No. 21 North 4th Ring Rd., Haidian Dist., Beijing 100190, China, chuhailun19@mails.ucas.ac.cn), Haibin Wang, Chao Li, Jun Wang, Yupeng Tai, and Yonglin Zhang (State Key Lab. of Acoust., Inst. of Acoust., Chinese Acad. of Sci., Beijing, China)

In active sonar detection systems, the self-radiated noise from the platform itself is a primary factor that interferes with system performance. The platform self-noise exhibits non-Gaussian characteristics, comprising line spectrum and continuous spectrum, where line spectrum exhibits higher intensity and greater frequency stability. To reduce the ship noise, this study introduces a deep-learning based time-domain model with an encoder-separator-decoder architecture. In detail, the encoder and decoder modules utilize learnable convolutional layers to generate distinguishable features with a symmetrical structure. For separator module, we observe that noise components prominently predominate within the noisy signal in low signal-to-noise ratio (SNR) scenarios. The extraction of noise components is typically more tractable than of signal components. Therefore, in contrast to conventional frameworks that focus on extracting the signal subspace, this module is dedicated to extracting the noise subspace. Besides, the modified scale invariant SNR (SI-SNR) is proposed as the loss function for model optimization, which yields better performance than non-noise extraction structure. The robustness of the model is validated on the DeepShip dataset with four ship categories, demonstrating that the model consistently outperforms baselines, including matched filter, in all ship categories.

4:35

1pUW13. Numerical predictions of underwater radiated noise from a non-cavitating model-scale propeller. Duncan McIntyre (Mech. Eng., Univ. of Victoria, 3800 Finnerty Rd., Victoria, BC V8P 5C2, Canada, dmcintyre@uvic.ca), Mohammad-Reza Pendar (Mech. Eng., Univ. of Victoria, Victoria, BC, Canada), Shameem Islam (Ocean, Coastal and River Eng., National Res. Council of Canada, St. John's, NF, Canada), and Peter Oshkai (Mech. Eng., Univ. of Victoria, Victoria, BC, Canada)

Propeller-induced acoustic noise from marine vessels is the largest source of anthropogenic underwater radiated noise (URN) and a significant threat to marine ecosystems. Under typical operation, cavitation dominates the URN emissions. Cavitation, which is a pressure-driven phase-change process that results in the violent formation and collapse of vapor bubbles in the wake of the propeller, is often unavoidable during realistic, full-scale operating conditions. However, at a model scale, inducing cavitation requires a depressurized flow facility that makes acoustic measurements difficult due to confinement effects. Numerical simulation is, therefore, appealing as a tool for predicting URN, but the simulation of the cavitation phenomenon and the associated acoustics involves considerable uncertainty and a range of potential sources of error. In propeller operation, cavitation frequently occurs in the core of the vortices shed from the tips of propeller blades. In the present work, we developed a delayed detached-eddy simulation (DDES) of a model-scale ship with the focus on predicting fluctuating pressure due to shed vorticity. The solution is compared to hydrophone measurements from non-cavitating tow-tank experiments. Finally, we numerically introduce cavitation at model scale in the numerical solution and examine its effects.

4:50

1pUW14. Acoustic signature and wake structure investigation of a cavitating marine propeller operating in proximity to a rudder with an optimized leading-edge pattern. Mohammad Reza Pendar (Mech. Eng., Univ. of Victoria, Eng. Office Wing, Rm. 248, 3800 Finnerty Rd., Victoria, BC V8P 5C2, Canada, pendar@uvic.com), Duncan McIntyre, and Peter Oshkai (Mech. Eng., Univ. of Victoria, Victoria, BC, Canada)

The present study implements high-fidelity numerical modeling to investigate the cavitating flow around a marine propeller operating upstream of a rudder with an optimized wavy leading-edge (WLE), based on a NACA 634-021 profile bio-inspired by a pectoral flipper of a humpback whale (Megaptera novaeangliae). The aim of the study work is to identify the acoustic signature and wake structure of the propeller-rudder system, comparing it to that with a straight leading-edge (SLE) rudder. The propeller (INSEAN E779A model) operated under diverse marine maneuvering conditions (rudder angles of attack $\alpha = 0°$, $10°$, and $20°$) with three distinct leading-edge patterns of the rudder. Large eddy simulations (LES) in conjunction with the Sauer cavitation method and the compressive volume of fluid (VOF) model were utilized to simulate the unsteady cavitating flow using the OpenFOAM platform. Additionally, the Ffowcs Williams–Hawkings (FW-H) acoustic analogy, continuous wavelet transform (CWT), and fast Fourier transform (FFT) were employed to predict and analyze the hydroacoustic response. We propose an optimized propeller-rudder configuration for minimizing the radiated sound levels, thus mitigating the harmful effects of noise pollution on marine ecosystems, while maintaining high propulsive efficiency over a wide range of operating conditions.

5:05

1pUW15. Effect of oceanographic fluctuations on geo-acoustic inversions and source localization using ships of opportunity. 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)

Several merchant ship-radiated noise events were recorded in two separate seabed characterization experiments in the New England Mudpatch region in 2017 (SBCEX17) and 2022 (SBCEX22). These shallow water acoustic experiments were conducted under different oceanographic conditions, resulting in fluctuations in sound propagation. Several statistical inference approaches are used to invert geo-acoustic parameters such as sound speed and density in a mud over sand sediment (See https://doi.org/10.1121/10.0008419). In this work, we explore the effect of sound speed profile fluctuations in the water column on geo-acoustic inversions in the 360–1100 Hz frequency band. Inversions based on measured and simulated data using acoustic models are provided to support our findings. The results demonstrate that sediment parameters become more sensitive to oceanographic variations as the frequency increases and, therefore, sound speed profile fluctuations in the water column need to be taken into close consideration for more accurate geo-acoustic inversions. [Work supported by ONR].

5:20

1pUW16. Application of attention-based transformer for ship-of-opportunity spectrogram prediction. Corey E. Dobbs (Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, cedobbs@byu.edu), Tracianne B. Neilsen (Phys. and Astronomy, Brigham Young Univ., Provo, UT), David P. Knobles (Phys., Knobles Sci. and Anal., Austin, TX), and William Hodgkiss (Marine Physical Labs., Scripps Inst. of Oceanogr., San Diego, CA)

Passive sonar is a useful tool for underwater acoustics that can be used to detect ships or other sound sources in the ocean. The input signals into the sonar system in turn can also be used to make inferences about the ocean environment. In recent work, ship noise has been used to infer seabed information from 15–20 minute spectrograms, with CPA in the center of the time window. Waiting for 15–20 minutes to record the full time window slows down real-time inference efforts. In this work, our goal is to complete ship-of-opportunity (SOO) spectrograms given the first few minutes. Our approach is to use a self-supervised attention-based transformer, which has been found to be effective for machine learning problems, particularly for natural language processing. The transformer is trained on simulated SOO spectrograms and tested on spectrograms from the Seabed Characterization Experiment in 2017 in the New England Mud Patch. The resulting predicted spectrograms contain the key features of the measured spectrograms over the full time window. If successful, this work may allow for real-time applications of ship detections and seabed inferencing methods. [Work supported by the Office of Naval Research, Grant N00014-22-12402.]

Session 2aAAa

Architectural Acoustics and ASA Committee on Standards: Show Your Data: Architectural Acoustics Metrics I

Ana M. Jaramillo, Cochair
Ahnert Feistel Media Group, 8717 Humboldt Ave. N, Brooklyn Park, MN 55444

Bruce Olson, Cochair
Olson Sound Design LLC, 8717 Humboldt Avenue, N, Brooklyn Park, MN 55444-1320

Invited Papers

7:55

2aAAa1. Non-standardized room acoustic metrics. Michael Vorlaender (IHTA, RWTH Aachen Univ., Kopernikusstr. 5, Aachen 52074, Germany, mvo@akustik.rwth-aachen.de)

The first edition of ISO 3382 was adopted in 1997. It already listed T30, Early Decay Time, Clarity, Definition, Center time, Lateral fraction and IACC as relevant room acoustic parameters. They are still up to date, only supplemented by the division into two dimensions of the spatial impression and by the stage support. In the last decade, new research has been conducted on sound perception in concert halls and the results seem to confirm that loudness, reverberation, clarity (intelligibility) and localization are the most important. More recently, auditory-visual interaction, loudness decay and directional reverberation have been found to be factors that describe dimensions of the overall experience of music in a concert hall. This paper highlights the challenges associated with some of the standard parameters and the opportunities presented by newly developed metrics.

8:15

2aAAa2. Establishing a definition for maximum hourly sound level. Samantha Rawlings (Veneklasen Assoc., 1711 16th St., Santa Monica, CA 90404, srawlings@veneklasen.com), John Lo Verde, and Wayland Dong (Paul S. Veneklasen Res. Foundation, Santa Monica, California, CA)

The use of a maximum hourly sound level is common within modern standards and design requirements, referenced in ANSI and ASHRAE standards, CHPS and LEED design guidelines, and is included in proposed language for draft ANSI standards, to name a few. Due to its wide adoption, its use is proscribed for education, hospitality, residential, retail, office, affecting nearly every design sector. The problem with this is that the metric lacks formal definition, leaving designers and evaluators without an agreed-upon approach to determine if an assessment is sufficient. Since 2014, the authors have presented multiple works assessing a proposed definition for maximum hourly sound level for vehicular sources. In this work, the authors will be presenting a continuation of the dataset. Long-term rooftop monitoring of the noise from an interstate was used to create an hourly max calculation noise model. The calculations are based on the statistical spread of the data and the learnings can be used to better design the structures for assessing environmental and traffic noise.

8:35

2aAAa3. Which absorption coefficients are correct? Kevin Herreman (Owens Corning, 2790 Columbus Rd., B75, GRANVILLE, Granville, OH 43023, kevin.herreman@owenscorning.com)

Each year absorptive material samples are sent to various accredited acoustic laboratories around the country by manufacturers who want to verify the performance of their products. Results are published as indisputable validation of the product absorption performance by the manufacturer. As no recent interlaboratory study results for ASTM C423—Standard Test Method for Sound Absorption and Sound Absorption Coefficients by the Reverberation Room Method are available, the Owens Corning Acoustic Research Center initiated a private interlaboratory study. A small set of accredited acoustical laboratories tested a set of the original National Voluntary Laboratory Accreditation Program reference panels. The resulting data and differences in the measured absorption coefficients will be reviewed.

8:55

2aAAa4. Recent sound absorption testing for baffles and blades. Kenneth W. Good (Armstrong World Industries Inc., 2500 Columbia Ave., Lancaster, PA 17601, kwgoodjr@armstrong.com) and Zachary A. Bock (Armstrong World Industries Inc., Lancaster, PA)

It's easy for laboratories to make sound absorption measurement for baffles and blades treatment options. The difficult part is to make measurements and report results in a method that can be useable across a wide variety of users. As example, the architectural acoustic community, the specifying community, and marketing teams, all need to communicate effectively to a diverse clientele each with different needs and understanding. This paper will explore the methodology of one company's latest investment to provide applicable baffles and blades data.

9:15

2aAAa5. Using absorption coefficients for baffles in room acoustics simulations. Ana M. Jaramillo (Olson Sound Design LLC, 8717 Humboldt Ave. N, Brooklyn Park, MN 55444, ana.jaramillo@afmg.eu) and Bruce Olson (Ahnert Feistel Media Group, Brooklyn Park, MN)

As consultants, we often find ourselves wondering how to apply sound absorption coefficients on simulated baffles. The reported coefficients are relative to the floor area represented by an assembly of several baffles, and these are often not spaced the same distance that will be used in a specific project. Furthermore, sound transmission through baffles without a hard core is not represented in the simulation. By modeling a lab scenario and reproducing the results of a number of measurements, we intend to create a best practice guide for using these coefficients in simulations.

9:35

2aAAa6. An in-depth look at empirical estimates of assembly performance. Michael Raley (PAC Int., 2000 4th Ave., Canby, OR 97013, mraley@pac-intl.com)

Acousticians often look for acoustical test data for wall and floor/ceiling assemblies that match the assemblies on their projects. However, with the endless possible permutations of assemblies, it can be difficult to find testing for an assembly that is an exact match. As a consultant, I frequently created engineering evaluations that documented my predicted performance of an assembly. As a manufacturer, I now have a say in what assemblies are tested, but I still can't test all the possible permutations. This presentation is an overview of my current approach to assembly estimates, including the development of test series that are specifically intended to produce the data necessary to create good estimates.

9:55

2aAAa7. Trampolines and insulation and data, Oh My!. Joseph Keefe (Ostergaard Acoust. Assoc., 1460 US Hwy. 9 North, STE 209, Woodbridge, NJ 07095, jkeefe@acousticalconsultant.com)

This presentation will discuss sound and vibration data gathered for two separate and unrelated acoustical consultations. The first is a consultation for an adventure park facility (trampolines, arcade games, ball pits, etc.) planned on the upper level of an existing mall; acoustical concerns included vibration from jumping on trampolines and noise from amplified music to adjacent tenant spaces. The second is a comprehensive set of sound level reduction measurements in accordance with ASTM E336 / ASTM E2235, as well as ASTC calculations in accordance with ASTM E413; these results facilitated an investigation of the effect of complete versus partial cavity insulation within partitions.

10:15–10:30 Break

10:30

2aAAa8. A comparative study of modeled and measured room responses. Nicolaus T. Dulworth (Threshold Acoust., 141 W Jackson Blvd Ste 2080, Chicago, IL 60604, ndulworth@thresholdacoustics.com), Shane J. Kanter (Threshold Acoust., Chicago, IL), and Carl P. Giegold (Threshold Acoust. LLC, Evanston, IL)

At the heart of Brown University's Lindemann Performing Arts Center is a shape-shifting hall capable of supporting a wide range of performance types. Acoustic simulation played a crucial role in optimizing each of the hall's five primary configurations during design. Leading up to the opening events 2023 and 2024, the lengthy commissioning and tuning process provided an invaluable opportunity to measure subtle and drastic changes in the hall's form and materiality for comparison to the predictive simulation. Drawing from the Lindemann set, the paper compares modeled and measured data from the PAC to that from other projects designed using similar tools. The paper identifies points of uncertainty during design and delves into commonalities and differences between ray-based simulation and measurement in performance spaces of varying form. This paper will address the inherent complexities of approximating acoustic behavior with side by side comparison of measurement and simulated data, we will share insights that inform our future modeling process.

10:50

2aAAa9. Measurements using three methods. K. Anthony Hoover (McKay Conant Hoover, 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362, thoover@mchinc.com) and Henry Ashburn (McKay Conant Hoover, Westlake Village, CA)

Room responses of various spaces were measured using three methods in each: interrupted pink noise using full-range loudspeakers, impulse using balloon pops, and sine sweep using dodec loudspeaker array. Our primary interest was to compare the results from each, as well as to better understand their relative advantages and disadvantages. Results will be shown and discussed.

11:10

2aAAa10. Comparison of reverberation times and average absorption coefficients in classrooms, based on *in situ* measurements versus calculated from material databases as during design. Sanjay Kumar (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, Omaha, NE) and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, PKI 100C, 1110 S. 67th St., Omaha, NE 68182-0816, lwang4@unl.edu)

Reverberation time (RT) is an acoustic descriptor used widely by acousticians for characterizing indoor acoustic environments, as specified in standards such as ANSI S12.60 "Acoustical Performance Criteria, Design Requirements, and Guidelines for Schools". It is often calculated in the design phase for simple spaces by applying equations like Sabine's or Eyring's, requiring information on room dimensions and the surface areas and acoustic absorption coefficients across frequency of room materials. In this project, measured RTs from a recent survey of over 200 K-12 classrooms are compared against the RTs predicted across speech frequency octave bands using Sabine's and Eyring's equations. Agreement between predicted and measured values varies based on the octave band and material selection. The working group currently revising ANSI S12.60 is considering providing guidance on the desired average absorption coefficient in classrooms, rather than reverberation time, so this study also compares the average absorption coefficients for the classrooms. The coefficients are back-calculated from the measured RTs using Sabine's and Eyring's equations, and compared against those calculated from surface dimensions and commonly available material absorption coefficients.

Contributed Papers

11:30

2aAAa11. Exploring variations to ASTM E336 directional sound source positioning. Pier-Gui Lalonde (Integral DX Eng. Ltd., 907 Admiral Ave., Ottawa, ON K1Z 6L6, Canada, pier-gui@integraldxengineering.ca) and Gregory E. Clunis (Integral DX Eng. Ltd., Ottawa, ON, Canada)

This paper explores variations to the procedures presented in ASTM E336 *Standard Test Method for Measurement of Airborne Sound Attenuation between Rooms in Buildings*. Per the Test Standard, directional sound sources must be aimed into corners away from the test partition. For source rooms horizontally adjacent to occupied spaces on all sides, multiple loudspeaker positions are needed to measure noise isolation in all directions. Measurements made in 4 office spaces are presented, each made in two ways: firstly in compliance with the Standard; and secondly, with sound sources orientated and operated differently, to explore whether using a single configuration of directional loudspeakers can produce similar results. Variations in the results are discussed, and suggestions are provided for future work.

11:45

2aAAa12. Comparing measured sound strength to theoretical sound strength for rooms of varying size. Joshua Dunham (Stantec, 720 Third Ave., Seattle, WA 98104, Joshua.Dunham@stantec.com)

ISO Standard 23591 shows how the sound pressure level in a room can be estimated with the following two pieces of information: the sound power of sources in that room, and the sound strength G of that room as defined in ISO 3382-1. In an earlier study, the theory connecting room volume V, reverberation time T, and sound strength G was summarized, and a comparison of measured G to this theory was explored for a wide range of rooms. That test setup used Odeon software. In the present study, a different test setup is used implementing Dirac software to expand G testing to additional rooms of various sizes for evaluation under the same mathematical model. After a description of the equipment using Dirac software is discussed, a brief review of the additional rooms in which sound strength was measured is provided. Both measured and theoretical sound strength results are presented for these rooms. The difference between measured G and theoretical G using the mathematical model is analyzed statistically both for the Dirac test setup and compared to the results from the previous study.

Session 2aAAb

Architectural Acoustics: Student Design Competition (Poster Session)

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

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

Design Scenario: A college with a very strong music and vocal program intends to construct a new 1,200-seat performance hall primarily for opera. Although the main purpose of the hall is to support their opera program, the hall will also be used for speaking engagements by the school's president and other invited speakers.

Entries to the 2024 competition will be posted in this session for viewing.

TUESDAY MORNING, 14 MAY 2024 ROOM 204, 8:00 A.M. TO 11:10 A.M.

Session 2aAB

Animal Bioacoustics, Computational Acoustics, Acoustical Oceanography, Underwater Acoustics, and Signal Processing in Acoustics: Data to Information, Navigating the Application of Acoustic Data for Conservation I

Megan F. McKenna, Cochair
Cooperative Institute for Research in Environmental Sciences, University of Colorado Boulder, 442 Gibson Ave., Pacific Grove, CA 93950

Carrie Wall, Cochair
University of Colorado, 325 Broadway, Boulder, CO 80305

Chair's Introduction—8:00

Invited Papers

8:05

2aAB1. A primer on terrestrial passive acoustic monitoring data management: Best practices and guidelines. Dena J. Clink (K. Lisa Yang Ctr. for Conservation Bioacoustics, Cornell Univ., 159 Sapsucker Woods Rd., Ithaca, NY 14850, djc426@cornell.edu), Chris Pelkie, Russell A. Charif, and Holger Klinck (K. Lisa Yang Ctr. for Conservation Bioacoustics, Cornell Lab of Ornithology, Cornell Univ., Ithaca, NY)

Recent advances in recording technology and data storage capabilities have revolutionized how we monitor vocal animals and their habitats. Recently, there has been an explosion of interest in the use of terrestrial passive acoustic monitoring (PAM) which relies on autonomous acoustic recording units (ARU), but terrestrial PAM studies are limited by a lack of standardized protocols for data collection, analysis, and archiving. Here, we draw on collective experiences of recording, analyzing, and archiving, hundreds of years of PAM data to provide guidelines and recommendations for PAM practitioners. First, we highlight some of the major pitfalls in field data collection, including inappropriate recording settings for the research question and incomplete metadata. We then provide recommendations for data management, including suggestions for backing up, compressing, and archiving the data. Our goal is to provide the community with standardized working guidelines that will help facilitate comparative, large-scale terrestrial studies using PAM.

8:25

2aAB2. Using acoustic monitoring to evaluate the co-benefits of urban restoration. Rachel T. Buxton (Inst. of Environ. and Interdisciplinary Sci., Carleton Univ., 1125, Colonel By Dr., Ottawa, ON K1S5B6, Canada, Rachel.buxton@carleton.ca), Christopher Dennison (Dept. of Biology, Carleton Univ., Ottawa, ON, Canada), Katherine Brown, and Amber L. Pearson (CS Mott Dept. of Public Health, Michigan State Univ., Flint, MI)

Restoring urban greenspaces is an increasingly common nature-based solution aiming to bolster biodiversity while benefiting human wellbeing. However, there is limited understanding of the co-benefits, including potential trade-offs and synergies between outcomes for nature and people. Although passive acoustic recording may be a valuable outcome monitoring tool, appropriate analytical techniques are unknown. We explore how different acoustic analyses capture patterns in species diversity and harmful noise pollution using 2021–2023 acoustic data from Detroit, Michigan. Data were collected as part of a longitudinal panel study aimed at quantifying the effects of park restoration on physical activity, stress, and cardio-metabolic health. We used AudioMoth recording devices at 100 sites: 10 in the neighborhood around each of 5 restored parks, where native plants were seeded, and 10 around each of 5 unmaintained control parks, matched to restored parks based on specified conditions. Recordings were analyzed by manually identifying bird species and noise in a subset of spectrograms, using an artificial neural network (BirdNET), and using acoustic indices. We found that using acoustic indices in a model could accurately predict bird species diversity and richness, but not community composition. We compared outputs from each type of analysis with mental health indicators of study participants.

8:45

2aAB3. Navigating the bioacoustic landscape: Standardization and interoperability of acoustic data products. Marie A. Roch (Dept. of Comput. Sci., San Diego State Univ., 5500 Campanile Dr., Dept. of Comput. Sci., San Diego State University, San Diego, CA 92182-7720, marie.roch@sdsu.edu), Simone Baumann-Pickering (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA), Douglas Gillespie (Sea Mammal Res. Unit, Univ. of St Andrews, St Andrews, Scotland, United Kingdom), Jasper Kanes (Sci. Services, Ocean Networks Canada, Victoria, BC, Canada), Katherine Kim (Greeneridge Sci., Inc., Santa Barbara, CA), Holger Klinck (K. Lisa Yang Ctr. for Conservation Bioacoustics, Cornell Lab of Ornithology, Cornell Univ., Ithaca, NY), Xavier Mouy (Passive Acoust. Res. Group, NOAA, Miami, FL), and Ana Sirovic (Norwegian Univ. of Sci. and Technol. (NTNU), Trondheim, Norway)

The importance of bioacoustics for understanding and protecting the natural world has grown significantly over the last few decades. Our capacity to monitor the acoustical landscape has increased with lower cost instrumentation, improved communication, infrastructure, and storage, as well as the development of reliable machine learning methods for analysis. Understanding long-term trends in animal populations as well as the impacts of anthropogenic noise requires the ability to retain and interpret records over decadal time scales, record in detail what type of analysis has been performed and understand when multiple analyses may be combined and when they should not be. This is not possible without a well-defined set of terms and meanings. In this talk, we will provide an overview of the work of Acoustical Society of America's working group S3-SC1-WG7 that has been charged with developing an American National Standards Institute standard for bioacoustics information derived from acoustic recordings. The proposed standard details what information should be recorded for bioacoustics recording and analysis effort, ranging from specification of data to be noted about instrumentation to information gleaned from recordings such as noise levels, animal calls, and acoustic source locations.

9:05

2aAB4. Five years of data collection, student theses, and disseminating information soundscape studies in the Gulf of Tribugá, Colombia. Kerri D. Seger (Appl. Ocean Sci., 2127 1/2 Stewart St., Santa Monica, CA 90404, kerri.seger.d@gmail.com), Christina E. Perazio (Environment and Sustainability, Univ. at Buffalo, Buffalo, NY), Nohelia Farias Curtidor (Fundación Macuáticos Colombia, Bogotá, Colombia), L. V. Huertas-Amaya (Ports, Humpbacks, y Sound In Colombia (PHySIC), Bogotá, Colombia), Maria P. Rey-Baquero (Fundación Macuáticos Colombia, Bogota, Colombia), Daniel Norena (Ports, Humpbacks, y Sound In Colombia (PHySIC), Bogota, Bogotá DC, Colombia), Mar Palanca (MadreAgua Colombia, Valencia, Spain), Astarte Brown (Ecology & Evolutionary Biology, Univ. of California, Santa Cruz, Santa Cruz, CA), Valentina Ramirez Caycedo (Ports, Humpbacks, y Sound In Colombia (PHySIC), Bogotá, Colombia), and Natalia Botero-Acosta (Fundación Macuáticos Colombia, Bogotá, Colombia)

In 2018, the Ports, Humpbacks, y Sound In Colombia (PHySIC) Project began to record soundscapes in the Gulf of Tribugá, Northern Colombian Pacific, for the first time. This was of interest to local conservation groups who were against building a megaport in the native mangrove habitat that provides livelihoods for the people of Chocó. Five years later, no port has been built, documentaries about the ecosystem have won awards, before-after/control-impact studies have been done on fish and humpback whale acoustic behavior, delphinid call catalogs have been assembled, propagation modeling has mapped acoustic connectivity, and a new vocalization category for humpback whales has been discovered. The work has been executed through internships, and undergraduate and masters theses at various universities. This project has a variety of data-to-information challenges: storing data for access across various countries, navigating data analysis and dissemination in two languages, transferring knowledge as students graduate, and tailoring results for general conservation groups and policymaking audiences. To celebrate the colleagueship and academic successes of the first five years of PHySIC, this talk will describe the list of discoveries and studies done in the Gulf of Tribugá, which was named a UNESCO World Heritage site in the summer of 2023.

9:25

2aAB5. From sounds to science on public lands: Using emerging tools in terrestrial bioacoustics to understand national park soundscapes. Cathleen Balantic (National Park Service, 1201 Oakridge Dr. Ste. 100, Fort Collins, CO 80525, cathleen_balantic@nps.gov)

The National Park Service manages over 400 units across the United States, each with unique resource protection needs. Many parks have natural resource conservation priorities that include the protection of natural soundscapes and the sound-producing species who contribute to them. It is challenging, however, to translate terabytes of raw audio into applied information for the diverse research and management questions posed by individual parks. Recent advances in terrestrial bioacoustics are making this process more tractable, particularly as BirdNET has emerged as an extensible off-the-shelf tool equipped to detect diverse assemblages of sound-producing species across varied park ecosystems. We illustrate the applied utility of using and adapting this emerging tool in combination with open code and visualizations to answer multi-taxa questions about occurrence, phenology, and wildlife responses to management. This work demonstrates both the iterative nature and challenges inherent to keeping a federal agency astride the leading edge in a rapidly progressing, data-heavy field, amid a backdrop of global change.

9:40–9:55 Break

9:55

2aAB6. Methods for managing soundscape quality: Evidence from Denali National Park and Preserve. Lauren A. Ferguson (Recreation Management and Policy, Univ. of New Hampshire, 4 Library Way, Hewitt Hall, Durham, NH 03824, lauren.ferguson@unh.edu), Peter Newman (Recreation, Parks and Tourism Management, The Penn State Univ., State College, PA), Megan F. McKenna (Cooperative Inst. for Res. in Environ. Sci., Univ. of Colorado Boulder, Pacific Grove, CA), Davyd H. Betchkal (Natural Sounds and Night Skies Div., National Park Service, Denali Park, AK), Zach Miller (Visitor Use Management, National Park Service, Lakewood, CO), Rose Keller (Norwegian Inst. for Nature Res., Lilehammer, Norway), Kurt Fristrup (Natural Sounds and Night Skies Div., National Park Service, Fort Collins, CO), and Derrick Taff (Recreation, Parks and Tourism Management, The Penn State Univ., State College, PA)

The purpose of the United States national park system is to preserve resources unimpaired for the enjoyment of future generations. At Denali National Park (DENA), high levels of air tour traffic and air transportation present challenges to preserving wilderness character. Accordingly, the park attached high priority to measuring visitor impressions of soundscape quality. Visitors in four distinct settings rated the acceptability of five randomly chosen recordings of aircraft noise. We used a cumulative link mixed model to fit visitor's response to acoustic and nonacoustic factors. The negative effects of increased noise were moderated for visitors who were interested in taking an air tour. To be conservative, we examined visitors uninterested in an air tour and found the probability of rating aircraft noise as unacceptable at 54 dB LAeq, 30 s or higher was 26%. For context, noise above 55dB is incompatible with outdoor, rural activities. Visitor response predictions were joined to a spatial model of aircraft noise propagation to create a map depicting the acceptability of aircraft noise in DENA's frontcountry area. The map can be used to forecast the range of soundscape conditions park visitors are exposed to, inform hiking recommendations for visitors, and evaluate park management zones.

10:10

2aAB7. Investigation of duty cycles for measuring activity in passive acoustic bat monitoring. Aditya Krishna (Elec. and Comput. Eng., Univ. of Washington, 1013 NE 40TH St., Henderson Hall (APL) Rm. No. 459, Seattle, WA 98105, adkris@uw.edu) and Wu-Jung Lee (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Echolocating bats are important bioindicators that can be monitored effectively using passive acoustic monitoring (PAM) techniques. In PAM, duty-cycle-based temporal subsampling is often used to collect data at ON/OFF intervals to circumvent the limitations of recorder battery and storage capacity to enable long-term monitoring. However, potential bias introduced by temporal subsampling has not been systematically investigated for bat monitoring. Here, we use continuous audio recordings from the Union Bay Natural Area in Seattle in Summer 2022 to simulate the effects of temporal subsampling using different duty cycle parameters. We detected bat calls automatically using a deep learning model [Aodha *et al.*, 2022, BioRxiv] and calculated three metrics as proxy for bat activity: number of calls, Activity Index (AI), and Bout Time Percentage (BTP). We found that although the number of calls and AI are more readily computable using the detected calls, BTP is likely a more accurate measure that relies less on the performance of automated call detection methods. In addition, reduced sampling for only a portion of the night (e.g., 4 hrs) was generally inadequate for capturing bat activity. Our results suggest that considering species-specific acoustic characteristics is crucial for reducing sampling bias for PAM of bats.

10:25

2aAB8. The past, present and future of underwater passive acoustic monitoring. John Hildebrand (Scripps Inst. of Oceanogr., Univ. of California San Diego, Mail Code 0205, La Jolla, CA 92093, jhildebrand@ucsd.edu), Sean Wiggins, Simone Baumann-Pickering, Kait Frasier (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA), and Marie A. Roch (San Diego State Univ., San Diego, CA)

We review the history of underwater passive acoustic monitoring and predict the future trajectory. Over the past three decades, advances in digital data storage capacity and low power electronics have made it possible to collect autonomous long-term broadband passive acoustic monitoring data. Concomitant advances have taken place in data analysis and curation. Standardized spectra are an excellent first-step in analysis, calculated for all data using multiple frequency bands. These spectra allow *in-situ* instrument calibration based on an understanding of underwater ambient noise. Software for efficient manual scanning and signal discovery is used for verification and error estimation. Automatic detectors/classifiers are obtained from both supervised and unsupervised machine learning. Detections are aggregated into a database that allows the combination of multiple datasets and association with environmental or other data. Future work will increasing use of arrays of acoustics sensors, the integration of passive acoustics with other sensors, and machine learning that integrates these data streams to provide better understanding of anthropogenic, biological and physical processes in the ocean.

10:40

2aAB9. Connecting separate monitoring programs through the SoundCoop. Carrie Wall (Univ. of Colorado, 325 Broadway, Boulder, CO 80305, carrie.wall@noaa.gov), Leila Hatch (ONMS, NOAA, Scituate, MA), Sofie Van Parijs (NEFSC, NOAA, Woods Hole, MA), Rob Bochenek (Axiom Data Sci., Anchorage, AK), Catherine Berchok (AFSC, NOAA, Seattle, WA), Genevieve Davis (NEFSC, NOAA, Woods Hole, MA), Peter Dugan (NUWC, Middeltown, RI), Kait Frasier (Scripps Inst. of Oceanogr., La Jolla, CA), Samara Haver (Oregon State Univ., Newport, OR), Megan F. McKenna (Cooperative Inst. for Res. in Environ. Sci., Univ. of Colorado Boulder, Pacific Grove, CA), Jennifer Miksis-Olds (Univ. of New Hampshire, Durham, NH), Clea Parcerisas (Flanders Marine Institute/Ghent Univ., Orstend, Belgium), Dimitri Ponirakis (Cornell Univ., Ithaca, NY), Timothy Rowell (SEFSC, NOAA, Beaufort, NC), John P. Ryan (Res., MBARI, Moss Landing, CA), and Karolin Thomisch (Alfred Wegener Inst. for Polar and Marine Res., Bremer, Germany)

Passive acoustic monitoring (PAM) data collection has been growing exponentially, resulting in petabytes of data that document ocean soundscapes, how they change over time, and what animals use these ecosystems at varying timescales. Efficiently extracting this critical information and comparing it to other datasets in the context of ecosystem-based management is a Big Data challenge that traditional desktop processing methods cannot address. The curation, management, and dissemination of PAM datasets is another challenge in need of collaborative progress. To meet these exigencies, a multi-agency funded Sound Cooperative (SoundCoop) project is building community-focused, national cyberinfrastructure capability for PAM data to promote improved, scalable and sustainable accessibility and applications for management and science. Driven by partnerships and framed by four case studies, the SoundCoop has established guidance on the standardized processing of sound level metrics using free software toolkits and begun developing core cyberinfrastructure components that future PAM projects can leverage. U.S. and international scientists contributed PAM data collected across 10 long-term monitoring projects to operationalize the production of hybrid-millidecade spectra across a diversity of labs/instruments. Collectively, the contributed data demonstrate the value of standardized processing that enables the creation of comparable results from disparate monitoring efforts.

10:55

2aAB10. Understanding variability in coral reef soundscapes to enable tool development and early warnings of ecosystem change. T. Aran Mooney (Biology Dept., Woods Hole Oceanographic Inst., 266 Woods Hole Rd., Marine Res. Facility 227 (MS# 50), Woods Hole, MA 02543-1050, amooney@whoi.edu), Sierra Jarriel, Nathan Formel, Nadége Aoki (Biology Dept., Woods Hole Oceanographic Inst., Woods Hole, MA), Seth A. McCammon (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA), Amy Apprill (Marine Chemistry and Geochemistry, Woods Hole Oceanographic Inst., Woods Hole, MA), David Mann (Loggerhead Instruments, Sarasota, FL), and Yogesh Girdhar (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

Coral reefs harbor some of the highest biodiversity on Earth. Their rich soundscape is vital to inhabiting animals and can provide a means of tracking community health. Reefs are facing immense climate stressors and are declining rapidly. Passive acoustic monitoring can provide a powerful, scalable tool for stakeholders, but these data are not evaluated on actionably relevant timescales. Here we present results from a long-term (10-years) acoustic and ecosystem study of multiple coral reefs in the U.S. Virgin Islands. Acoustic measurements (including snap rates, sound levels in low-frequency fish, and high-frequency shrimp bands) were made in-tandem with traditional, diver-based benthic and fish visual surveys. These key baseline data over multiple spatial and temporal scales provide a means of examining how soundscape changes are driven by climate-related stressors, and an important testbed for developing new analyses and tools. Here, we show how physical changes (temperature, light, coral disease) can influence the cue rates of snapping shrimp and fish, and apply novel tools including real-time recorders and underwater robots listening for biodiversity. These initial steps were then implemented into a novel rapid acoustic assessment, a key step toward providing actionable information to stakeholders monitoring for habitat change and weighing resource management.

Session 2aAO

Acoustical Oceanography: Topics in Acoustical Oceanography I

John A. Colosi, Cochair
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Julien Bonnel, Cochair
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Contributed Papers

8:30

2aAO1. Theoretical model for correlation analysis of wideband signal received by two vertical arrays and retrieval of modal dispersion curves in shallow water. Marina Yarina (Marine Geoscience, Univ. of Haifa, Abba Khoushy 199, Haifa 3498838, Israel, myarina@campus.haifa.ac.il), Boris Katsnelson (Marine Geosciences, Univ. of Haifa, Haifa, Israel), and Oleg A. Godin (Phys., Naval Postgrad. School, Monterey, CA)

Goal of this work is to carry out theoretical modeling extraction of characteristics of layered bottom (sound speeds $c_{;2}$ in layer, c in water, $c_{;b}$ in half-space and layer's thickness h using construction of correlation function (matrix) of wideband synthetic sound signal radiated by the moving source and imitating shipping noise. This work's motivation was experiment carried out by authors in shallow water. Combination of two vertical line arrays acts as a diffraction grating with 2N hydrophones that uses interference pattern in vertical and horizontal directions and gives more information than one-dimensional array or single receiver. Our waveguide model constitutes water layer above low-speed thin layer ($c_{;2} \ll c$, h is about a few meters) and half space ($c_{;b} > c$). In this waveguide there are two sorts of modes: in dependence on frequency and thickness of sediment's layer– modes with vertical dependence distributed in water layer and modes trapped in the near bottom layer. It is shown that correlation function constructed in coordinates (phase speed, frequency) allows us to retrieve dispersion curves and to estimate critical frequencies—determining creation of trapped modes, which in turn give parameters of layered bottom. [Work was supported by ISF, Grant No. 946/20.]

8:45

2aAO2. Kauai Beacon receptions and analysis with open-access hydrophones in the North Pacific Ocean. John Ragland (Elec. and Comput. Eng., Univ. of Washington, 185 W Stevens Way NE, Seattle, WA 98195, jhrag@uw.edu), Nicholas Durofchalk (Phys., Naval Post Graduate School, Monterey, CA), David Dall'Osto (Appl. Phys. Lab. at the Univ. of Washington, Seattle, WA), Kay L. Gemba (Phys., Naval Post Graduate School, Monterey, CA), and Shima Abadi (Univ. of Washington, Seattle, WA)

The Kauai Beacon began regularly transmitting a 75 Hz, maximum length sequence encoded, signal that has been received by hydrophones as far as 4000 km from the source. The received signal, which has travelled across the entire ocean basin, contains integrated environmental information about the full path between the transmitter and the receiver. This allows us to simultaneously understand how the environment effects long-range acoustic propagation, and to use the long-range acoustic propagation to estimate environmental variables such as path-averaged ocean temperature. In this talk, we present continued work to analyze the receptions of the Kauai Beacon using open-access ambient sound data recorded in the North Pacific Ocean. We present long-term trends of acoustic arrival times, the spread of the acoustic energy, and frequency shift of the signal due to mooring motion, or ocean state fluctuation. [Work supported by ONR.]

9:00

2aAO3. Acoustic tomography in a deep stratified lake; toward reliable estimates of large-scale currents by differential travel time. John C. Wells (Civil & Environ. Eng., Ritsumeikan Univ., Noji Higashi 1-1-1, Tricea I, Civil & Env'l Eng., Kusatsu, Shiga 525-8577, Japan, jwells@se.ritsumei.ac.jp) and Naokazu Taniguchi (Graduate School of Adv. Sci. and Eng., Hiroshima Univ., Higashi-Hiroshima, Japan)

Lakes worldwide are under increasing stress. For example climate change tends to strengthen stratification, promoting hypoxia and other nefarious effects. Targeting scientific investigation of such changes, and operational monitoring, we have been developing the lacustrine application of Coastal Acoustic Tomography ("CAT") at Lake Biwa. The largest lake in Japan, Biwa supplies water to 14 million, and has a mean depth in its main basin of 41 m. Since November 2016, we have used M-sequences (5 kHz carrier) to execute the first successful tests of CAT at multikilometer ranges in any lake, with multiweek deployment of 3 (5) stations in Nov 2017 (2018). In 2021, we confirmed reciprocal transmission in spring through early summer between two stations at 7 km. In autumn, we often observed two distinct groups of arrivals. Ray-tracing simulations, based on sound-speed distributions from realistic hydrodynamic simulations, revealed a warm "surface channel" and cold "deep channel" whose predicted arrival times corresponded well with the observations. In continuing work to be reported, we are targeting reliable measurement of raypath-averaged currents by differential travel time, as already achieved by the second author's team for the stronger tidal currents in the Seto Inland Sea at ranges of 3 km. Preliminary processing has revealed continuous evolution of phase difference between reciprocal receptions. By exploiting such phase differences we hope to achieve a useful level of accuracy in differential travel time despite the weak currents.

9:15

2aAO4. Trans-dimensional joint inversion of modal amplitude and arrival-time data for geoacoustic properties in layered muddy sediments. Yong-Min Jiang (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC V8W2Y2, Canada, minj@uvic.ca), Stan Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada), and Julien Bonnel (Woods Hole Oceanographic Inst., Woods Hole, MA)

Long-range sound fields in a shallow-water waveguide can be expressed as a summation of several dispersive propagating modes. Modal arrival times as a function of frequency, extracted via warping time-frequency analysis, have been used by many researchers to estimate sound speeds and densities for layered models of the seabed sediments. This paper explores joint inversion of modal amplitude spectra together with arrival times to also estimate sediment-layer attenuations. To consider the common case in underwater-acoustic applications where the source amplitude spectrum is unknown, we include this spectrum as unknown parameters in the inversion, formulated implicitly within the likelihood function. Trans-dimensional

Bayesian inversion is applied to estimate marginal probability profiles of the geoacoustic properties, providing quantitative uncertainties. The data considered here were collected on the New England Mud Patch during the 2017 Seabed Characterization Experiment. [Work supported by the Office of Naval Research.]

9:30

2aAO5. Acoustic propagation through a biological deep scattering layer at the New England shelf break. Natalie Kukshtel (Appl. Ocean Phys. & Eng., Woods Hole Oceanographic Inst., 98 Water St., Woods Hole, MA 02543, nkukshtel@whoi.edu), Andone C. Lavery (Appl. Ocean Phys. & Eng., Woods Hole Oceanographic Inst., Woods Hole, MA), and Ying-Tsong Lin (Scripps Inst. of Oceanogr., La Jolla, CA)

The New England Shelf Break is a dynamic ocean environment with strong spatial variability due to complex physical processes and interactions with warm core rings originating from the Gulf Stream. An acoustic autonomous underwater vehicle (AUV) was deployed in the New England Shelf Break Acoustics (NESBA) experiment in May 2021 to investigate environmental variability and its effect on sound propagation. The AUV system was comprised of a modified REMUS 600 vehicle with a hull-mounted 3.5 kHz transducer, a downward-facing EK80 echosounder, and a towed linear hydrophone array. Shipboard EK80 data measured during the AUV mission showed the presence of a biological deep scattering layer. Preliminary analysis of AUV source signals passing through the layer suggested significant attenuation compared to signals not passing through the layer. To investigate this further, a parabolic equation (PE) model with range-dependent attenuation patches in the water column was constructed to replicate the biological attenuation through the scattering layer. The model uses estimates of absorption and scattering derived from shipboard and AUV EK80 data. Through this study we aim to better understand the scattering effects of this deep biological layer on acoustic propagation. [Work supported by the Office of Naval Research.]

9:45–10:00 Break

10:00

2aAO6. Mid-frequency acoustic tracking of breaking waves. Ryan Saenger (Scripps Inst. of Oceanogr., Univ. of California San Diego, 9500 Gilman Dr., San Diego, CA 92093, rsaenger@ucsd.edu), Luc Lenain, William Kuperman, and William Hodgkiss (Scripps Inst. of Oceanogr., Univ. of California San Diego, San Diego, CA)

Large surface wave breaking events in deep water are acoustically detectable by beamforming at 5-6 kHz with a mid-frequency planar array located 130 m below the surface. Due to the array's modest 1 m horizontal aperture, wave breaking events cannot be conveniently tracked by beamforming alone. However, their trajectories can be estimated by splitting the array into left-right and upper-lower sub-array pairs, beamforming each sub-array toward the source, and computing the left-right and upper-lower beam cross-correlations. The cross-correlations can be used to estimate the relative time delay between sub-arrays of noise generated by breaking waves. The time delays map to a source location on the surface that can be tracked over time. Source tracks estimated from the sub-array cross-correlations match the trajectories of breaking waves that are visible in aerial images of the ocean surface above the array. [Work supported by the Office of Naval Research.]

10:15

2aAO7. Listening to the acidity of the ocean: Inversion of passive deep sea acoustic data for pH. Ernst Uzhansky (Oceanogr., Dalhousie Univ., Izhaq Greenboim Str., Apt. 12, (Koren Family), Haifa 3498793, Israel, ernstuzhansky@gmail.com) and David R. Barclay (Oceanogr., Dalhousie Univ., Halifax, NS, Canada)

Ocean acidification is an ongoing concern due to its impact on the marine ecosystem. The volume integrated pH of sea water can be determined from the depth-dependence of ambient sound, which depends on the acoustic absorption properties of seawater. For a wind-driven noise in the ocean over the band 1–10 kHz, two main contributions to sound attenuation are associated with the ionic relaxation of boric acid (<3 kHz), related to pH, and magnesium sulfate (>3 kHz), unrelated to pH. When local winds are strong (>10 m/s), the ambient noise is dominated by locally generated surface noise and has a depth-independent directionality and a weakly frequency and depth-dependent intensity, due to sound absorption. By measuring the attenuation of sound in a wide frequency band, it is possible to estimate pH by comparing the experimentally measured attenuation with an analytical theory of passive acoustic absorption spectroscopy. Measurements of the depth-dependent ambient sound field were carried out in the Philippine Sea, Mariana Trench, and Tonga Trench throughout 2009—2021. The wideband (5 Hz—30 kHz) acoustic data were recorded with untethered free-falling autonomous recording systems carrying two or four hydrophones.

10:30

2aAO8. Acoustic measurements of photosynthetically formed gas bubble distributions in Posidonia oceanica seagrass meadows. Anthony P. Lyons (Univ. of New Hampshire, Durham, NH, anthony.lyons@unh.edu), Angeliki Xenaxi, and Yan Pailhas (Ctr. for Maritime Res. and Experimentation, La Spezia, Italy)

The formation of gas bubbles on seagrass leaves commonly occurs during periods of high productivity in many aquatic ecosystems. Gas bubbles produced by seagrass meadows and subsequently released to the atmosphere can be a major component of primary production that is not quantified by measurements of dissolved oxygen. Acoustic propagation measurements have been explored as a possible tool for estimating photosynthetically relevant properties *in situ*. However, a lack of knowledge of bubble densities and size distributions has hindered acoustic attenuation as a quantitative measurement technique. Here we explore using measurements of acoustic backscattering from gas bubbles in and above Posidonia oceanica seagrass meadows as a more direct method for estimating bubble densities and size distributions. Acoustic backscattering data for this study was collected in September 2023 in Biodola Bay, Elba Island, Italy, using 1i–16 kHz linear frequency modulated pulses. Inversions of the broadband acoustic data yielded log-normal bubble size distributions with mean sizes between 0.4 and 0.5 mm in agreement with previously reported values. Estimated bubble densities yielded predicted attenuations in line with those found by propagation experiments. Results demonstrate the possible use of direct acoustic backscattering measurements as a supplement to dissolved oxygen measurements to better estimate rates of gas production which in turn can be used to better understand seagrass ecosystem properties, such as productivity and health.

10:45

2aAO9. Acoustic observations of individual bubble release events from melting glacier ice in an arctic fjord. Hayden A. Johnson (Scripps Inst. of Oceanogr., Univ. of California, San Diego, 8622 Kennel Way, La Jolla, CA 92037, h3johnso@ucsd.edu), Grant B. Deane (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA), Oskar Glowacki (Inst. of Geophys., Polish Acad. of Sci., La Jolla, CA), M. Dale Stokes (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA), Mandar Chitre, Hari Vishnu (Acoust. Res. Lab., National Univ. of Singapore, Singapore, Singapore), and Elizabeth Weidner (Scripps Inst. of Oceanogr., Univ. of California, San Diego, Durham, NH)

Melting below the waterline of marine-terminating glaciers and ice sheets has the potential to remove ice that is holding back land-based ice upstream, and increase the rate of global mean sea level rise. It is difficult to observe this melting directly, and the physical processes involved remain an active area of study. As glacier ice melts, the air bubbles trapped in the ice are released into the water. The air bubbles are often at greater pressure than the surrounding hydrostatic pressure in the water, causing them to be released explosively and generate sound through resulting monopole oscillations. Studying this sound at the level of individual bubbles provides an opportunity to learn more about the role of the bubbles in affecting heat flux across the ice-water boundary layer. It is also essential for proposed efforts to use acoustic emissions as a tool to quantify glacier-scale submarine melting. We will present and discuss observations made at the ice-water interface of floating pieces of glacier ice in an Arctic fjord. These observations included measurements of the water thermal structure, velocity, and salinity,

as well as ablation of the ice face, made concurrently with acoustic measurements of bubble release pulses from a 2-element hydrophone array.

11:00

2aAO10. Measurements of the frequency dependence of geo-alpha in clayey silt at low frequencies. Orest Diachok (Poseidon Sound, 3272 Fox Mill Rd., Oakton, VA 22124, orestdia@aol.com) and Altan Turgut (Naval Res. Lab., Washington, DC)

Transmission loss (TL) measurements in the Santa Barbara Channel, together with co-located chirp measurements, show that geo-alpha (dB/l) in clayey silt increases approximately linearly with frequency (f) at $f < 3$ kHz. TL was measured between a fixed source and fixed vertical array at 3.7 km at $0.3 < f < 5$ kHz. Geo-alpha was inverted from the TL data at 0.3, 1.1 and 1.9 kHz. Inversion calculations at 1.1 and 1.9 kHz considered both geo-alpha and bio-alpha due to layer of anchovies at 12 m. Inversion calculations at 0.3 kHz considered only the effect of geo-alpha; the effect bio-alpha on TL was small. Geo-alpha was also inferred from chirp sonar measurements at 3.2 and 6.2 kHz at the ends and middle of the TL track. Chirp sonar showed that the bottom consisted of a layer of unconsolidated sediments overlying a layer of sand. Co-located cores revealed that the unconsolidated layer consisted of clayey silt. Bio-alpha data were compared to Biot and VGS theories and the theory of attenuation in suspended sediments (Pierce *et al.*, 2017). The measured frequency dependence is consistent with Pierce *et al.*'s theory. This research was supported by The Office of Naval Research Ocean Acoustics Program.

11:15

2aAO11. Angular dependence of acoustic scattering statistics for rocky outcrops. Lindsey Darling (Oceanogr., Naval Postgrad. School, 1 University Circle, Monterey, CA 93943, lindsey.darling@nps.edu), Derek Olson (Oceanogr., Naval Postgrad. School, Monterey, CA), and Marc Geilhufe (Norwegian Defence Res. Establishment, Kjeller, Norway)

The use of automated detection algorithms in undersea remote sensing and target detection allows for decreased timelines and risk to human assets. However, understanding the performance of these systems requires a thorough understanding of the physical processes that affect the acoustic scattering statistics of the sonar image. Many studies have reported on the scattering statistics of the seafloor generally, and recent work has quantified the dependence of scintillation index on range for sandy seafloors. To date, no studies have examined the angular dependence of acoustic scattering statistics for rocky bottoms, which have more complex spatial texture. This paper analyzes synthetic aperture sonar (SAS) images of rocky outcrops imaged near Bergen Norway to quantify the grazing angle dependency of scattering statistics for rocky seafloors. Outcrops are categorized by texture and analyzed using scintillation index and relative scattering strength with respect to both true grazing angle and a flat seafloor assumption. Comparisons between results utilizing calibrated (beampattern removed) and non-calibrated images will be presented to display the complexity of the grazing angle dependence of scattering statistics for rocky seafloors. SAS imagery provided by the Norwegian Defence Research Establishment. [Funding provided by the Office of Naval Research]

TUESDAY MORNING, 14 MAY 2024 ROOM 212, 7:55 A.M. TO 12:00 NOON

Session 2aBAa

Biomedical Acoustics, Physical Acoustics, Signal Processing in Acoustics, and Structural Acoustics and Vibration: Ultrasound Beamforming and its Applications I

Jian-yu Lu, Chair
Bioengineering, The University of Toledo, 2801 West Bancroft Street, Toledo, OH 43606

Chair's Introduction—7:55

Invited Papers

8:00

2aBAa1. Some thoughts on ultrasound futures. Kai E. Thomenius (Dept. of Radiology, Massachusetts General & Brigham, 74 Van Vranken Rd., Clifton Park, NY 12065, kaius@mac.com)

Development of ultrasound technology began in early 1950s and has maintained a fast pace. In this talk I will review some of this history with the goal of setting the stage for future development. I have opted to split this development chronology into four stages by the nature of the electronics with the culmination of software-based systems. This talk will consider near-term futures that are gained from technologies such as software beamformation. An aid in defining this future will be information gained from the programs of various technical conferences. It is assumed that much of the work presented at such meetings has gone through broader review processes such as reviews of grant proposals and manuscript submissions. A hopefully interesting recommendation that arises from this analysis is towards research for improved understanding of sound/tissue interactions based on received echo data (e.g., channel data) and how this understanding may be used to improve image formation and increase our understanding of pathologies. An aspect that may arise from this is the potential of being able to define an upper limit on performance of ultrasound scanners in a broad patient population.

8:20

2aBAa2. Ultrasparse ultrasonic synthetic aperture beamforming via passive sensing. Francesco Lanza di Scalea (Structural Eng., Univ. of California San Diego, 9500 Gilman Dr., MC 0085, La Jolla, CA 92093-0085, flanza@ucsd.edu), Chengyang Huang, and Ali Hosseinzadeh (Structural Eng., Univ. of California San Diego, La Jolla, CA)

This presentation will describe the implementation of a passive ultrasonic sensing technique to achieve ultrasparse-transmission Synthetic Aperture Focusing (SAF) with Full Matrix Capture (FMC) capabilities. SAF ultrasonic arrays with sparse transmissions have been employed in both medical and industrial NDT imaging to increase imaging speed and simplify multiplexer hardware by reducing the number of high-voltage output channels, at the expense of a reduced beamforming matrix. A technique based on passive reconstruction of the ultrasonic Impulse Response Function (IRF) between two receivers will be demonstrated to create a "virtual" FMC capture while keeping a minimum number of physical transmitters, hence an ultrasparse-transmit array with full synthetic focusing capabilities. Several key steps of this passive beamforming approach will be discussed from both a theoretical and an experimental standpoint. The technique will be demonstrated for the imaging of drilled holes in an aluminum block using a linear, 64-element transducer array and only 1–4 physical transmitters. While the results presented have direct application to NDT imaging of solids, many of the aspects can be potentially applied to ultrafast imaging in the medical field, as well as to seismic interferometry for the health monitoring of civil structures.

8:40

2aBAa3. Towards computational super-resolution ultrasonic array imaging of material defects. Homin Song (Gachon Univ., Gachon, Korea (the Republic of)) and Yongchao Yang (Michigan Technolog. Univ., 1400 Townsend Dr., Houghton, MI 49931, ycyang@mtu.edu)

The resolution of sensing systems is fundamentally governed by the diffraction limit, which indicates that the minimum resolvable feature size is in the order of the wavelength of a propagating wave. Imaging smaller features (e.g., hidden material defects) requires short wavelengths (or high frequencies), which could be dangerous and causes low signal-to-noise ratio due to high attenuation of the propagating wave and coherent noise due to material backscattering. Computational super-resolution sensing, aiming to recover the subwavelength object features (e.g., material defects) from measurements taken with insufficient wavelength, is widely pursued across many applications; however, it is an ill-posed inverse problem that remains a significant challenge. Here we present a multi-scale deep learning approach to enable super-resolution ultrasonic beamforming that computationally exceeds the diffraction limit and visualizes sub-wavelength material defects. We investigate and discuss the applicability of this approach for computational super-resolution ultrasonic array imaging of hidden sub-wavelength defects of metallic structures.

9:00

2aBAa4. Model-based deep learning for ultrasound beamforming. Tristan S. Stevens (Elec. Eng., Eindhoven Univ. of Technol., 's Gravesandestraat 7s, Eindhoven, Noord-Brabant 5612JM, the Netherlands, t.s.w.stevens@tue.nl), Ben Luijten, and Ruud J. Sloun (Elec. Eng., Eindhoven Univ. of Technol., Eindhoven, the Netherlands)

Recently, the ultrasound signal processing pipeline has shifted from hardware-based solutions to the digital domain, enabling more intricate reconstruction techniques. We highlight several beamforming methods from a statistical inference perspective, connecting deep learning techniques to established signal processing models grounded in fundamental principles. The ultrasound acquisition process can be modeled as a linear combination of scatterers $\mathbf{x}$ (i.e. tissue reflectivity) which amounts to the received channel data $\mathbf{y}$, captured by a forward model $\mathbf{y} = \mathbf{Hx} + \mathbf{n}$, with the measurement matrix $\mathbf{H}$. Solving this system relies heavily on priors to yield a unique and anatomically feasible solution. A naive linear estimator for $\mathbf{x}$ is $\mathbf{H}^T\mathbf{y}$, also known as the delay-and-sum beamformer, which crudely disregards all off-axis scattering as zero-mean white noise. Based on the structured-noise assumption for off-axis scattering in the minimum variance beamformer, ABLE implements fast statistical inference of the optimal linear beamformer through deep learning. In addition, we can infuse prior knowledge on the distribution of $\mathbf{x}$ resulting in the neural maximum-a-posteriori beamformer. All aforementioned methods assume no dependency between individual pixels. Ultimately, the use of generative models allows us to extend the methods with more informative spatial priors.

9:20

2aBAa5. Improved beamforming by accounting for multiple domain shifts and sources of degradation. Ying-Chun Pan (Biomedical Eng., Vanderbilt Univ., Nashville, TN), Siegfried Schlunk, Christopher Khan (Vanderbilt Univ., Nashville, TN), and Brett Byram (Vanderbilt Univ., 2301 Vanderbilt Pl, Nashville, TN 37235, brett.c.byram@vanderbilt.edu)

A number of mechanisms are known to degrade ultrasound images. To address these issues, many beamforming strategies have been proposed including a slew of recent approaches centered around deep learning techniques. Most deep learning methods target a single type of image degradation. Here, we will simultaneously address multiple sources of image degradation within the same beamformer. In particular, we will address phase aberration, reverberation and off-axis scatting within a single network. Within this context of training networks to address multiple sources of image degradation, we will highlight the problem of domain shift in ultrasound beamforming deep networks. Domain shift is the mismatch in the statistics of the training data and the data encountered in the intended use case that reduces the performance of the trained deep network. We will show that ultrasound beamformers struggle with domain shift on both the input side and the output side of the networks, and we will present strategies for addressing both domain shifts including the shift on the output side where there is no ground truth data to learn from.

9:40

2aBAa6. Transcranial photoacoustic computed tomography image reconstruction with consideration of uncertainties in skull properties. Hsuan-Kai Huang (Elec. and Comput. Eng., Univ. of Illinois, Urbana-Champaign, 208 North Wright St., Urbana, IL 61801, hkhuang3@illinois.edu), Joseph Kuo (Elec. and Comput. Eng., Univ. of Illinois, Urbana-Champaign, Urbana, IL), Umberto Villa (Oden Inst. for Computational Eng. and Sci., The Univ. of Texas at Austin, Austin, TX), Lihong V. Wang (Elec. Eng., California Inst. of Technol., Pasadena, CA), and Mark Anastasio (Bioengineering, Univ. of Illinois Urbana-Champaign, Urbana, IL)

Transcranial photoacoustic computed tomography (PACT) is an emerging human neuroimaging modality but encounters challenges in achieving accurate image reconstruction due to aberrations induced by the skull. To compensate for aberrations, model-based image reconstruction methods that are based on the elastic wave equation have been proposed. However, to provide effective compensation, such methods often require the elastic and acoustic properties of the skull to be precisely known, which is challenging in practice. Moreover, these methods impose significant computational demands. In response to these challenges, a novel image reconstruction framework that was based on a learning method was proposed. The method involves the use of a U-Net based learning model to map an efficient but approximate reconstructed image to a de-aberrated, high-quality estimation of the initial pressure distribution within the cortical region of the brain. Computer-simulation studies that involved a realistic 3D stochastic head phantom were performed. The results of our studies demonstrate that the learning-based method exhibited similar performance to a state-of-the-art model-based technique when the assumed skull properties were precise, and notably surpassed the performance of the model-based method when uncertainties in the skull parameters were accounted for.

10:00–10:20 Break

10:20

2aBAa7. Differentiable beamforming for adaptive ultrasound imaging. Dongwoon Hyun (Radiology, Stanford Univ., 3155 Porter Dr., Palo Alto, CA 94304, dongwoon.hyun@stanford.edu)

Differentiable beamforming (DB) is a powerful new approach to image reconstruction that optimizes the beamforming algorithm to the given imaging target via autodifferentiation. In DB, the beamformer is expressed as a function of some unknown parameters, e.g., the sound speed throughout the medium, the shape of a flexible transducer, or the position and orientation of a swept transducer. Gradient descent is then used to find the parameters that optimize the focusing quality of the beamformer. With the appropriate choice of focusing criterion, the optimal beamforming parameters coincide with the ground truth. We demonstrate DB in applications of ultrasound autofocusing and sound speed imaging, flexible array shape estimation for wearable applications, and sensorless freehand swept synthetic aperture imaging.

10:40

2aBAa8. Abstract withdrawn.

11:00

2aBAa9. Parallel- and micro-beamforming challenges in real-time, high-frame-rate, ultrasound imaging. Piero Tortoli (Information Eng., Università di Firenze, via Santa Marta 3, Firenze 50136, Italy, piero.tortoli@unifi.it), Lorenzo Castrignano, Claudio Giangrossi, Valentino Meacci, Enrico Boni, and Alessandro Ramalli (Information Eng., Università di Firenze, Florence, Italy)

Delay-and-sum beamforming for classic (line-by-line) ultrasound imaging is usually performed in real-time by FPGAs and GPUs. However, when a high-frame-rate (HFR) has to be achieved, beamforming becomes challenging. Here, the echoes following the transmission of multi-line focused beams, plane waves, or diverging waves, must be simultaneously beamformed along multiple view lines. Such parallel beamforming is feasible online when the scanner is endowed with high flexibility and processing power. This talk will show how the FPGAs of the ULA-OP 256, a hardware-based open scanner, were efficiently utilized to enable parallel beamforming at high speed. The talk will also discuss how the data transfer between the scanner boards impacts the frame rate, which actually achieved 4400 frames per second through a new communication topology. The talk will also examine the image quality deterioration emerging when HFR imaging sequences are implemented in probes equipped with a microbeamformer (uB) that was designed for focused beam transmission. Simulations show how the transmitted beamwidth and uB size impact the image contrast and resolution, taking into account that the uB ASIC cannot handle multiple sets of delays and apodization weights after each transmit event. Technological improvements needed in the next generation uBs will be finally discussed.

11:20

2aBAa10. Real-time 3D ultrasound imaging with a clip-on device attached to common 1D array transducers. Zhijie Dong (Elec. and Comput. Eng., Univ. of Illinois Urbana-Champaign, Urbana, IL), Shuangliang Li (Elec. and Comput. Eng., Texas A&M Univ., College Station, TX), Chengwu Huang (Radiology, Mayo Clinic, Rochester, MN), Matthew R. Lowerison, Dongliang Yan, Yike Wang (Elec. and Comput. Eng., Univ. of Illinois Urbana-Champaign, Urbana, IL), Shigao Chen (Radiology, Mayo Clinic, Rochester, MN), Jun Zou (Elec. and Comput. Eng., Texas A&M Univ., College Station, TX), and Pengfei Song (Elec. and Comput. Eng., Univ. of Illinois Urbana-Champaign, 405 N. Mathews Ave., Beckman Inst. 4041, Urbana, IL 61801, songp@illinois.edu)

Performing 3D ultrasound imaging at a real-time volume rate (e.g., >20 Hz) is a challenging task. While 2D array transducers remain the most practical approach for real-time 3D imaging, the large number of transducer elements (e.g., several thousand) that are necessary to cover an effective 3D field-of-view impose a fundamental constraint on imaging speed. Although solutions such as multiplexing and specialized transducers, including sparse arrays and row-column-addressing arrays, have been developed to address this limitation, they inevitably compromise imaging quality (e.g., SNR, resolution) in favor of speed. Coupled with the high equipment cost of 2D arrays, these compromises hinder the widespread adoption of 3D ultrasound imaging technologies in clinical settings. In this presentation, we introduce an innovative transducer clip-on device comprising a water-immersible, fast-tilting electromechanical acoustic

reflector and a redirecting reflector to enable real-time 3D ultrasound imaging using common 1D array transducers. We will first introduce the principles underlying our novel technique, followed by validation studies incorporating simulation and experimental data. We will also demonstrate the feasibility of using the clip-on device to achieve a high 3D imaging volume rate that is suitable for advanced imaging modes such as shear wave elastography, blood flow imaging, and super-resolution ultrasound localization microscopy.

11:40

2aBAa11. Three-dimensional coherence beamforming using row-column arrays and matrix arrays. Joseph Thomas T. Hansen-Shearer (Bioengineering, Imperial College London, Royal School of Mines, Imperial College London, London SW7 2AZ, United Kingdom, jh12718@ic.ac.uk), Matthieu Toulemonde, Kai Riemer, Biao Huang (Bioengineering, Imperial College London, London, United Kingdom), Johanna Tonko (Univ. College London, London, United Kingdom), Jipeng Yan, Marcelo Lerendegui, Tan Qingyuan, Christopher Dunsby, Meng-Xing Tang, and Peter Weinberg (Bioengineering, Imperial College London, London, United Kingdom)

Much advancement in 3D ultrasound imaging has been achieved in recent years, allowing for its implementation in many research capacities. Translation of 3D imaging to a clinical setting has been hindered by small fields of view, expensive equipment and poor image quality. For ultrafast imaging, these challenges are accentuated by the need for a small number of transmission events. The row-column array is an emerging technology which can produce large field-of-view 3D ultrasound images, however, they can produce major artefacts when operated at high frame rates. Here we present some techniques which have been developed to improve the image quality and reduce the artifact level of row-column arrays by exploiting the incoherence of the transmission schemes. Presented will be the Row-Column specific Frame Multiply and Sum beamforming along with a technique for implementing Acoustic Sub-Aperture Processing which improves signal-to-noise ratio whilst also increasing the effective frame rate. Additional coherence-based algorithms, which have previously been developed for 2D imaging, are implemented and presented here for 3D imaging using a matrix array probe. All of the above algorithms have been optimised for super-resolution but should improve all types of 3D ultrasound imaging.

Session 2aBAb

Biomedical Acoustics, Education in Acoustics and Physical Acoustics: Return of the Writer

Julianna C. Simon, Cochair
Graduate Program in Acoustics, Pennsylvania State University, Penn State, 201E Applied Sciences Building, University Park, PA 16802

Karla Patricia Mercado-Shekhar, Cochair
Biological Engineering, Indian Institute of Technology Gandhinagar, Academic Block 6/207, Gandhinagar 382355, India

Kevin J. Haworth, Cochair
Department of Internal Medicine, University of Cincinnati, 231 Albert Sabin Way, CVC 3939, Cincinnati, OH 45267

Chair's Introduction—8:00

Invited Papers

8:05

2aBAb1. On the elements of an excellent research paper. Alfred Yu (Univ. of Waterloo, 200 University Ave. West, Waterloo, ON N2L3G1, Canada, alfred.yu@uwaterloo.ca)

Journal publications play a key role in the dissemination of research findings in the biomedical acoustics community. However, authors are often unsure of how to write a paper that will be considered as excellent by their peers, because the notion of excellence is arguably subjective and ill-defined. To demystify such an issue, this presentation will discuss four major elements (novelty, significance, rigor, writing quality) that are often found in excellent journal papers in biomedical ultrasound. Since 2022, these four elements have been adopted as the peer review criteria by the flagship ultrasonics journal of the IEEE press. Multi-day writing workshops with interactive, hands-on tutorials have been designed to engage prospective authors on how to plan and prepare journal manuscripts with the four elements of excellence. Reviewer training workshops have also been offered to inform journal reviewers about these evaluation criteria and expectations. Through these initiatives, we aim to establish a quality-driven publishing emphasis within the biomedical acoustics community. Achieving so will have long-term benefits on the reputation and growth of this research field.

8:25

2aBAb2. Systematic literature review in five steps. Costas Arvanitis (School of Mech. Eng., Dept. of Biomedical Eng., Georgia Inst. of Technol., 311 Ferst Dr. Northwest, Atlanta, GA 30332, costas.arvanitis@gatech.edu)

A systematic review requires a considerable amount of time and resources. But if conducted properly it gives the opportunity to (i) summarize the current state of the art, (ii) identify the next questions that need to be answered to move one's field forward, (iii) Inform one's own research, (iv) support effective writing, and (v) receive recognition from your peers. This presentation will discuss the following five steps for conducting effective systematic literature review: (i) frame the question for the review; (ii) identify relevant work; (iii) assess the quality of studies; (iv) summarize the evidence; and (v) interpret the findings. The PRISMA (Preferred Reporting Items for Systematic Reviews and Meta-Analyses) flow diagram and checklist will be presented. A case study related to the estimation of cavitation thresholds during microbubble-enhanced focused ultrasound targeted drug delivery in the brain from a recent review paper by the author (Schoen *et al.*, Advanced Drug Delivery Reviews, 2022) will be discussed.

8:45

2aBAb3. Teaching the next generation of engineers and scientists to write: Taking responsibility and overcoming challenges. Michael Alley (College of Eng., Penn State, 201 Hammond Bldg., Penn State Univ., University Park, PA 16802, mpa13@psu.edu)

In educating engineering and science students as authors, four challenges exist today that did not exist thirty years ago. First, the decreasing attention over the past three decades on teaching grammar to students has left us with many students unable to analyze writing at the sentence level. Second, the increased focus by writing teachers on the writing process to the exclusion of the product has given many students the impression that properly finishing a paper is not important. No doubt, developing a process to write is important, but so is the final product. Third, many engineering and science instructors mistakenly assume that the responsibility of teaching students to write resides solely with English departments. To develop as authors in their disciplines, engineering and science students also need thoughtful feedback from experts in the field, particularly with regard to the precision of technical terms and the proper emphasis of details. Fourth, the emergence of artificial intelligence has led many students to assume that they need not learn anymore how to write. This paper presents three undertaught writing skills that engineering and science instructors should emphasize to the next generation of engineers and scientists for them to achieve excellence in their documents.

9:05

2aBAb4. But I did science so I wouldn't have to write essays. Eleanor P. Stride (Univ. of Oxford, Inst. of Biomedical Eng., Oxford OX3 7DQ, United Kingdom, eleanor.stride@eng.ox.ac.uk)

A dislike of writing, sometimes bordering on a phobia, seems to be a common experience for many science students and indeed practising scientists. This represents a serious challenge, as regular publication is a key requirement for most areas of research. In seeking to understand the origins of this aversion, a few interesting findings have arisen. First, there are some crucial differences between technical and creative writing, but these are rarely highlighted as part of either high school or university science courses. Second, there are surprisingly few degree courses in which the scientific method is introduced and discussed explicitly. In this talk, strategies for addressing these deficiencies and taking the pain out of report and paper writing will be discussed.

9:25

2aBAb5. Writing for a "different" audience. Kat Setzer (Acoust. Society of America, Acoust. Society of America, Salem, MA 11747, ksetzer@acousticalsociety.org), Micheal Dent (Psych., Univ. at Buffalo, SUNY, Buffalo, NY), and Arthur N. Popper (Biology, Univ. of Maryland, College Park, MD)

Written communication is a critical part of scholarship. However, scholarly writing is often incomprehensible to a broad audience, though we know that broader communication has great value in helping non-experts understand and support science. To bring acoustics to a larger audience, ASA publishes Acoustics Today (AT) to communicate the breadth of the field to members and to the public. To be successful, AT articles must be written so that they are scholarly, but also readable, understandable, and engaging to a college freshman, a newspaper reporter, a regulator, a funder, or one's bosses. Thus, a challenge for those of us working on AT is to help authors understand how to mold their ideas and material to our audience. Most authors find writing for AT a challenge since it is so different from their normal writing. As a result, we may go through many iterations of an article to help authors reach our audience. However, the reward is not only a well-written article that authors can share with colleagues in other disciplines, their bosses, and their parents, but many report that they have a new understanding of communicating with a wider audience.

9:45–10:00 Break

10:00

2aBAb6. Best practices for publishing in the Journal of the Acoustical Society of America. James Lynch (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., 86 Water St., Falmouth, MA 02543, jlynch@whoi.edu) and Elizabeth A. Bury (Acoust. Society of America, Melville, NY)

Scientists, engineers, and practitioners all have good reasons to publish in technical journals such as *The Journal of the Acoustical Society of America* (*JASA*), and so need to understand the procedures and best practices involved in doing so. While *JASA* has a technical focus in acoustics, it is also similar to other journals in many aspects of its writing and procedural requirements. We will address both the "business communication" and technical content aspects of journal writing. We will also stress some of the standard rules and protocols of journal writing, such as (1) polite and professional communications; (2) following specific journal requirements; (3) quoting and citing appropriately; (4) responding to comments from editors and reviewers; (5) complying with ethical standards and journal policies, etc. Useful references will be supplied to help with particular topics.

10:20

2aBAb7. Learnings from developing a Scientific Writing Certification program at IIT Gandhinagar. Karla Patricia Mercado-Shekhar (Dept. of Biological Sci. and Eng., Indian Inst. of Technol. Gandhinagar, Academic Block 6/207, Indian Inst. of Technol. Gandhinagar, Gujarat 382355, India, karlamshekhar@iitgn.ac.in)

Effective written communication is essential to an impactful career. The Certification in Scientific Writing initiative at the Indian Institute of Technology (IIT) Gandhinagar was developed to encourage researchers to work on scientific writing from an early stage in their careers and to promote skill development. This program serves as a milestone for gauging students' skills and knowledge in scientific writing. Since the certification program was established in July 2020, over 200 Ph.D. students and postdoctoral fellows from all departments at IIT Gandhinagar passed the program with approximately a 60% success rate. This program spans 2 months throughout the semester and offers training, evaluation, and feedback to students on their scientific writing competencies. In this talk, I will share the development of the program from inception to implementation and the feedback received from the IIT Gandhinagar community.

10:40

2aBAb8. How to write a peer-polished proposal in 15 weeks. Christy K. Holland (Internal Medicine, Div. of Cardiovascular Health and Disease, and Biomedical Eng., Univ. of Cincinnati, Cardiovascular Ctr. Rm. 3935, 231 Albert Sabin Way, Cincinnati, OH 45267-0586, Christy.Holland@uc.edu)

Creating a meticulously crafted proposal requires a strategic approach and systematic planning. An overview of a semester-long graduate course on how to write successful NIH grant applications will be provided. Particular emphasis is given to developing proposals for the Ruth L. Kirschstein Predoctoral Individual National Research Service Award (https://researchtraining.nih.gov/programs/fellowships/F31) or to disease-based foundations. The writing process involves drafting components in several key phases. The initial four weeks focus on understanding the proposal requirements, identifying the target audience, organizing a brilliant biosketch and remarkable resources and environment pages, and establishing a clear hypothesis and specific aims. An extensive literature review is conducted in the subsequent two weeks to contextualize the proposal, identify a gap in knowledge, and stress the significance and innovation of the proposed work. Weeks 7 and 8 are devoted to the development of a robust research approach and methodology, including data collection, analysis techniques, expected outcomes, potential challenges, and alternative approaches. The review process, refinement and enhancement take center stage for the remaining weeks. Peer review and feedback mechanisms are incorporated to iteratively improve each proposal's coherence, logic, and persuasiveness. This systematic 15-week timeline emphasizes iterative refinement through peer input, ensuring a polished proposal ready for submission.

Contributed Papers

11:00

2aBAb9. Graduate-level research communication course: Reflections on the first offering. Julianna C. Simon (Penn State Univ., Penn State201E Appl. Sci. Bldg., University Park, PA 16802, jcsimon@psu.edu)

In Fall 2022, I offered a new course on research communication within the Graduate Program in Acoustics at Penn State. With 8 students from Acoustics and Biomedical Engineering, the course was split into three modules of grant writing, paper writing, and outreach. I found the students were reasonably adept at critically reading single papers. They also generally understood the mechanics of writing sentences, although combining sentences into paragraphs, connecting ideas, reducing fluff, and adding emphasis to key points were areas that significantly improved throughout the semester. In the middle of the course, two mock grant review panels were held, which highlighted a need to help students develop strategies to cope with critical feedback. Thus, for the final project, students had the option to either write a journal paper draft or revise their proposal; they also presented orally on their final project. Additionally, communicating with the public and developing outreach activities were included in the course, which culminated in an Acoustics Day at a science center. Overall, I found identifying gaps in the literature using multiple sources and developing a cogent argument on how their proposed work would fill the gap needed the most improvement. [Partially supported by the NSF CAREER:1943937].

11:15

2aBAb10. Empowering research: The role of writing accountability groups. Nour Al Rifai (Internal Medicine, Univ. of Cincinnati, Cincinnati, OH), Eric Smith (Dept. of Internal Medicine, University of Cincinnati College of Medicine, Cincinnati, OH 45221-0663, Smithep4554@gmail.com), Danielle N. Tap, Yolanda Y. Wess, and Kevin J. Haworth (Internal Medicine, Univ. of Cincinnati, Cincinnati, OH)

Productive writing is a challenge for researchers balancing multiple roles. Writing Accountability Groups (WAGs) can increase productivity by providing dedicated writing time. Patterned after a validated structure established by Kimberly Skarupski, Ph.D. (Johns Hopkins School of Medicine), the University of Cincinnati Department of Internal Medicine initiated four 10-week-long WAGs. Weekly virtual one-hour writing sessions were organized by staff with protected time and included an emphasis on principles of time management. Two WAGs were for junior faculty and two for residents/postdoctoral fellows, with 3–6 participants each. Attendance was approximately 80%, and 18 of 20 participants continued for the entire 10-week period. 14 of 20 participants completed a post-WAG survey. 85% reported improved time management skills. Additionally, most participants felt the WAG provided communal support and accountability. Rigorous writing project completion tracking is a future goal. Motivated by the experience, one postdoctoral fellow participant established a laboratory-based WAG. The participants included one faculty member, two postdocs, one Ph.D. student, two research staff, and ten undergraduate students. WAG participation was 50 ± 17 % over 44 weeks. In summary, similar to the experience of Skarupski, the UC WAGs facilitated consistent focus on writing, fostered a sense of community, and provided accountability.

11:30–12:00
Panel Discussion

Session 2aCA

Computational Acoustics: Computational Acoustics Methods Evaluation

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

Contributed Papers

8:00

2aCA1. A learned born series for highly scattering media. Antonio Stanziola (Medical Phys. and Biomedical Eng., Univ. College London, Dept. of Medical Phys. and Biomedical EngineeringUniversity College London, London WC1E 6BT, United Kingdom, a.stanziola@ucl.ac.uk), Simon Arridge (Univ. College London, London, United Kingdom), Ben Cox (Dept. of Medical Phys. and Biomedical Eng., Univ. College London, London, United Kingdom), and Bradley Treeby (Univ. College London, London, United Kingdom)

The Helmholtz equation with heterogeneous material properties is essential in various fields requiring single-frequency wave simulations, including optics, seismology, acoustics, and electromagnetics. Traditional methods for solving this equation, such as the Born Series, face limitations in converging for high-contrast scattering potentials. To address this challenge, we introduce a novel method called the Learned Born Series (LBS). The LBS is derived from the convergent Born Series but employs components that are learned through training. It demonstrates significantly improved accuracy compared to the conventional convergent Born Series, especially in scenarios with high-contrast scatterers, while maintaining similar computational complexity. The LBS can rapidly generate reasonably accurate predictions of the global pressure field with only a few iterations, and the errors decrease as more iterations are learned. We show its effectiveness through experiments on simulated datasets. The LBS offers promising prospects for accelerating simulations in scenarios with strong sound speed contrasts, potentially revolutionizing applications in transcranial treatment planning and full waveform inversion.

8:15

2aCA2. Application of differentiable programming to wave simulation. Antonio Stanziola (Medical Phys. and Biomedical Eng., Univ. College London, Dept. of Medical Phys. and Biomedical EngineeringUniversity College London, London WC1E 6BT, United Kingdom, a.stanziola@ucl.ac.uk), Simon Arridge (Univ. College London, London, United Kingdom), Ben Cox (Dept. of Medical Phys. and Biomedical Eng., Univ. College London, London, United Kingdom), and Bradley Treeby (Univ. College London, London, United Kingdom)

Wave simulations play a crucial role in a wide range of scientific and engineering applications, including seismic imaging, optical design, and acoustic modeling. Here we explore the advantages of differentiable programming in the context of acoustic wave simulation. Differentiable programming enables us to treat wave simulation as a differentiable function, allowing for the automatic computation of gradients with respect to any continuous input parameter. We demonstrate how this approach can be applied to various types of wave simulations, such as Pseudo-spectral time-domain solvers or iterative solvers. Implementing wave simulators via differentiable programming achieves several benefits. First, it enables efficient and accurate sensitivity analysis: This is particularly valuable for optimization and uncertainty quantification tasks. Second, it facilitates the incorporation of wave simulations into machine learning frameworks, enabling the integration of simulation-based models with data-driven approaches. Third, differentiable programming can accelerate the calibration and inversion of wave simulation models, making it easier to match simulated results to observed data. We present practical examples and discuss potential applications in fields such as geophysics and medical imaging. Our findings highlight the potential of this approach to advance the state-of-the-art in wave simulation techniques and their integration into larger computational pipelines.

8:30

2aCA3. Propagation of sonic booms around urban landscapes utilizing ray tracing. Brett A. Schankin (Phys., Farmingdale State College, 2350 NY-110, Farmingdale, NY 11735, schaba5@farmingdale.edu) and Kimberly A. Riegel (Phys., Farmingdale State College, Farmingdale, NY)

With the increasing interest from airlines and technology startups to return usage of commercial supersonic flight, there are still issues that need to be addressed. One of the more prominent barriers remains to be the high level of noise pollution produced by the shockwaves of sonic booms. While NASA and Lockheed Martin are currently developing the X-59 QueSST, an experimental aircraft with the aim of reducing the noise caused by these shockwaves, a closer look at the phenomena surrounding sonic booms can be examined by building computer models. The research being done looks to code a model that simulates the behavior of sound waves produced by sonic booms when they interact with urban geometries and structures. This is accomplished by utilizing a ray tracing method in a developed Python script. This script was initially developed using Fortran, but was ported to Python to take advantage of the language's ease of use and extensive libraries. The Python code must be vigorously validated to ensure it is correctly calculating the physical properties and direction of the sonic booms. Two types of atmospheres are included in the numerical simulations, a uniform atmosphere and the standard atmosphere. Comparisons of both types of atmospheres are shown compared to other numerical prediction models and measurement from the NASA SonicBOBS field measurements from 2009.

8:45–9:00 Break

9:00

2aCA4. Stochastic ray tracing implementation for sonic boom propagation modeling. Chloe A. Marini (Phys. and Astronomy, Univ. of Alabama in Huntsville, 6854 Governors West NW, Apt 2211, Huntsville, AL 35806, chloe.marini@yahoo.com) and Kimberly A. Riegel (Phys., Farmingdale State College, Farmingdale, NY)

Sonic boom propagation in large urban areas needs to be understood to determine the impact it will have on residents. In previous work, a combination of ray tracing and radiosity method was used to model the reflections of the sonic booms around large structures. Radiosity is a memory intensive method which requires extensive computational resources as the environment becomes more complex, whereas stochastic ray tracing does not require substantially more resources for more complicated environments. This study examines the feasibility of using stochastic ray tracing to simulate the diffuse reflections of sonic booms. Receiver graphs comparing stochastic ray tracing and radiosity methods for several environments will be shown, along with their computation times.

9:15

2aCA5. Fourie series decomposition of a modified propagation operator in wide-angle parabolic diffraction models: Accuracy analysis. Philippe Blanc-Benon (CNRS, 36 Ave. Guy de Collongue, Ecully 69134, France, philippe.blanc-benon@ec-lyon.fr), Elena O. Konnova, Maria M. Karzova, Vera Khokhlova, and Petr V. Yuldashev (Lomonosov Moscow State Univ., Moscow, Russian Federation, Moscow, Russian Federation)

Wide-angle parabolic equations are often used to model acoustic wave propagation in inhomogeneous media assuming weak backscattering effects. However, for three dimensional problems, efficient implementation of numerical methods to solve such equations using Padé approximations of the propagation operator is still a challenge. Fourier series approximation of the propagation operator can be used as an alternative approach since operator splitting methods are applicable to each Fourier harmonic of the series. Then calculations along two coordinates perpendicular to the primary wave propagation direction can be separated. High accuracy of the Fourier approximations can be achieved by constructing modified version of the propagator, which is smooth and continuous in a periodic window of the angular spectrum space under constraints of the maximum diffraction angle. Segments of the modified propagator complex valued function are connected using high-order Hermite interpolation method, which is capable to provide continuity up to the sixth order derivatives. Several free parameters available in the modified propagator can be used to diminish the approximation error appearing due to the Gibbs oscillations. The accuracy of the proposed Fourier approximations is tested for different maximum propagation angles and numbers of Fourier harmonics. [The work was supported by RSF 23-22-00220.]

9:30

2aCA6. Enabling support for Octave within the FOCUS software package. Jacob S. Honer (Elec. and Comput. Eng., Michigan State Univ., East Lansing, MI) and Robert J. McGough (Elec. and Comput. Eng., Michigan State Univ., 428 S. Shaw Ln., Rm. 2120, Michigan State University, East Lansing, MI 48824, mcgough@egr.msu.edu)

FOCUS, the "Fast Object-Oriented C++Simulator" (https://www.egr.msu.edu/~fultras-web/), is a free software package that enables rapid calculations of continuous-wave and transient pressure fields generated by single transducers and phased arrays. FOCUS achieves small errors in relatively short computation times through linear memory-efficient calculations with the fast nearfield method. This capability extends to quick and accurate biomedical imaging simulations within FOCUS. Moreover, FOCUS supports some nonlinear pressure calculations, such as the continuous-wave and transient Khokhlov-Zabolotskaya-Kuznetsov (KZK) equations for circular and spherical transducers. Additionally, the Angular Spectrum Approach (ASA), a frequency-domain solution to linear computational methods, ideal for large volumetric field computations, is also included within the FOCUS package. Initial success with MATLAB motivates the creation of an Octave version that replicates the core functionalities of the original FOCUS package. GNU Octave is similar to MATLAB, but unlike MATLAB, Octave is a free software. Building and compiling FOCUS in Octave required a few minor alterations due to discrepancies in syntax and function calls. The use of Octave-compiled MEX files simplified the conversion process while introducing addition memory overhead compared to rewriting C++ code for Oct-file structuring. The performance and capabilities of FOCUS in Octave are demonstrated and discussed.

Session 2aEA

Engineering Acoustics, Physical Acoustics and Structural Acoustics and Vibration: Multidomain Modeling of Acoustical Systems

Stephen C. Thompson, Chair
Graduate Program in Acoustics, Penn State University, 201 Applied Science Building, University Park, PA 16802

Chair's Introduction—10:15

Invited Paper

10:20

2aEA1. Coupled models for designing and predicting performance of active sonar systems. Thomas E. Blanford (Univ. of New Hampshire, University of New Hampshire, Durham, NH 03824, thomas.blanford@unh.edu), Shawn Johnson, Jason Philtron, and Daniel C. Brown (The Penn State Univ., State College, PA)

Active sonar systems used to survey and image the seafloor typically transmit a pulse on a projector, receive echoes on an array of hydrophones, and condition and digitize the signals using on-board electronics. The quality of the data produced by these sensors is a function of the transducers, the electronics, and the environment. Simultaneous modeling of all these factors is necessary for accurate prediction of the system performance. Data products from these systems are often uncalibrated because absolute calibration is not required for their applications. However, modeling active sonar systems in their natural units within each domain provides clear benefits such as allowing system limitations to be accurately identified early during the design phase and easing validation of system performance in comparison with experimental data. This presentation will discuss the use of coupled multi-domain models in the design of two active sonar systems: the Sediment Volume Search Sonar and a multibeam echosounder architecture based on sigma-delta conversion. By coupling electro-mechanical transducer models and electrical hardware models with the Point-based Sonar Signal Model (PoSSM), it was possible to identify critical design parameters and sensitivities in the design of these sensors.

Contributed Papers

10:40

2aEA2. Multidomain modeling of acoustical transducers and arrays using electrical network theory. David A. Brown (ECE/CIE, Univ. of Massachusetts Dartmouth, 151 Martine St., UMass Dartmouth, Fall River, MA 02723, dbAcousticsdb@gmail.com)

Modeling of piezoceramic acoustic transducers involves the transformation of energy in the electrical, mechanical, acoustical radiation domains. By using generalized coordinates and Lagrangian mechanics, in which principle of least action is applied in each domain, the electrical, piezoelectric, mechanical and sound radiation problems can be solved separately and then combined in a multi-contour equivalent electro-mechanical circuit with each mechanical vibrational resonant modes representing separate degrees of freedom in the coupled electrical circuit. This energy approach involves the calculation of the potential and kinetic energies of each practical mode of vibration, which can be determined analytically, experimentally, or by finite-element-analysis. This is an alternative to the use of Newtonian mechanics which requires an accurate description of the real boundary conditions in the device and is common in many FEA modeling approaches. The availability of software (e.g., Matlab, Python, LTSpice, etc.) to model electrical circuits (networks in the case of multi-resonant devices) makes for a powerful and efficient modeling approach. The historical emergence of this approach stems from adapting the Rayleigh-Ritz method for mechanical and electrical domains with the advent of piezoelectric and magnetostrictive bodies. Several example of piezoceramic transducers are presented.

10:55

2aEA3. Abstract withdrawn.

Invited Paper

11:10

2aEA4. Modeling nonlinearities and thermal effects in moving coil speakers. Stephen C. Thompson (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, sct12@psu.edu)

Simple explanations of moving coil speaker behavior assume that the magnetic force factor, the mechanical spring stiffness and the coil electrical inductance are all constants of the design. The work of Klippel identifies all of these as varying with speaker cone displacement. The electrical resistance also varies with temperature, which varies because of electrical heating in the coil. Methods of measuring these nonlinearities are available in commercial equipment. Using this data to model speaker performance requires a model that considers behavior I the electrical, mechanical, thermal and acoustical domains. This paper describes an open source method to use the measured nonlinear parameters to calculate the time waveform of radiated pressure with large signal excitation for speakers.

Contributed Paper

11:30

2aEA5. Hydrostatic pressure effects in periodically structured polyurethane acoustic tiles. Luke R. Hacquebard (Defence Res. and Development Canada (DRDC), 2635 Provo Wallis St., Halifax, NS B3K 5X5, Canada, luke.hacquebard@ecn.forces.gc.ca) and Jeffrey P. Szabo (Defence Res. and Development Canada (DRDC), Halifax, NS, Canada)

Traditional underwater acoustic materials commonly make use of periodic air voids to improve their ability to absorb or deflect sound through wave-mode conversion and scattering mechanisms. These air voids have the potential to deform or completely collapse under hydrostatic pressure loading, which may significantly affect the material’s acoustic performance. This study investigates the effects of hydrostatic pressure on the acoustic response of two different types of fabricated polyurethane samples. Developed numerical finite element simulations help identify the main factors associated with the acoustic pressure dependence of such tiles. The results of this work may aid in the development of pressure resistant materials that can better maintain acoustic performance at depth.

Session 2aMU

Musical Acoustics: Winds Instruments I

Gary Scavone, Cochair
Music Research, McGill University, 555 Sherbrooke Street West, Montreal, H3A 1E3, Canada

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

Chair's Introduction—8:40

Invited Papers

8:45

2aMU1. Acoustical behaviour of "baroque" trumpets with vent holes. Murray Campbell (School of Phys. and Astronomy, Univ. of Edinburgh, James Clerk Maxwell Bldg., Peter Guthrie Tait Rd., Edinburgh EH9 3FD, United Kingdom, d.m.campbell@ed.ac.uk), Arnold Myers (Royal Conservatoire of Scotland, Glasgow, United Kingdom), and Michael Newton (Reid School of Music, Univ. of Edinburgh, Edinburgh, United Kingdom)

The trumpets for which baroque composers including Bach and Handel wrote viruoso solo parts were natural (valveless) instruments approximately twice the length of a modern C trumpet. Playing chromatically in the high register of a natural trumpet requires great skill, since only the approximately harmonic resonances of the fixed-length air column are available. In the second half of the twentieth century several makers collaborated with performers to develop variants of the natural trumpet incorporating three or four finger-holes. Such instruments, widely used in modern ensembles playing baroque music, are best described as 'vented trumpets'. The way in which the finger-holes are used differs from the use of tone-holes on the cornett, the keyed trumpet and the keyed bugle. In general, vented trumpets are designed to approach the timbre of natural trumpets while facilitating control of intonation and accuracy in playing. In the present paper, the acoustical foundations for the function of the three- and four hole systems are explored. The study draws on input impedance measurements, computational modelling, and playing tests of typical vented and natural trumpets. Parameters quantifying the timbre of vented trumpets are compared with corresponding values for natural trumpets.

9:05

2aMU2. A comparison of modeled and measured impedance of brass instruments and mouthpieces. Miranda Jackson (Music Res., CIRMMT, McGill Univ., Schulich School of Music, Montreal, QC H3A 1E3, Canada, miranda.jackson@mail.mcgill.ca) and Gary Scavone (Music Res., CIRMMT, McGill Univ., Montreal, QC, Canada)

The impedance of a brass instrument has an important influence on its playability and sound timbre. The geometry of the mouthpiece has various features, such as the cup volume and shape, opening diameter, and length, that determine the characteristics of the overall impedance of the instrument-mouthpiece combination. Brass instruments, and especially mouthpieces, are designed for specific purposes, and horns or mouthpieces are chosen depending on the musical requirements. In order to investigate the relationship between the physical parameters and the impedances of instruments and mouthpieces, they have been modeled with transfer matrix and finite element model techniques, and the results are compared with impedance measurements of instruments, mouthpieces, and combinations of instruments and mouthpieces. Trumpets, flugelhorns, (French) horns, trombones, and the corresponding mouthpieces have been used. A detailed analysis of the estimation of the viscothermal losses has been performed, as the loss estimation in the narrow throat of the mouthpiece and in the flaring part of the brass instrument bell departs from the ordinary transfer matrix calculations. The effect of varying the physical parameters of mouthpieces and instruments is investigated by means of impedance considerations and sound synthesis, and the resulting influences on intonation, playability, and timbre are presented.

9:25

2aMU3. Pitch change of the stopped French horn. Robert Pyle (Stephens Brass Instruments, 11 Holworthy Pl., Cambridge, MA 02138, rpyle@icloud.com)

Composers sometimes ask for a form of muting on the French horn called "hand stopping," where the player closes the bell with the heel to the right hand as completely as possible. This alters the timbre of the instrument and its pitch as well. Players are taught to compensate for the pitch change by transposing down a semitone on the F horn. Stopping thus appears to have raised the pitch of the F horn by a semitone. However, if the heel of the hand is gradually closed from its normal "open" position to full stopping while the horn is played, the pitch falls smoothly. So does hand stopping raise the pitch, or lower it? This has been the topic of sometimes heated discussion among hornplayers for perhaps 250 years. An experiment shows that both viewpoints can be considered correct. A mouthpiece was fitted with an earphone and a microphone so that the acoustical round-trip time through the horn can be determined from the response to

a click emitted by the earphone. As the hand is closed, the round-trip time increases ("pitch falls") but as the hand approaches full stopping, an earlier reflection appears and eventually dominates ("pitch rises").

9:45

2aMU4. Power and good music: The Indigenous southern plains flute tradition. Paula Conlon (School of Music, Univ. of Oklahoma, 593 MacLaren St., Ottawa, ON K1R 5K8, Canada, pconlon@ou.edu)

In traditional Indigenous southern plains culture, a young man could not talk to a young woman alone when they were not yet married. Instead, he would play his flute at the edge of the encampment in the evenings, and each young man had his own love song. In southern plains flute origin stories, power is attributed to good music. If a flute song achieves its intended goal of convincing the young woman to marry the flute player, one can assume the song would be considered "good." But what criteria distinguish good from bad? Which elements typify a "good" flute song? What about the flute itself? Which features epitomize the quintessential flute? Another set of possible criteria in determining that quality is information about the flute player. In her chapter, "Culture and Aesthetics," ethnomusicologist Marica Herndon (1980) observes the community-centered perspective of Indigenous North America. Our last set of criteria involves an assessment of the moral character of the flute player regarding service to their tribal community. This talk discusses the "good music" of two master Indigenous southern plains flute players—Belo Cozad (1864-1950) (Kiowa) and Doc Tate Nevaquaya (1932-1996) (Comanche).

10:05–10:20 Break

10:20

2aMU5. Impedance phase-dependent correction factor for a tonehole model. Michael Prairie (Elec. and Comput. Eng., Norwich Univ., 158 Harmon Dr., Northfield, VT 05663, mprairie@norwich.edu)

The upper and lower end corrections of an open tonehole in a woodwind instrument are often accommodated by adding them to the physical thickness of the wall to create an effective thickness used in calculating the impedance of the tonehole. A rough estimate of the correction often used is 1.5 time the tonehole radius (1.5b), while other more rigorous treatments yield values in this vicinity. These values are usually independent of frequency or are constrained to a range of geometric tonehole and bore dimensions. The data from this study showed a strong and significant frequency dependency that produced a large range of correction values between about 0.5b and 1.5b. This variation correlated with the phase of the impedance of the highest open tonehole, and generally increased with increasing sounding frequency. This paper will discuss the conditions under which the measurements were made and how the correction factors were obtained, and will propose how the observed frequency dependence can be incorporated into the correction factor through the phases of the impedances of the open toneholes.

10:40

2aMU6. Multiphonics in the recorder: Some observations on the effects of selective basis tone removal induced by localized perturbations of window-region air flow. Katherine L. Saenger (Phys., Auburn Univ., Ossining, NY, klsaenger@yahoo.com) and Nicholas Giordano (Phys., Auburn Univ., Auburn, AL)

Multiphonic tones in the recorder are readily produced with certain fingerings and pressure/flow conditions. These tones typically comprise two strong basis tones that do not have a harmonic relationship, along with weaker tones whose frequencies are linear combinations of the basis tone frequencies. In this work, an artificial blower was used to generate multiphonic tones in three recorder body types (each attached to the same Yamaha soprano recorder head): (i) a forked fingering for G5 in a standard tapered body; (ii) a C5 in a cylindrical body; and (iii) a C5 in a capped cylindrical body. It was found that a thin post placed in the recorder's window region could selectively remove one of the basis tones (as well as the combination tones) with little effect on the sound power of the other basis tone. Navier-Stokes simulations were performed to help understand why a post in this position can have these effects. [Work supported by NSF under Grant No. PHY2306035.]

11:00

2aMU7. Measuring the behavior of the acoustic standing wave exiting a flue organ pipe: Is the decay sinusoidal or exponential? Lauren K. Schefter (Phys., Rollins College, 1000 holt Ave., Box #6539, Winter Park, FL 32789, lschefter@rollins.edu), Whitney L. Coyle, Ashley E. Cannaday (Phys., Rollins College, Winter Park, FL), and Eric Rokni (Phys., Rollins College, Mequon, WI)

Correctly predicting the playing frequencies of a musical instrument is dependent on the length of the resonator with the addition of an end correction. There are multiple theories describing this end correction, perhaps the simplest being that the end correction of a pipe is a physical extension of the sinusoidal pressure standing wave inside the pipe. However, recent optical imaging of the flow in a flue organ pipe found an unexpected exponential decay of pressure just outside of the pipe. This work looks to validate those findings acoustically. A flue organ pipe was played at the 1st, 5th, and 7th harmonics and the pressure just inside and immediately outside the end of the pipe played was measured using a zero-degree PU Match Microflown sound intensity probe. These measurements were fit to both exponential and sinusoidal curves and compared to the optical images. While an exponential trend is in fact apparent in some cases, the goodness-of-fit appears to be dependent on which harmonic is sounding. Future work includes exploration of a potential transitional region, assessing the impact of altered pipe geometry (both cross-sectional shape and size), and investigating potential sensor interference by using other measurement equipment.

Contributed Papers

11:20

2aMU8. Spectral behavior of recorder tones during transitions between notes. Nicholas Giordano (Phys., Auburn Univ., College of Sci. and Mathematics, Auburn, AL 36849, njg0003@auburn.edu) and Katherine L. Saenger (Phys., Auburn Univ., Ossining, NY)

We have studied note transitions in a soprano recorder as tone holes are opened or closed. Experimental results with an instrument excited by an artificial blowing machine are compared to Navier-Stokes-based simulations for an essentially identical instrument geometry. Several different cases have been studied. (1) Transitions involving the opening or closing of a single tone hole, with the notes separated by two semitones; e.g., transitions between C5 and D5. (2) Transitions involving the opening or closing of several tone holes simultaneously; e.g., transitions between C5 and G5. (3) Transitions involving a single tone hole but for notes separated by a major fifth; transitions between C5 and G5 using a forked fingering. Our work has focused on tone hole openings and closings at speeds typical of a human player but interesting behavior was also observed for longer tone hole switching times for cases (2) and (3), in which the sound field inside and outside the instrument was found to take much longer than the tone hole switching time to reach steady state. Work supported by NSF grant PHY2306035

11:35

2aMU9. Some observations on the sound powers of flute combination tones produced by singing while playing. Katherine L. Saenger (Independent Researcher, Ossining, NY, klsaenger@yahoo.com)

Singing while playing" is an extended flute technique that can produce weak, but audible, combination tones that are heard along with the note being sounded on the flute and the note that the player is singing or humming. The present work was motivated by an interest in obtaining a quantitative understanding of the factors determining the sound powers of these combination tones in hopes of making them louder and/or more audible. The tones under study were produced in a conventional flute powered by an artificial blower apparatus whose pressure/flow could be modulated at audio frequencies (by a speaker connected to the gasline) to mimic player singing. While quantitative results were elusive, one takeaway finding was clear: The combination tones are loudest when they have a frequency that aligns with one of the flute's passive resonances, a situation which can be facilitated with the use of forked fingerings that preserve the strength and position of the resonance used for the sounded flute note while shifting the positions of the unused resonances so that one of them better coincides with the frequency of the desired combination tone.

TUESDAY MORNING, 14 MAY 2024 — ROOM 205, 8:00 A.M. TO 11:15 A.M.

Session 2aNSa

Noise, Computational Acoustics and Psychological and Physiological Acoustics: Advanced Air Mobility Noise: Noise from New Air Transportation in Urban and Underserved Communities I

Matthew Boucher, Chair
NASA Langley Research Center, 2 N. Dryden St., M/S 463, Hampton, VA 23681

Chair's Introduction—8:00

Invited Paper

8:05

2aNSa1. Optimization of propellers under consideration of aeroacoustic and aerodynamic goals and installation effects. Michael Schmähl (Chair of Aircraft Design, Tech. Univ. of Munich, Boltzmannstraße 15, Garching 85748, Germany, michael.schmaehl@tum.de) and Mirko Hornung (Chair of Aircraft Design, Tech. Univ. of Munich, Garching bei München, Germany)

Advanced Air Mobility (AAM) vehicles like air taxis and cargo unmanned aerial vehicles (UAVs) operate close to urban areas. Therefore, it is necessary to keep the noise footprint of such aerial vehicles at a minimum to gain societal acceptance for AAM vehicle operations. Cargo UAVs are typically highly integrated in terms of functions leading to a high extent of aerodynamic interactions between propellers and the airframe. As a result of these installation effects, unsteady loading noise can become the dominant part of the aerial vehicle noise emissions which leads to a situation, where classical propeller noise reduction measures like blade tip Mach number reduction do not necessarily lead to a reduction in the overall noise emissions. Overcoming these uncertainties in propeller design necessitates propeller noise optimization on an aircraft configuration level. In this work, an existing propeller optimization framework for isolated propellers was extended accordingly: Consideration of propeller inflow velocity perturbations due to installation in the blade element momentum theory-based propeller performance prediction and utilization of a blade element method code (BEM) to consider the scattering of propeller noise on the airframe accounts for configuration noise aspects. Optimization results of a pusher propeller UAV configuration conclude this work.

8:25

2aNSa2. Computational optimization of trailing-edge designs to reduce airfoil self-noise. Behzad Amirsalari (Dept. of Mech. and Aerosp. Eng., Carleton Univ., 1125 Colonel By Dr., Ottawa, ON K1S 5B6, Canada, behzadamirsalari@cmail.carleton.ca) and Joana Rocha (Dept. of Mech. and Aerosp. Eng., Carleton Univ., Ottawa, ON, Canada)

This paper presents an investigation and optimization of Trailing Edge (TE) design to reduce airfoil self-noise using Computational Fluid Dynamics (CFD). The case for study is a NACA0012 airfoil with a chord length (C) of 0.2286 m, a varying Angle of Attack (AoA) between 0° and 15°, and free stream velocity between 35 and 70 m/s. The flow domain consists of a c-type domain with a length and height of 18C and 9C, respectively. The parametric mesh maintains a structured mesh on the entire domain for different designs and TE shapes. Simulations employ a hybrid Stress-Blended Embedded Large-Eddy Simulations (SB-ELES) model to calculate the flow properties. Different turbulence models are tested to address their performance in determining pressure fluctuations. A correlation length also accounts for spanwise effects in the Ffowcs-Williams and Hawkings (FW-H) acoustic analogy approach to forecasting the far-field noise. Furthermore, multi-objective optimization is employed to determine the optimum airfoil TE configuration for different flow velocities and AoAs. The optimum designs generate the lowest Sound Pressure Level (SPL) without significantly sacrificing the aerodynamic performance of the airfoil within specified parameters.

8:40

2aNSa3. Multirotor broadband noise modulation. Ze Feng (Ted) Gan (Aerosp. Eng., The Penn State Univ., 229 Hammond Bldg., University Park, PA 16802, tedgan@psu.edu), Vitor Tumelero Valente, Kenneth S. Brentner, and Eric Greenwood (Aerosp. Eng., The Penn State Univ., State College, PA)

Rotor broadband noise spectra are typically analyzed over time scales on the order of one or more rotor periods. However, modulation of the broadband noise spectrum with the blade passage frequency (BPF) has been shown to be significant for noise levels and perception of wind turbines and helicopters. In contrast, time-varying broadband noise has not been extensively studied for aircraft with many rotors, such as unmanned aerial vehicles (UAVs) or advanced air mobility aircraft. In this work, significant broadband noise modulation was measured in flight and anechoic chamber tests of hexacopter UAVs at various observer angles. This modulation is aperiodic with the BPF such that the modulation amplitude varies substantially between blade passages, even when the BPFs are controlled to be nearly constant between all rotors at all times. Furthermore, the azimuthal phasing between rotors greatly affects the measured modulation, such that the modulation of multiple rotors may be less than or greater than for a single rotor, depending on the phase offsets. The effects of phase variations on acoustic interactions between rotors is studied by comparing the sum of the modulation of individual rotors to the modulation of those rotors operating simultaneously. This is done not only using measurements, but also noise predictions made using PSU-WOPWOP. These results contribute understanding to how the noise modulation of rotors sum together, including the resulting directivity and aperiodicity.

Invited Papers

8:55

2aNSa4. Use-case study for a smaller German community and beyond, with realistic traffic scenario and three alternative vertiport locations. Michael W. Bauer (Munich Aeroacoustics, Kirchheim, Kirchheim 85551, Germany, sub@muc-aero.com)

Future individual air traffic will be based on air vehicles, such as drones and air-taxis. While drones, e.g., for delivery of goods may be processed decentralized, passenger air-taxis will be operated at vertiports in city centers, e.g., connected to big rail hubs, close to civil airports, but also in smaller communities in the vicinity of larger cities. Here, air-taxi noise—mainly take-off and approach—can impact residential areas inside the community but also in its neighborhood. In cooperation with a midsized community near to Munich/Germany, three potential vertiport locations in the community's area are investigated under the aspect of noise and potentially related noise annoyance. To create a use-case with results, transferable to similar situations, realistic numbers for daily air-taxi movements between the use-case vertiport and five neighbor cities are included. These connections imply two different types of air-taxis: one tilt-wing eVTOL for the ranges up to 75 miles, and one multicopter aircraft serving on distances up to 30 miles. The UAM noise use-case study is performed for all three vertiport positions, optimized flight paths, and different flight profiles. The results regarding community noise, but also for en-route noise between the neighboring vertiports, are discussed in this paper.

9:15

2aNSa5. Risks and benefits of air taxis as perceived in Germany. Hinnerk Eißfeldt (FL-SEG, DLR German Aerosp. Ctr., Bachstraße 94, Braunschweig, Lower Saxony 22083, Germany, hinnerk.eissfeldt@dlr.de)

In a telephone survey on the acceptance of civilian drones in Germany conducted at the end of 2022 a certain part of questions was dedicated to air taxis explicitly. While the attitude towards civilian drones tended to be slightly more positive than in the preceding study, in the again nationwide representative study attitudes concerning air taxis were revealed to be relatively balanced, with a slight negative tendency. Inferential statistical analyses showed factors such as age, gender, active experience with drones, and interest in environmental protection to be significantly associated with the attitudes towards both, civilian drones and air taxis. According to prior findings on the relevance of noise concerns for drone acceptance in general, the current study asked for some noise related information, revealing subjective noise sensitivity and general noise annoyance being associated with the acceptance of civilian drones as well as air taxis. Evaluating information from answers to free format questions about air taxis, this contribution reveals reasons why in Germany—although the contributing factors are the same—the general attitude towards air taxis is less positive compared to civil drones, and what role noise considerations might play.

9:35–9:50 Break

Contributed Papers

9:50

2aNSa6. Aeroacoustic and aerodynamic analysis of a Conceptual Unmanned Aerial Vehicle (UAV) using Computational Fluid Dynamics (CFD). Mayur Nunkoo (Carleton Univ., Ottawa, ON, Canada) and Joana Rocha (Carleton Univ., 1125 Colonel By Dr., Ottawa, ON K1S 5B6, Canada, Joana.Rocha@carleton.ca)

The current investigation aims to conduct a comprehensive analysis of the aerodynamics and aeroacoustics of Manta, a conceptual Unmanned Aerial Vehicle (UAV) in the Bio-inspired Environmentally Friendly Aerial Vehicle (BEFAV) project at Carleton University. The primary objective involves exploring the effects of a Blended-Wing-Body design on critical parameters such as lift, drag, and Sound Pressure Level (SPL), through Computational Fluid Dynamics (CFD). This analysis considers the cruise phase of flight, featuring an airspeed of 67 m/s at an altitude of 1500 m. The incorporation of bio-inspired elements into Manta's design, aiming at increased operational efficiency and noise reduction, is also investigated.

10:05

2aNSa7. The reduction of noise and vibration of composite panels in aircraft through multi-dimensional particle swarm optimization. Noah Veenstra (Carleton Univ., 1125 Colonel By Dr., Ottawa, ON K1S 5B6, Canada, NoahVeenstra@cmail.carleton.ca) and Joana Rocha (Carleton Univ., Ottawa, ON, Canada)

The reduction of noise and vibrations on aircraft poses a unique challenge. This paper explores the use of multi-dimensional Particle Swarm Optimization (PSO) to reduce vibrations and transmitted noise in composite fuselage panels excited by turbulent boundary layer flow. Turbulent boundary layer Power Spectral Density (PSD) data is used to simulate the excitation of simply supported aircraft panels. A comparison is made between isotropic aluminum 2024-T3 panels and optimized composite panels of varying composition for the assessment of success in noise transmission reduction as well as structural function. The algorithm for PSO is implemented in Python and iterates through composites with varying thickness, ply makeup, ply orientation, and materials. Python allows for complex algorithms to be easily interfaced with Finite Element Analysis (FEA) programs. In the current study, ANSYS is used to determine the spectral response of the panels subject to turbulent flow using the superposition of the modes of vibration. During the optimization process, optimization parameters are updated in each iteration based on the success of reducing the spectral response of the panel without compromising its structural integrity or increasing its weight beyond a reasonable threshold.

10:20

2aNSa8. Acoustic characterization and optimization of a subsonic closed loop wind tunnel. Nicholas P. Cunnington Bourbonniere (Mech. and Aerosp. Eng., Carleton Univ., Ottawa, ON, Canada), Joana Rocha (Carleton Univ., 1125 Colonel By Dr., Ottawa, ON K1S 5B6, Canada, Joana.Rocha@carleton.ca), and Peter Waudby-Smith (Aiolos Eng. Corp., Toronto, ON, Canada)

A low-speed closed-loop wind tunnel with an open-jet test section having a maximum windspeed of $60\,m/s$ is being modified for future aeroacoustic testing. The wind tunnel is new to Carleton University as of 2021 and no acoustic characterization tests have been completed until recently. The objective of this research is to acoustically characterize the wind tunnel before and stepwise through its acoustic modifications. Multiple tests have been made using 1/4-inch microphones at different locations inside the plenum to record the existing out-of-flow background noise. The first planned wind tunnel modification is a height extension to the current plenum with

dimensions $0.74 \times 1.8 \times 1.3$ m $(H \times L \times W)$ to increase the observer distance for the microphones. Once the plenum volume is increased, acoustic treatment will be applied to the interior of the plenum to create a hemi-anechoic measurement environment. An in-flow microphone mount was designed and manufactured with computer-controlled vertical traversing; it will be installed after the plenum modifications are complete.

Invited Papers

10:35

2aNSa9. Individual response trends to urban air mobility noise in a laboratory study. Aaron B. Vaughn (Structural Acoust. Branch, NASA Langley Res. Ctr., 2 N Dryden StMS 463, Hampton, VA 23681-2199, aaron.b.vaughn@nasa.gov) and Andrew Christian (Structural Acoust. Branch, NASA Langley Res. Ctr., Hampton, VA)

Advanced Air Mobility, of which Urban Air Mobility (UAM) is a subset, presents new opportunities for more dynamic aviation transportation systems. It is essential to understand the human response to the noise of these vehicles for sustainable operations. This research aims to investigate the relationship between the noise level and number of events on individual human annoyance to UAM vehicle noise. A "noise and number" laboratory psychoacoustic study was conducted in the Exterior Effects Room at NASA Langley Research Center in January 2023. A total of 38 participants listened to 4-minute audio clips of UAM vehicle flyover noise and provided their annoyance response on an 11-point numerical scale. A test hypothesis was that annoyance to multiple flyovers can be found by adding the annoyance responses to individual flyovers. Results for the overall test subject pool suggest that annoyance grows faster than hypothesized as the number of aircraft events increases. This trend of faster annoyance growth only seems to hold when the individual flyovers are heard as distinct events. Parsing the data down to individuals provides further insight into the mechanisms contributing to the increased annoyance with number of events.

10:55

2aNSa10. Studying holistic perception and response to unmanned aerial vehicle sound within acoustic environments: Bridging the gap between annoyance and soundscape. Marc C. Green (Acoust., Univ. of Salford, Newton Bldg., University of Salford, Salford M5 4BR, United Kingdom, m.c.green@salford.ac.uk), Michael J. Lotinga (Acoust., Univ. of Salford, Salford, United Kingdom), and Antonio J. Torija Martinez (Acoust. Res. Ctr., Univ. of Salford, Manchester, United Kingdom)

Given the projected increase in Unmanned Aerial Vehicle (UAV) deployment in the coming years, there is increased interest in studying their auditory impact. Most previous studies into perception of UAV sound have narrowly focused on noticeability and annoyance, typically presenting sound events in isolation. Whilst it is useful to understand these effects, isolated stimuli are not reflective of how these sources are likely to be experienced within ambient soundscapes. Furthermore, asking only about annoyance could bias participants, in effect prompting them to consider certain sound sources annoying. This can be seen as an instance of the conventional "environmental noise" approach, with the focus on disturbance and annoyance, in contrast with a "soundscape" approach, which adopts a more holistic focus. In soundscape, environmental sound is considered as a resource, with the potential for positive or negative effects, rather than only as a waste byproduct (noise). Here we present the results of a study involving UAV stimuli embedded within recorded acoustic environments. Participants evaluated sounds using a "Self-Assessment Manikin" to rate their affective response alongside a conventional noise annoyance scale. Associations between these two approaches are investigated, in an attempt to link "environmental noise" and "soundscape" literature with regards to UAV sound.

Session 2aNSb

Noise, Architectural Acoustics and ASA Committee on Standards: Soundscape—Focus on Applications I

Brigitte Schulte-Fortkamp, Cochair
HEAD Genuit Foundation, Ebert Straße 30 a, Herzogenrath, 52134, Germany

Bennett M. Brooks, Cochair
Brooks Acoustics Corporation, 49 N. Federal Highway, #121, Pompano Beach, FL 33062

Chair's Introduction—8:30

Invited Papers

8:35

2aNSb1. Applying soundscape: A matter of participation. Brigitte Schulte-Fortkamp (HEAD Genuit Foundation, Ebert Straße 30 a, Herzogenrath 52134, Germany, bschulte_f@web.de) and Bennett M. Brooks (Brooks Acoust. Corp., Pompano Beach, FL)

Defined as the acoustic environment understood by people in context, the soundscape encapsulates the myriad sounds that shape our daily lives (ISO12913 series). In the intricate tapestry of our sensory experiences, the soundscape emerges as a profound element, weaving together the diverse threads of auditory stimuli and social implications that surround us. Soundscape is not merely an assortment of sounds; rather, it is a complex interplay of natural and human-made elements that contribute to the acoustical and social identity of a place. Participation in the planning and modification of an acoustic environment requires an active and creative approach, it calls for a targeted engagement with the proposed redesigns or reorganizations. Participating in the soundscape approach also entails promoting acoustic ecology and the responsible management of acoustic environments. This involves efforts to minimize noise pollution, protect natural soundscapes, and create spaces where diverse sounds can coexist harmoniously. The paper will discuss different aspects of participation that offers numerous benefits, and as a participatory experience it opens new avenues for creativity, well-being, and community building in our ever-evolving acoustic living situations.

8:55

2aNSb2. Soundscape techniques applied to urban planning goals. Bennett M. Brooks (Brooks Acoust. Corp., 49 N. Federal Hwy., #121, Pompano Beach, FL 33062, bbrooks@brooksacoustics.com) and Brigitte Schulte-Fortkamp (HEAD Genuit Foundation, Herzogenrath, Germany)

The soundscape method is a powerful tool for understanding the perception of our acoustical environment. Urban planning is the technical and political process by which we manage and mold our cities. Smart growth is a movement within urban planning which emphasizes the livability of our surroundings. These disciplines all seek to shape the form of our communities to improve the quality of life for those who live, work, and play there. Yet, there has been little formal interaction between soundscaping and urban planning. It is imperative that soundscapers who wish to implement improved urban designs and interventions adopt the language and methods of urban planners in order to succeed and accomplish their mutual goals. Soundscape protocols must engage the urban planning process, including the comprehensive, or master, plan to achieve implementation on a city-wide scale. Several recent initiatives which illustrate the integration of soundscape techniques with urban planning methods are presented and discussed.

9:15

2aNSb3. Exploring soundscape methods and interventions in the Urban Soundscape of 3 cities. Gary W. Siebein (Siebein Assoc.,Inc., 625 NW 60TH St., Ste. C, Gainesville, FL 32607, gsiebein@siebeinacoustic.com), Keely M. Siebein, Marylin Roa, Gary Siebein Jr., Nicolas Ospina (Siebein Assoc.,Inc., Gainesville, FL), and Martin Gold (School of Architecture, Univ. of Florida, Gainesville, FL)

Vital mixed-use urban communities in the United States struggle to achieve balances among citizens moving back into cities, commercial activities, and entertainment venues that many people find desirable in live, learn, work, and play environments. Dynamic documentation, analysis, design, codes, and enforcement activities are required to achieve this balance in rapidly evolving, sustainable cities. Active engagement of the acoustical communities and the full range of stakeholders in each case were essential in understanding and addressing the issues. Multiple meetings with parties individually and in groups provided ways for all to understand the points-of-view of others as a building block to achieve consensus. Simple, but sophisticated, measurement and modeling of the soundscape were necessary elements of the methods used. Case studies in 3 large cities are presented of needs, issues, methods, analysis, and proposed solutions to a wide variety of acoustical issues encountered in the cities. Reflections on soundscape theory posed by the case studies and possible adjustments to theory are discussed.

9:35

2aNSb4. A soundwalk tutorial and exercise in Sydney, Australia. David s. Woolworth (Roland, Woolworth & Assoc., 356 County Rd. 102, Oxford, MS 38655-8604, dwoolworth@rwaconsultants.net)

The previous Acoustical Society of America's meeting #185 in Sydney, Australia joint with Australian Acoustical Society (AAS), Western Pacific Commission for Acoustics (WESPAC) and Pacific Rim Underwater Acoustical Conference (PRUAC) provided an opportunity to for an introductory tutorial on soundwalks followed by a walk through the city to gather measurement data and perceptive feedback from the participants. This presentation will provide the structure of the tutorial and preparation and results of the walk with analysis.

9:55–10:10 Break

10:10

2aNSb5. Psychoacoustic analyses of urban noise from long-term monitoring—Findings and outlook. Andre Fiebig (Eng. Acoust., TU Berlin, Einsteinufer 25, Berlin 10587, Germany, andre.fiebig@tu-berlin.de), Moritz Schuck (Eng. Acoust., TU Berlin, Berlin, Germany), Timo Haselhoff, and Susanne Moebus (Inst. for Urban Public Health, University Hospital Essen, Univ. Duisburg-Essen, Essen, Germany)

Urban noise is usually assessed using sound pressure level indicators to assess harmful noise effects. However, studies have shown that human responses to environmental noise are driven by psychoacoustic properties even beyond sound pressure level indicators. Although the value of psychoacoustic parameters for an improved characterization of environmental noise appears plausible and has often been shown in laboratory studies, a systematic examination of psychoacoustic parameters over long measurement intervals is lacking. In the research project SALVE+, long-term acoustic measurements at several locations in a German city were subject to psychoacoustic analyses and the location-dependent behavior of psychoacoustic parameters over time was examined. The measurements were analyzed to determine the acoustic quality of urban sound beyond sound pressure level and noise annoyance. Based on statistical analyses, the psychoacoustic properties of urban locations considering their land use are discussed regarding an improved characterization of acoustic environments. Based on this approach, the suitability of psychoacoustic parameters for mapping of spatial and temporal patterns is investigated. The presentation aims to highlight the value of noise monitoring in cities and analyze opportunities for improved urban noise management that additionally considers issues such as restoration quality and health promotion.

10:30

2aNSb6. Soundscape, attention and cognitive load. Adrian K. C. Lee (Speech & Hearing Sci., Univ. of Washington, 1417 NE 42nd St., Seattle, WA 98105, akclee@uw.edu)

The cocktail party problem was first coined by E Colin Cherry in 1953 and it describes the archetypal challenge of listening in a complex soundscape (e.g., multiple talkers conversing vying for your attention in a crowded restaurant). In the past two decades, the field of psychoacoustics has steadily marched towards understanding how we listen in these naturalistic environments. Many studies have focused on studying the psychological and physiological aspects of auditory scene analysis and object-based attention as well as how these processes differ in typical listeners from others with listening difficulties. In recent years, there has also been a burgeoning interest in understanding how cognitive load and listening effort affect our ease of listening in different situational contexts (e.g., talking while driving). In this talk, a brief survey of modern psychoacoustic experimental approaches will be presented in the hope to spur new collaborative research ideas with those who study soundscape.

10:50

2aNSb7. Soundscape augmentation for people with dementia requires accounting for disease-induced changes in auditory ccene analysis. Arezoo Talebzadeh (Ghent Univ., 126 Tech Ln. Ghent Sci. Park, Ghent 9052, Belgium, arezoo.talebzadeh@ugent.be), Dick Botteldooren, and Paul Devos (Information Technol., Ghent Univ., Ghent, Belgium)

Recently, there has been an increased interest in adapting the sonic environment to support people with cognitive difficulties, such as dementia. Research shows that incorporating "pleasant" sounds into the environment positively impacts behaviour and reduces psychological symptoms of dementia. Introducing sound into the acoustic environment creates an enhanced auditory experience, an augmented soundscape, resulting in an improved interpretation of the environment. People with dementia experience changes in their perception, which includes misperceptions, misidentifications, hallucinations, delusions, and time-shifting. Sound augmentation can support a better understanding of the environment and help in navigation time during the day. Dementia is a broad name for a degenerative disease; different types of dementia result in different syndromes and diverse auditory scene analyses. Some key auditory symptoms of different variants of the disease are auditory hallucinations, auditory disorientation, increased sound sensitivity, auditory agnosia (difficulty processing auditory input), agnosia for environmental sounds, amusia (tonal deafness) and Musicophilia. These different syndromes of auditory perception need more understanding when designing a soundscape augmentation for people with dementia. This talk aims to discuss different auditory symptoms of dementia based on a literature review and introduce ways to design an augmented soundscape to foster individual auditory needs.

11:10

2aNSb8. The Catalogue of Soundscape Interventions (CSI) project—A tool to bridge soundscape research and practice. Francesco Aletta (Univ. College London, Central House, 14 Upper Woburn, London N19DD, United Kingdom, f.aletta@ucl.ac.uk), Xiaochao Chen (Univ. College London, London, United Kingdom), Cleopatra Moshona (TU Berlin, Berlin, Germany), Jian Kang, Andrew Mitchell, Tin Oberman (Univ. College London, London, United Kingdom), Brigitte Schulte-Fortkamp (HEAD Genuit Foundation, Herzogenrath, Germany), and Andre Fiebig (TU Berlin, Berlin, Germany)

The Catalogue of Soundscape Interventions (CSI) project addressed the growing interest in urban soundscapes by establishing a comprehensive taxonomy for soundscape design. The project development is described, emphasizing its role in connecting soundscape research with practical applications. The proposed concepts for soundscape design interventions and other soundscape interventions are discussed, which seek to inform the creation of "soundscape design briefs." These briefs, intended for use by local authorities, aim to enhance communication among soundscape consultants and researchers, and other relevant stakeholders. This work also presents the intersection of the CSI project with the technical specifications outlined in the ISO/AWI TS 12913-4 Acoustics Soundscape Part 4: Design and Intervention, currently in development. By aligning with emerging standards, the CSI project seeks to contribute to a unified framework for soundscape practices, and to establish a model to describe phases and necessary action points in the life cycle of a soundscape design intervention.

Contributed Paper

11:30

2aNSb9. Taipei performing arts center surrounding soundscape study and its application. Roxana Ghadiri (Dept. of Architecture, National Taiwan Univ. of Sci. and Tech., 43 Keelung Rd. Sec. 4, RB809, Taipei 106, Taiwan, M11213803@mail.ntust.edu.tw), Shiang-I Juan, Stijn Zeger van Brug, Nikita Grace Manullang, Juliana Manuela Muet, Tuan Sanh Diep, Khaing Thinzar, Ni Made Putri Indriyani, Phoa Angela Grace Wibowo, Gabriela Niederberge, Clarissa Averina, Chau N. Truong, and Lucky S. Tsaih (Dept. of Architecture, National Taiwan Univ. of Sci. and Tech., Taipei, Taiwan)

Taipei Performing Arts Center designed by architect Rem Koolhaas and his firm OMA, features three theaters stacked vertically. The cube-shaped structure features a distinctive geometric facade, serving as the seating areas for the three main theaters. These giant facades, due to their sloped seating surfaces, could also function as significant sound-reflecting and focusing panels, potentially amplifying the nearby MRT and traffic noise. The project was strategically located near Shihlin Nightmarket for the area's cultural and economic growth. ISO-12913 soundscape study method is used to document essential sound sources, paths, and levels. The average sound pressure level of 67.4 decibels around the building provides an indication of the ambient noise in the immediate vicinity. While the acoustic comfort level in the plaza beneath the prominent facades may not be optimal, it is crucial to acknowledge their dual role as guiding individuals into the performance hall and serving as a buffer between external traffic and internal events. In addition to evaluating the impact of building mass and envelope design decisions on sound levels, this study uncovers a noteworthy finding related to urbanized traffic beeping. The observed and mapped patterns of local traffic sounds reveal unique cultural messages embedded in the auditory landscape.

Session 2aPA

Physical Acoustics, Noise, Engineering Acoustics, and Signal Processing in Acoustics: Wind Noise

Gregory W. Lyons, Cochair
DEVCOM Army Research Laboratory, 2800 Powder Mill Rd, Adelphi, MD 20783

W. C. K. Alberts, Cochair
CCDC-Army Research Laboratory, 2800 Powder Mill Road, Adelphi, MD 20783

Chair's Introduction—8:00

Invited Papers

8:05

2aPA1. Predicting infrasonic wind noise levels using local topographic features. Garth Frazier (NCPA, Univ. of MS, NCPA, University of MS, P.O. Box 1848, Oxford, MS 38677, frazier@olemiss.edu), Roger M. Waxler (NCPA, Univ. of MS, University, MS), and Claus Hetzer (NCPA, Univ. of MS, Tempe, AZ)

This presentation describes a method for predicting RMS wind noise levels within a user-specified infrasonic frequency band (e.g., 1–10 Hz) using local topographic features. This capability is especially valuable when performing site selection for infrasound sensor array deployment. The method is based on building models using measured infrasound data from known sensor site locations and the local topographic features corresponding to the site. The presented results correspond to using features such relative terrain elevation and relative vegetation height obtained from publicly available sources. Mean wind speed and direction are also included as predictive parameters and these can be from data measured at the infrasound sensor site or from estimates obtained from a tool such as the Weather Research Forecast (WRF) model as was utilized for this investigation. A novel feature of the approach is that the terrain specification is not uniquely site specific but also depends on wind direction. Essentially, the topographic features are specified relative to wind direction and not in absolute coordinates. This enables a much richer set of samples from which to build the predictive models. While many options are potentially available for model building, this work focused on the use of multiple-layer artificial neural networks as a basis for regression. The results presented correspond to data from several infrasound sensor sites in the U.S Array Project.

8:25

2aPA2. The use of *in situ* infrasound calibration to correct wave parameter estimation. Samuel Kristoffersen (CEA, DAM, DIF, Bruyeres-le-Chatel, France), Alexis Le Pichon (CEA, DAM, DIF, Arpajon, France), Michaela Schwardt (PTB, Braunswchweig, Germany), Paul Vincent (CEA, DAM, DIF, Bruyeres-le-Chatel, France), Benoit Doury (CTBTO, Vienna, Austria), Franck Larsonnier (CEA, DAM, DIF, Bruyeres-le-Chatel, France), Christoph Pilger (BGR, Hannover, Germany), and Patrick Hupe (BGR, Stilleweg 2, Hannover 30655, Germany, patrick.hupe@bgr.de)

As part of the Comprehensive Nuclear-Test-Ban Treaty (CTBT), the International Monitoring System (IMS), which includes infrasound stations, has been established to monitor nuclear testing, as well as many other infrasound sources of interest to the scientific community. To minimize the effects of wind-noise, wind-noise reduction systems (WNRS) are installed as part of each sensor. In order to provide the best wave parameter (azimuth, trace velocity, amplitude) estimates, the response of the WNRS must be properly taken into account. Therefore, it is important to perform an *in-situ* calibration of the sensor using a co-located reference sensors and ambient signals to determine this response. This *in-situ* calibration can be used to monitor the status of the sensors, and provide feedback to the station operators. Experiments were performed at the IS26 site in Germany using a temporary WNRS to provide quantitative measurements of the effects of the WNRS and its calibration on the wave parameter estimation. These calibration results were used to provide corrections to the raw signal data for retrieval of the corrected wave parameters and were compared to the IS26 measurements, demonstrating that accurate measurements can be retrieved using the *in-situ* calibration.

8:45

2aPA3. Richard Raspet's contributions to our understanding of low-frequency wind noise. Jeremy Webster (Los Alamos National Labs, LANL, MS F665, Bikini Attoll Rd., Los Alamos, NM 87544, jwebster@lanl.gov)

Richard Raspet's research in wind noise spanned three decades and took the acoustics community from a rudimentary understanding of how low-frequency wind noise is generated in pressure sensors to a firm theoretical understanding derived from first principles in fluid dynamics. This talk will present a technical overview of his work starting with early quantitative studies, progressing through the development of the theoretical framework for wind noise studies, and finishing with a summary of research including wind noise at the ground surface, wind noise under tree canopies, wind fences, and wind induced seismic noise.

9:05

2aPA4. Towards a rapid-distortion theory model of turbulent flow for wind noise within an impermeable windscreen. Gordon M. Ochi (U.S. Army ERDC, 2902 Newmark Dr., Champaign, IL 61822, Gordon.M.Ochi@erdc.dren.mil) and Carl R. Hart (U.S. Army Engineer Res. and Development Ctr., Cold Regions Res. and Eng. Lab., Hanover, NH)

Current models for the wind noise reduction (WNR) of microphone windscreens in atmospheric turbulence generally work under the assumption of a homogeneous surface pressure averaging theorem. Under this theorem, the surface pressures should average in such a way that the noise spectrum at low wavenumbers relative to the windscreen dimension should approach that of an unscreened microphone. Contrasting this, experiments have actually observed a significant WNR at these wavenumbers. In this work, we examine a Rapid Distortion Theory model for linking turbulent distortion to the unsteady pressure received inside the windscreen. The theoretical background and underlying assumptions of Rapid Distortion Theory are examined in detail [R. Zamponi, *et al*., J. Fluid Mech. 915**:**A27 (2021)], then results for the case of an impermeable cylinder are presented. Future work on the project involving both theoretical development and experimental verification are then discussed.

9:25

2aPA5. Pseudospectrum-based methods for estimating the wind speed and direction using coherence decay-dependent steering vectors. Daniele Mirabilii (WSAudiology, Henri-Dunant-Straße 100, Erlangen, Bavaria 91058, Germany, daniele.mirabilii@wsa.com) and Emanuël A. Habets (Int.. Audio Labs., Erlangen, Bavaria, Germany)

In a recent work, we employed acoustic array processing to measure the wind speed and direction based on wind-induced noise recorded with multiple microphones. In particular, we proposed beamforming and signal subspace-based methods for estimating the wind speed and direction via pseudo-spectrum maximization. Based on experimental observations using a compact microphone array, we make more stringent assumptions on the wind-induced noise advection to delineate its weakly coherent propagation. To account for the wind turbulence dissipation across space and frequency, we introduce a speed- and direction-dependent anisotropic decay factor that models the relative propagation of the wind-induced noise source between microphones. The decay term is incorporated in the relative transfer functions of the noise source and, consequently, in the steering vectors used for the computation of the pseudo-spectra. Finally, we re-evaluate the proposed methods in terms of speed and direction accuracy under these new assumptions.

9:45–10:05 Break

Contributed Papers

10:05

2aPA6. Abstract withdrawn.

10:20

2aPA7. Dealing with wind noise on resource constrained acoustic sensors. W. C. K. Alberts (DEVCOM Army Res. Lab., 2800 Powder Mill Rd., Adelphi, MD 20783, william.c.alberts4.civ@army.mil) and Gregory W. Lyons (DEVCOM Army Res. Lab., Adelphi, MD)

Single and multiple microphones in small or hand-held devices, i.e., mobile phones, are often single port and without windscreens. In a typical outdoor situation, this configuration can be subject to strong impulsive wind noise events and wind-induced tones. Smart mobile phones can sometimes utilize their significant computational power to employ sophisticated signal processing to reduce or remove wind-associated noise. However, in systems where battery conservation is paramount and computational processing resources are severely limited, alternative strategies must be employed. This presentation will discuss exemplar wind noise data from a single channel, resource-constrained system, methods for wind noise reduction, and results.

10:35

2aPA8. Wind noise due to flow induced around a windscreen in large-scale turbulence. Gregory W. Lyons (DEVCOM Army Res. Lab., 2800 Powder Mill Rd., Adelphi, MD 20783, gregory.w.lyons3.civ@army.mil)

Turbulent wind noise on microphones is a ubiquitous problem for sound measurement and recording outdoors. Windscreens, such as open-cell foam spheres, are effective at reducing wind noise. The unsteady pressure sources induced on the windscreen surface must contribute less noise on average than the stagnation pressure on a bare microphone, or windscreens would not be effective. To quantify this noise reduction mechanism, a model is presented for the unsteady pressure within an impermeable windscreen of arbitrary shape immersed in an initially isotropic turbulent flow. Only turbulent scales larger than the windscreen are considered, which contribute the most intense low-frequency band of the noise spectrum in most practical situations. The inhomogeneous velocity field around the windscreen is modeled in terms of an induced irrotational blocking field that meets the kinematic boundary condition. The interior pressure is expressed as a convolution of the surface pressure field with a filter function in wavevector space. The wind noise wavevector spectrum is then shown to be only a function of the inflow isotropic velocity spectrum and the windscreen shape. Solutions are obtained for the pressure frequency spectrum within a spherical windscreen for a von Kármán turbulence model and compared with prior results in literature.

10:50

2aPA9. Very short time Fourier transform for turbulent analysis. Roger Oba (Acoust. Div., US Naval Reseach Lab., 4555 Overlook Ave. S.W., Washington, DC 20375, roger.oba@nrl.navy.mil) and Ravi Ramamurti (Naval Res. Lab., Washington, DC)

Nominally stationary acoustic sensors, e.g., buoy suspended or bottom anchored, are subject to various environmental flows. Direct numerical simulation of hydrodynamic flow around the acoustic sensor housing show turbulence along the housing surface. Time-space analysis of pressure variation along the surface of the housing shows significant acoustic-like surface interaction at low frequencies, even at very low mean flow velocity. In this case turbulence manifests as intermittently generated, irregular, coherent vortex structures that separate from the surface of the housing and advect downstream. These structures drive rapid surface pressure transients that propagate within the housing to the acoustic sensors as noise. Fractional Fourier Analysis provides joint time-frequency methods to extract transient features of turbulence. Use of Hermite-Gauss functions and phase relations are of particular interest. [This research is supported by 6.2 NRL base program sponsored by the Office of Naval Research. Distribution Statement A: Approved for public release. Distribution unlimited.]

11:05

2aPA10. A reduction in near-field hydroacoustic flow noise using shark skin-inspired surfaces. Jonathan Stocking (US Naval Res. Lab., 4555 Overlook Ave., SW, Bldg. 2, Rm. 129L, Washington, DC 20375, jonathan.stocking@nrl.navy.mil), Kaushik Sampath (KS Res. Inc., Greenbelt, MD), Nicole Xu (Univ. of Colorado, Boulder, Boulder, CO), Jason Geder (US Naval Res. Lab., Washington, DC), and Silvia Matt (US Naval Res. Lab., Stennis, MS)

Recent studies have demonstrated improved hydrodynamic performance for hydrofoils and uncrewed underwater vehicles (UUVs) using passive surface geometries such as riblets and shark skin-inspired denticles. However, little is known about the impact of added surface roughness on the production of flow-induced noise, a critical design consideration for UUVs in sensitive environments. We present here experimental results of near-field hydroacoustic noise measurements for boundary layer flow over smooth and bioinspired denticle-covered surfaces. A 3 in. wide and 10 in. long segment of staggered denticles was fabricated in-house using photopolymer 3D printing, and flow and acoustic measurements were made in a custom-built flow channel using particle image velocimetry (PIV) and miniature hydrophones. Comparing acoustic power spectra over flat and denticle-covered plates, we find both a scale- and frequency-dependent response in radiated noise. At low flow speeds ($Re \sim 20{,}000$), the denticle surface shows a 20 dB enhancement in radiated noise at frequencies below 1000 Hz, but no change at higher frequencies. While at high flow speeds ($Re \sim 60\,000$), the denticle surface reduces radiated noise by $\sim$10 dB for frequencies between 4000 and 8000 Hz. Analysis of PIV data will elucidate the flow physics responsible for these effects. [Work sponsored by Office of Naval Research.]

TUESDAY MORNING, 14 MAY 2024 — ROOM 208, 9:30 A.M. TO 11:20 A.M.

Session 2aPP

Psychological and Physiological Acoustics: Toward More Inclusive Research Practices in P&P II

Monita Chatterjee, Cochair
Center for Hearing Research, Boys Town National Research Hospital, 555 N 30th St., Omaha, NE 68131

Peggy Nelson, Cochair
University of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55455

Invited Papers

9:30

2aPP1. Gender-expansive listeners invite us to reconsider voice categorization as a simple matter of pitch and timbre. Jay Marchand Knight (Individialized (Science), Student, 7547 rue Centrale, Lasalle, QC H8P 1K4, Canada, juanita.marchand@gmail.com)

Past literature has focused on pitch and timbre cues as the main determinants of the voice of an actor (a speaker or a singer). But in real life, judges rely on non-auditory cues such as the actor's facial features, height, body size and shape. Resisting these visual biases could provide more accurate assessment of a voice but requires a stronger dissociation between auditory attributes and physical appearance, an ability hypothetically acquired by gender-expansive participants. In three online studies, we examine *Faching* (allocating voices into traditional categories). In study 1, 166 participants (85 cis, 81 gender-expansive) rated 144 audio (A) samples from 18 different actors (3 from each major *Fach* category, excluding countertenor) along a slider labeled low/dark to high/bright. Participants then guessed the voice types of the same actors in silent videos (V), before rating them in AV combinations. The cis group exhibited 30% more visual bias than the gender-expansive group. To further understand how gender-expansive participants obtain this benefit, we manipulated the fundamental frequency (in study 2) or the vocal tract length (in study 3) by $\pm$ 3 semitones from the original stimuli. Preliminary findings suggest that the two groups differ in the face of timbre but not pitch manipulations.

9:55

2aPP2. Comparing LTASS across languages using natural and AI speech samples. Abhijit Roy (Commun. Sci. and Disord., Northwestern Univ., 124 Callan Ave., Apt. 2B, Evanston, IL 60202, abhijitroy2025@u.northwestern.edu), Ann Bradlow (Linguist, Northwestern Univ., Evanston, IL), and Pamela E. Souza (Commun. Sci. and Disord., Northwestern Univ., Evanston, IL)

Hearing aid prescription protocols rely on estimates of speech spectra. One such estimate, the International Long-Term Average Speech Spectrum (ILTASS), is a common reference for the distribution of spectral energy within speech signals. However, recent research has suggested differences in speech spectra between some languages and the ILTASS standard. These disparities raise questions regarding the applicability of a language-general approach to hearing aid gain application. This study presents a comparative analysis of Long-Term Average Speech Spectra (LTASS) across various languages. Utilizing speech samples from both human and AI sources, LTASS was measured for speakers of different languages. Subsequently, these LTASS profiles were contrasted with the established

ILTASS reference. Results reveal that while LTASS was very similar across languages, deviations were observed between LTASS derived from the human speech and AI speech data compared to ILTASS. Specifically, measured LTASS demonstrated a reduced 500 Hz peak in contrast to ILTASS. This research underscores the importance of accounting for spectral variations introduced during the recording process in the field of speech and hearing research. It also emphasizes the importance of standardized recording protocols to enhance the precision of hearing technologies.

Contributed Papers

10:20

2aPP3. The relationship in phonotactics between cross-linguistic sonority principles and within-language probabilistic distribution of segment sequences. Peiman Pishyar-Dehkordi (Linguist Dept., Univ. of Canterbury, University of Canterbury, Private Bag 4800, Christchurch 8140, New Zealand, peiman.pishyardehkordi@pg.canterbury.ac.nz)

The worlds' languages contain a variety of cross-linguistic phonotactic patterns, in which certain sound sequences are more likely to occur in languages than others. Are these patterns also reflected as probabilistic distributions within individual languages? We investigate this question in the context of sonority constraints on syllable formation. Phonotactic patterns in syllable structure have been argued to be governed by sonority hierarchies which are loosely based on the acoustic intensity of speech sounds. Cross-linguistic sonority principles argue that the simplest syllable is one with the maximal rise in sonority at the beginning, and the minimal drop in sonority at the end. We also know that in addition to categorical constraints on their phonology, languages contain probabilistic phonotactic patterns within the sequences that they allow, meaning that in each language some sequences tend to be over-represented, and some sequences under-represented. If cross-linguistic patterns are also reflected as within-language probabilistic constraints, we hypothesize that in a language that allows both simple and complex syllables, simpler syllables will be more frequent than more complex ones. To test this, syllables within 8 languages were explored. Our findings are generally compatible with the above hypothesis for all the 8 languages, suggesting that there is a general alignment between sonority principles as cross-linguistic universals and within-language probabilistic distribution of segment sequences within syllables.

10:35

2aPP4. Lack of racial diversity in language science journals. Olivia Tobin (Linguist, Univ. of Iowa, Iowa City, IA), Paras B. Bassuk, Samantha Chiu (Psychol. Brain Sci., Univ. of Iowa, Iowa City, IA), and Ethan Kutlu (Linguist, Univ. of Iowa, 459 Phillips Hall, Iowa City, IA 52242, ethan-kutlu@gmail.com)

Numerous studies and reports have pointed out the lack of diversity in scientific spaces (Liu *et al*., 2023; Singh *et al*., 2023) which has a direct impact on shaping our research practices, questions, and scientific pursuits in general (Kutlu and Hayes-Harb, 2023). While the lack of diversity in science has been documented widely, there is no report on the diversity of editorial board members of language science journals, who play an integral role in diversifying science. This ongoing study aims to document to what extent language science journals engage with diversity and how diversity is represented in these spaces. To achieve this goal, we looked at 100 language journals and their editorial board members list. We hand-coded the editorial board members' perceived gender and race. We used multiple sources to check our coding (e.g., Wikipedia, websites, videos). Our preliminary findings suggest that while there is gender balance in editorial spaces, there was a significant lack of racial representation in editorial space (i.e., more White scholars in editorial spaces). Our next step is to collect archival data on editorial board members, document systematic inequalities, and provide suggestions for how racial diversity can be increased in editorial spaces.

10:50–11:20
Panel Discussion

Session 2aSA

Structural Acoustics and Vibration, Engineering Acoustics and Physical Acoustics: Acoustic Metamaterials I

Christina Naify, Cochair
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Alexey Titovich, Cochair
Naval Surface Warfare Center, Carderock Division,

Bogdan-Ioan Popa, Cochair
Univ. of Michigan, 2350 Hayward St., Ann Arbor, MI 48109

Kathryn Matlack, Cochair
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Dylan Kovacevich, Cochair
Mechanical Engineering, University of Michigan, 2350 Hayward St., Ann Arbor, MI 48109

Abigail D. Willson, Cochair
Acoustics, Penn State University, PO Box 30, Mail Stop 3220B, State College, PA 16804

Chair's Introduction—9:00

Invited Paper

9:05

2aSA1. Energy absorption from an oscillating fluid using an engineered acoustic diode. Rico Schmidt (Mech. and Aerosp. Eng., Univ. at Buffalo (SUNY), Buffalo, NY), Indradip Roy (Mech. Eng., Purdue Univ., West Lafayette, IN), Hosam Yousef (Mech. and Aerosp. Eng., Univ. at Buffalo (SUNY), Buffalo, NY), Alex Aboueria, Carlo Scalo (Mech. Eng., Purdue Univ., West Lafayette, IN), and Mostafa Nouh (Mech. and Aerosp. Eng., Univ. at Buffalo (SUNY), 240 Bell Hall, Attn: Mostafa Nouh, Buffalo, NY 14260, mnouh@buffalo.edu)

Over the past decade, engineered subsurface structures inspired by acoustic metamaterial configurations have shown promising results in boundary layer control through different energy stabilization mechanisms. Despite the potential shown via numerical simulations, the underlying interfacial dynamics between a fluid column and a compliant metamaterial remain largely unexplored. Our preliminary studies have shown that conventional phononic band gaps are insufficient for effective control since the bulk of the absorbed energy remains confined to the fluid-structural interface, a direct consequence of localized modes associated with band gap truncation resonances. In this talk, we will discuss the mathematical intricacies of impedance boundary conditions (IBCs) at the interface of a one-dimensional fluid and an engineered structure, shedding light on both linear and non-linear subsurface behaviors. Through which, we will present a new pathway to effectively transmit a pressure wave from an oscillating fluid away from the interface, and into a structural medium.

Contributed Papers

9:25

2aSA2. Variational study of a model for near-perfect transmission through lossy media with acoustic sources. Nathan P. Geib (Appl. Res. Labs., Univ. of Texas at Austin, 1587 Beal Ave Apt 13, Ann Arbor, MI 48105, geib@umich.edu), Samuel P. Wallen, Michael R. Haberman, and Christina Naify (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Complementary acoustic metamaterials have been proposed as a means of compensating for the high impedance mismatches of aberrating layers that disrupt the acoustic field and hence distort acoustic images. Recently, a complementary acoustic metamaterial featuring active components was shown in principle to compensate for both the impedance mismatch and energy attenuation of lossy materials, but a physical realization of the concept has not yet been implemented. Here, we present results from a one-dimensional acoustic model showing how a plane wave incident on a lossy material can be augmented by point monopole and dipole sources to allow for near-perfect transmission, thus rendering the lossy medium acoustically transparent. We present general expressions for source magnitudes that are dimensionless with respect to frequency, material thickness, and the background medium. We explore the sensitivity of the performance to variations in each of the model parameters, considering both theoretical and practical limitations to the proposed method. We show that these findings are consistent with three-dimensional finite element simulations.

9:40

2aSA3. Study of defect mode characteristics in acoustic metamaterials.
Vinod Ramakrishnan (Mech. Sci. and Eng., Univ. of Illinois at Urbana-Champaign, 1324 MEL,105 S Mathews Ave., Urbana, IL 61801, vinodr@illinois.edu) and Kathryn Matlack (Mech. Sci. and Eng., Univ. of Illinois at Urbana-Champaign, Urbana, IL)

The development of novel material architectures incorporating unconventional geometric designs (e.g., microstructural instabilities, multi-length scale geometries), or adaptive constituent elements (e.g., piezo electrics) has created a powerful tool to design metamaterials for efficient dispersion manipulation and precise control of acoustic waves. Various aspects of the acoustic dispersion (e.g., wave speed, damping ratio, band gap position and widths) have been investigated with a primary focus on wave characteristics in the acoustic pass bands, promoting their utility in applications, e.g., wave guiding, energy harvesting. However, recent studies on topological waves, truncation and bandgap resonances have highlighted potential benefits of leveraging acoustic waves in the bandgap, e.g., signal propagation, passive flow control. Motivated by this, we explore the dispersion behavior of finite mass-spring-damper acoustic metamaterial models with material defects. The eigen analysis reveals a (defect) resonance in the acoustic band gap, producing a highly localized defect mode. The dependence of this resonance frequency, the degree of localization and phase response of the eigenmode, on the defect parameters are explored. Subsequently, the potential utility of these defect modes in flow control applications is also explored and presented.

9:55

2aSA4. Enhanced viscous dissipation of sound in phononic supercrystal.
Arkadii Krokhin (Dept. of Phys., Univ. of North Texas, 1155 Union Circle # 311427, Denton, TX 76203, arkady@unt.edu), Martin Ibarias, and José Sánchez-Dehesa (Wave Phenomena Group, Universitat Politecnica de Valencia, Valencia, Valencia, Spain)

Sound wave propagating in a homogeneous viscous medium decays due viscous losses with decay coefficient $\gamma_{;0}\sim\eta\omega^2$, where η is shear viscosity. Presence of a hard wall strongly increases viscous losses which now occurs mainly within a narrow boundary layer δ. Viscous losses in a 2D phononic crystal are analytically calculated in the low-frequency limit for arbitrary Bravais lattice. The decay coefficient is expressed through series over reciprocal lattice vectors. If the phononic crystal possesses less than 3-fold rotational symmetry it behaves as anisotropic viscous medium. Otherwise, the decay coefficient is isotropic. Depending on the crystal structure and the filling fraction of solid cylinders the decay coefficient $\gamma_{;\mathrm{ph}}$ may exceed $\gamma_{;0}$ by two-four orders of magnitude. The decay coefficient may be further enhanced in a supercrystal – a structure with doubly periodicity. The supercrystal is a 2D structure of two sets of aluminum cylinders in air. A periodic set of larger cylinders is imbedded in another set of smaller cylinders arranged in a lattice with smaller period. We present analytical, numerical, and experimental results for decay of sound in a hexagonal supercrystal of aluminum cylinders in air. [This work is supported by the NSF under EFRI Grant No. 1741677.]

10:10–10:25 Break

10:25

2aSA5. Localization-antilocalization of soundwaves in a disordered phononic crystal with mirror symmetry.
Michael McKinstry (Phys., Univ. of North Texas, 1155 Union Circle, Denton, TX 76203, michaelmckinstry@my.unt.edu), Dmitrii Shymkiv, and Arkadii Krokhin (Phys., Univ. of North Texas, Denton, TX)

Anderson localization in a disordered potential leads to exponential decay of an incoming wave. However, if a 1D disordered potential possesses mirror symmetry, $V(-x) = V(x)$, then the eigenstates are either even or odd functions of x. As such, a wave localized on one side of such a symmetric disordered potential should elicit a corresponding antilocalized peak at the symmetric position on the other side. The distribution of pressure in a symmetric disordered potential is similar to the wave function profile in a symmetrical double-well potential. This similarity opens a way to demonstrate quantum tunneling using acoustic waves. This effect has an important application to secure communications, as the transmitted signal is reduced by disorder to the level of noise, thus excluding the possibility of signal interception, then enhanced by the symmetric part of the potential at the receiver. This is a secure method of information transmission without the need for encryption and decryption. We developed a methodology for identifying the frequency spectrum consisting of narrow doublets that correspond to eigenstates of different parities. A 2 × 30 phononic crystal with orientational disorder was fabricated for experimental observation of antilocalization. [This work is supported by the NSF under EFRI Grant No. 1741677.]

10:40

2aSA6. Nonreciprocal energy transmission in short discrete systems with strong spatiotemporal modulations.
Jiuda Wu (Concordia Univ., Montreal, 1455 De Maisonneuve Blvd. W., Rm. EV-4.139, Montreal, QC H3G 1M8, Canada, w_jiuda@encs.concordia.ca) and Behrooz Yousefzadeh (Concordia Univ., Montreal, Montreal, QC, Canada)

Introducing spatiotemporally varying properties in a phononic lattice can enable nonreciprocal transmission of energy. The hallmark of this phenomenon is the unidirectional energy transmission in infinitely long systems. In short lattices, however, identifying nonreciprocity through differences in transmitted energies (norm bias) becomes challenging because the primary contributor to nonreciprocity is the transmitted phases. Although stronger modulation can achieve a higher norm bias, it can also result in parametric instabilities and trigger large-amplitude oscillations that lead to device failure. To better understand the tradeoff between stability and norm bias in strongly modulated systems, we investigate the parametric stability of a finite one-dimensional lattice with spatiotemporally modulated elasticity. We use Floquet theory to compute the stability charts for strongly modulated systems. Thus, we can identify operating conditions that allow for a relatively large norm bias while maintaining a stable, bounded response.

10:55

2aSA7. Inverse design of three-dimensional architected elastic metamaterials using graph neural networks.
Yu-tong Wang (Acoust., Penn State Univ., 201 Appl. Sci. Bldg., Graduate Program in Acoust., University Park, PA 16802, ybw5392@psu.edu), Mourad Oudich (Acoust., Penn State Univ., State College, PA), Marco Maurizi, Xiaoyu Zheng (Mater. Sci. and Eng., Univ. of California Berkeley, Berkeley, CA), and Yun Jing (Acoust., Penn State Univ., State College, PA)

Architected metamaterials exhibit novel mechanical properties shaped by the spatial arrangement of periodic structures, rather than their constituent materials. Truss lattices, a notable subtype, are recognized for their high strength-to-weight ratio; thus, they hold significant potential for applications in robotics, aerospace engineering, and other fields. Despite recent advances in deep learning (DL) have revolutionized traditional design, researchers have mainly focused on the inverse design of quasi-static properties, leaving a gap in addressing dynamic behavior or simultaneous considerations for both aspects. To overcome this gap, we develop a novel inverse design framework to generate truss metamaterials with tailored quasi-static (stress-strain curve) and dynamic (transmission curve) properties. Our data-driven framework, based on graph neural networks, integrates a forward model into an inverse model, trained using deep reinforcement learning. . To demonstrate the model performance, finite element method simulations, uniaxial compression tests, and vibration tests are conducted to verify the properties of the optimized structures. The successful realization of user-desired properties in both quasi-static and dynamic domain can potentially accelerate the inverse design of novel materials towards applications such as lightweight and high-strength vibration isolators.

11:10

2aSA8. Mitigating low-frequency broadband aircraft noise through a structured assembly of acoustic material and devices. Tenon Charly Kone (Flight Res. Lab., National Res. Council Canada, 1200 Montreal Rd., Ottawa, ON, Ottawa, ON K1A 0R6, Canada, tenoncharly.kone@nrc-cnrc.gc.ca), Sebastian Ghinet (Flight Res. Lab., National Res. Council Canada, Ottawa, ON, Canada), Raymond Panneton (Ctr. de Recherche Acoustique-Signal-Humain, Université de Sherbrooke, Sherbrooke, QC, Canada), and Anant GREWAL (Flight Res. Lab., National Res. Council Canada, Ottawa, ON, Canada)

Addressing the challenge of attenuating low-frequency broadband noise emerges as a critical concern within the fields of aeronautics, ground transportation, and construction industries, requires innovative solutions for enhanced acoustic control. Over the last few decades, the literature has seen an increase in low-frequency noise control solutions centered around acoustic metamaterial designs. These proposed technologies exhibit promising acoustic performance, especially proving superior to conventional sound insulation materials in constrained spaces, such as in aerospace applications. Despite the efficacy of typical metamaterials in attenuating tonal noise through narrow resonant frequency maxima, practical applications reveal some challenges, as even slight variations in tonal noise frequencies can compromise the overall effectiveness of such solutions. In response to this, the present paper introduces a novel thin acoustic metamaterial design aimed at improving broadband noise attenuation at low frequencies. This design uses carefully arranged structured metamaterials within a fiberglass layer to create optimal resonance frequency bands for maximum low-frequency noise attenuation. Performance assessment in the low-frequency domain employed COMSOL Multiphysics finite element methods, predicting sound absorption coefficient and transmission loss. Results confirm the effectiveness of the proposed metamaterial design, showcasing broad noise attenuation at low frequencies.

11:25

2aSA9. Active acoustic metamaterials with independently programmable bulk modulus and full mass density tensor. Dylan Kovacevich (Mech. Eng., Univ. of Michigan, 2350 Hayward St., Ann Arbor, MI 48109, dkovac@umich.edu) and Bogdan-Ioan Popa (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

Active acoustic metamaterials consist of unit cells with sensor-driver pairs that produce a coherent response to incident waves. The effective acoustic properties of the metamaterials depend on the gains programmed between the sensing and driving components. The strength of the monopole response to the local pressure determines the effective bulk modulus and the dipole response to the local particle velocity determines the effective mass density. The unit cells are controlled individually, so in theory the metamaterials can be scaled to an arbitrary size and geometry, but prior realizations were limited to only a few cells with 1D operation. Here, we present an active acoustic metamaterial of nine cells in 2D with programmable bulk modulus and mass density tensor. We demonstrate the ability to independently program the property components (bulk modulus and four mass density elements) for any desired set within stable limits, including previously unattainable acoustic properties necessary for the fabrication of transformation acoustics devices. We also demonstrate complex effective properties, specifically a lossy layer with impedance matched to the background, such that incident waves are absorbed with no reflection. The effective properties of the active metamaterial are validated by comparing the experimental total and scattered fields to ideal simulation results.

Session 2aSC

Speech Communication: VowelFest: Honoring the Past and Celebrating the Present I

Ewa Jacewicz, Chair
Speech and Hearing Science, The Ohio State University, 1070 Carmack Road, 110 Pressey Hall, Columbus, OH 43210

Chair's Introduction—8:00

Invited Papers

8:05

2aSC1. A vowel research retrospective. Robert A. Fox (Speech and Hearing Sci., The Ohio State Univ., 1070 Carmack Rd., 110 Pressey Hall, Columbus, OH 43210-1002, fox.2@osu.edu)

One of the earliest attempts at describing/characterizing vowel quality was Daniel Jones' Cardinal Vowel system (1918) representing the total range of vowel quality in languages. It was based on auditory quality (evaluated by a trained phonetician). Much later, Ladefoged (1967) demonstrated that despite auditory equivalence, vowels were acoustically different—a function of physiological and production among speakers. The acoustic nature of vowel quality was elaborated through the introduction of easy spectral analysis (using the sonograph) exemplified by the seminal work by Peterson and Barney (1952). In the past five decades we have seen how vowels differ as a function of speaker sex, speaker size, age, speech disorder and dialect group and development of vowel spaces in second-language acquisition (much of the research through contributions of speakers in this session). In concert with these acoustic studies, we have better understanding of the nature of the perception of vowels through multidimensional scaling studies, adaptation/perceptual magnet/categorical perception studies, cross-linguistic studies, and developmental studies. Most recently, we have seen work done with regard to brain mapping and neural models of vowel perception. This talk will provide a review of past vowel research (voluminous as it is) and possible future trends.

8:25

2aSC2. Modeling vowel-inherent spectral change in varying consonant context. Michael Kiefte (Commun. Sci. and Disord., Dalhousie Univ., 1256 Barrington St., Halifax, NS B3J 1Y6, Canada, mkiefte@dal.ca)

A wide variety of experimental evidence has shown that vowel inherent spectral change (VISC) is important in vowel identification. This evidence is drawn from production data, statistical pattern recognition, and perceptual experiments with both synthetic or manipulated naturally produced speech. Experimental studies often consider vowels in a constant consonant context which makes it difficult to factor out context effects from the VISC itself. In order to examine the magnitude of these consonant effects, we developed a statistical procedure inspired by Broad and Clermont [(2014). J. Phon. 47, 47–80] in which vowel formant frequencies are approximated by a linear combination of vowel and consonant influences which vary as a function of time. Vowels extracted from a database of both spontaneous and read speech were analyzed to produce context-normalized vowel-formant tracks. Results show that vowel formant frequencies vary systematically across their duration in both spontaneous and read speech and that all consonants in both onset and coda position show significant effects on vowel production across their entire duration. Although these formant patterns are seemingly complex, perceptual evidence suggests that listeners may only attend to onsets and offsets and that deviations from a straight-line interpolation between onset and offset must be relatively large for listeners to discriminate them.

8:45

2aSC3. American English monophthong tenseness. Richard McGowan (CReSS Books, 1 Seaborn Pl., Lexington, MA 02420, mcgowan.richard.s@gmail.com)

In General American English, the point vowels of the F2 – F1 verus F1 quadrilateral, [i], [u], and [ɑ] are tense. More generally, Lindau noted the connection between acoustic peripherality and vowel tenseness [Lindau, M. (1975). "Vowel Features." *Working Papers, Phonetics Laboratory, Lund University*, **11**, p. 1]. In the case of [ae], it is the tensing of this point vowel that initiated the Northern Cities Chain Shift [Labov, W. (1994). *Principles of Linguistic Sound Change: Internal Factors.* Blackwell Publishers, Malden, MA]. The theses of this talk are firstly that a high degree of acoustic sensitivity to area function change is the reason that peripherality is highly correlated with tenseness. However the reason for the acoustic sensitivity of [ae] is different than it is for the other three point vowels. Second, a more fundamental characterization of tenseness is in its articulatory kinematics during production. The reason for this has to do with the persistence of tenseness with diachronic sound change away from the periphery. Perhaps, though, tense/lax changes appear to be more likely on the periphery than elsewhere.

9:05

2aSC4. Lifespan anatomic and vowel acoustic studies confirm prepubertal sexual dimorphism of the human vocal tract and absence of lengthening in aging. Houri K. Vorperian (Waisman Ctr., Univ. of Wisconsin-Madison, University of Wisconsin-Madison, Waisman Ctr., 1500 Highland Ave., Vocal Tract Development Lab # 427, Madison, WI 53705, vorperian@waisman.wisc.edu) and Raymond D. Kent (Dept of Commun. Sci. & Disord., Univ. of Wisconsin-Madison, Madison, WI)

To study anatomic-acoustic relations across the lifespan, anatomic studies of the developing vocal tract (VT) using medical imaging studies were related to vowel acoustic space as synthesized from published data. However, the aggregate acoustic data highlighted the need to acquire sex-specific vowel acoustic data across the lifespan using controlled methodology for data collection and analysis; also, to acquire measurements of the higher formants. Lifespan anatomic and acoustic (4-to-92 years) data helped resolve two questions. First, anatomic studies revealed significant prepubertal sex differences in the oral region of the VT (3-to-7 years) that are masked by growth rate differences between males and females. Three-dimensional anatomic findings also revealed prepubertal sexual dimorphism in select regions of the mandible and pharynx. Second, anatomic findings confirmed that VT length does not increase with aging. Acoustic studies similarly confirmed sex differences in fundamental frequency emerging at age 7; and the presence of significant prepubertal sexual dimorphism of the higher formant frequencies (F3-F4) at the earliest age studied (4 years). Findings for adults showed significant age-related decreases in fundamental frequency in women only, and no changes in formant frequencies in either sex across several decades. The results underscore the importance of sex-specific and non-uniform growth of the VT structures. [NIDCD funding support R01DC006282.]

9:25

2aSC5. Vowel perception research from formant thresholds to sentence intelligibility. Diane Kewley-Port (Speech and Hearing Sci., Indiana Univ., 3803 E St Remy Dr., Bloomington, IN 47401, kewley@indiana.edu)

Historically, research on the contribution of vowels to speech understanding lagged that of consonants. Speech synthesis techniques were then developed that established that the primary acoustic features of vowels are formant frequency, fundamental frequency, speech dynamics and naturalness. For speech perception research, precise control of the features in vowel stimuli was required. Starting in the 1980s, the Klatt formant synthesizer was the first important tool, and by 2000 Kawahara's STRAIGHT synthesizer generated nearly natural speech. As a baseline for vowel perception, my research sought to determine psychophysical thresholds for F1 and F2 under ideal conditions. This talk reports contributions from my vowel perception studies on three questions. First, how do formant thresholds change with speech context (isolated vowels up to sentences), across age and with hearing impairment? Second, do vowels or consonants contribute more to intelligibility in noise-interrupted sentences? Third, given formant dynamics, how is intelligibility affected as consonant-vowel boundary conditions are manipulated across age and hearing impairment? Significant results include: (1) Vowels carry more information about sentence intelligibility than consonants for both young and older listeners; (2) Even though older listeners' performance is reduced compared to young, hearing impairment has a greater negative impact than age-related cognitive decline.

9:45–10:00 Break

10:00

2aSC6. Vowels in clear and conversational speech: Intelligibility for listeners with hearing loss. Sarah H. Ferguson (Commun. Sci. and Disord., Univ. of Utah, 390 South, 1530 East, Rm. 1201, Salt Lake City, UT 84112, sarah.ferguson@hsc.utah.edu)

While my research during the past two decades has explored numerous perceptual properties of clear versus conversational speech, it all began with vowel intelligibility. In this presentation, I review my research on vowel intelligibility in listeners with hearing loss and whether and how it changes when talkers adopt a clear speaking style. I also discuss acoustic properties that may explain these changes and the potential interaction between clear speech acoustic properties and listener hearing status.

10:20

2aSC7. What babies bring to our understanding of vowel perception. Linda Polka (School of Commun. Sci. & Disord., McGill Univ., 2001 McGill College Ave., 8th Fl., SCSD, McGill University, Montreal, QC H3A 1 G1, Canada, linda.polka@mcgill.ca)

Vowels are produced and perceived very early in infancy and occupy a central role in speech communication across the lifespan. Many phenomena first uncovered in vowel research with adults were later explored with infants. This produced important findings that inform our understanding of vowel perception and production development. In this talk, I will highlight several perception findings that were initially discovered in infants which now also direct fruitful lines of research with adults. I will focus on two perceptual biases—the focal vowel bias and the infant talker bias—each involving information conveyed by vowels. The focal vowel bias identifies a universal vowel perception bias that is germane across the lifespan. Elaborating the mechanism(s) behind this bias can lead us to a more principled understanding of basic vowel perception processes. The infant talker bias – first identified in infants and now also in adults – reveals a robust bias favoring infant conspecific vocalizations. This bias appears to impact infant development directly and also indirectly via its' positive effect on parenting behaviors. Going forward, the interplay of research across age groups will continue to bring us a deeper understanding and appreciation of the ubiquitous role of vowels in human cognition.

10:40

2aSC8. Perception of age, gender, and talker height from children's vowels. Peter F. Assmann (Psych., Univ. of Texas at Dallas, 800 W Campbell Rd., Richardson, TX 75080, assmann@utdallas.edu), Abbey L. Thomas (Brain and Behavioral Sci., The Univ. of Texas at Dallas, Richardson, TX), and Santiago Barreda (Linguist, UC Davis, Davis, CA)

In his 1989 review of vowel perception [J. Acoust. Soc. Am. 85, 2088–2113], Terry Nearey observed that the acoustic properties of vowels vary substantially across talkers due to differences in talker size. Taller speakers tend to have lower formant frequencies and lower fundamental frequencies than shorter speakers, and listeners appear to take these relationships into account in vowel identification. In this talk, we follow up some of the questions raised in that paper, reviewing recent findings on the perception of age, gender, and

talker height in children's voices. Children exhibit substantial age-related changes in fundamental and formant frequencies due to growth of the larynx and vocal tract, presenting an interesting opportunity to explore the perceptual consequences of these changes. Listeners' perceptual judgments of age and gender reveal complex interdependencies, consistent with the idea that these indexical attributes are jointly estimated, in a manner that reflects their shared dependence on acoustic parameters related to perceived talker height.

11:00

2aSC9. My great big problem with vowels. James E. Flege (Speech & Hearing Sci., Univ. of Alabama at Birmingham, Via del Moro 12, Tuscania, Viterbo 01017, Italy, jimflege@gmail.com)

How vowels are produced in a second language (L2) is determined by several factors. One is cross-language mapping: how/if vowels in the L2 phonetic system are categorized in terms of native language (L1) vowels and the extent to which they are judged to resemble the L1 vowels perceptually. Another is input: the quantity and quality of the L2 vowels to which learners have been exposed. L1-L2 mapping patterns are sometimes reported in studies of L2 vowel learning, but they may differ across individuals and change as learners gain L2 experience. Coarse estimates of quantity of L2 input are usually reported but estimates of exposure to foreign-accented L2 input are virtually non-existent. Even if we had adequate measures of L2 input, other serious problems would remain. Once the L1 phonetic system is established, language users may attend less to the surface phonetic properties of vowels they hear when conversing in the L1 and also in the L2. This talk will focus on problems inherent in (a) understanding how the phonetic input received in an L2 is used, and (b) gauging degree of L2 learning success.

11:20

2aSC10. Unexpected findings from L2 speech research can help us understand how humans process vowels. Ocke-Schwen Bohn (Aarhus University, Sejts Alle 20a, Risskov DK-8240, Denmark, engosb@hum.au.dk)

Research on nonnative speakers' vowel production and perception has provided us with a number of surprising insights on how learners cope with nonnative vowels. Three of these initially unexpected, yet later solidly replicated, findings will be presented: (1) A decline in intelligibility of nonnative vowels as general proficiency improves (when an increase in intelligibility will be expected), (2) the use of acoustic cues in L2 vowel perception that cannot be attributed to transfer from the native language, and that are nonfunctional in the L2, and (3) the maintenance from infant speech perception in adult cross-language vowel perception of a bias favoring peripheral vowels. This presentation will discuss implications of these findings for our understanding of how vowel categories coexist in the minds of multilinguals, and of universally (native-language independent) preferred ways of vowel perception.

Session 2aSP

Signal Processing in Acoustics, Biomedical Acoustics, Physical Acoustics, Computational Acoustics, and Underwater Acoustics: Data Augmentation in Signal Processing: Advancing Performance through Artificial Data Generation

Yongsung Park, Cochair
University of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0238

David R. Dowling, Cochair
Mechanical Engineering, University of Michigan, Dept. of Mech. Eng., Univ. of Mich., 1231 Beal Avenue, Ann Arbor, MI 48109

Chair's Introduction—8:00

Invited Papers

8:05

2aSP1. The origins of frequency-difference and frequency-sum beamforming. Shima Abadi (Univ. of Washington, 185 Stevens Way, Paul Allen Ctr. – Rm. AE100R, Seattle, WA 98195, abadi@uw.edu), David R. Dowling (Dept. of Naval Architecture and Marine Eng., Univ. of Michigan, Ann Arbor, MI), Heechun Song (SIO, UCSD, La Jolla, CA), and Kevin J. Haworth (Dept. of Internal Medicine, Univ. of Cincinnati, Cincinnati, OH)

Beamforming the signals recorded by an array enables the determination of sound source location(s) or the arrival directions of ray paths between a sound source and the receiving array. Frequency-difference and frequency-sum beamforming are beamforming techniques that provide out-of-band information from in-band signal frequencies. Interestingly, the out-of-band frequencies can be chosen by the user, within limits set by the signal recordings, to achieve desired properties of the beamformed output, such as: increased resolution, reduced sidelobes, or greater robustness to random scattering. Both techniques are general and are not limited to any particular acoustic environment, frequency range, or array geometry. Frequency-sum beamforming, generates higher-frequency information from lower frequency signal components, enhancing beamforming results in scenarios with random scattering between the source and the receivers. However, it is limited by artifacts arising from cross-terms when multiple source signals are present in the same bandwidth. Conversely, frequency-difference beamforming manufactures lower-frequency information from higher frequency signal components, effectively mitigating the impact of spatial aliasing in situations where the receiving array is sparse. This presentation delves into the origins of frequency-difference and frequency-sum beamforming, presents the fundamental mathematics underlying their algorithms, and showcases their performance via simulations and experimental results. [Work supported by ONR.]

8:25

2aSP2. Beamforming applications of the frequency-difference acoustic autoproduct. Alexander S. Douglass (Oceanogr., Univ. of Washington, 2010 AL, 1231 Beal, Ann Arbor, MI 48109, asdougl@umich.edu), Heechun Song (SIO, UCSD, La Jolla, CA), and David R. Dowling (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

Frequency-difference beamforming (Abadi *et al.*, 2012, JASA, 132, 3018–3029) is an array signal processing technique that overcomes the limitations of the spatial Nyquist criterion by utilizing the acoustic autoproduct to shift the processing to below-band frequencies. This is accomplished using a quadratic product of complex signal amplitudes at different frequencies, resulting in wave propagation information at the out-of-band difference-frequency. The resulting field is capable of mitigating many of the in-band challenges often associated with high frequency acoustic signal processing, including sparse receiver arrays and features of the physical environment that are on a scale significant for the in-band wavelength. This presentation uses both laboratory and ocean-based experiments to demonstrate capabilities of the method. The mitigation of sparse array aliasing effects on beamforming (caused by elements that are spaced by many wavelengths) in both a laboratory water tank and an ocean environment utilizing data collected during the KAM11 experiment are considered. Additionally, the method is used to localize a high frequency source in the presence of strong, random scatterers in a water tank experiment. [Work sponsored by NAVSEA and ONR.]

8:45

2aSP3. Augmenting sparse arrays in ocean acoustics: A Gaussian Process approach. Zoi-Heleni Michalopoulou (Dept. of Mathematical Sci., New Jersey Inst. of Technol., 323 M. L. King Boulevard, Newark, NJ 07102, michalop@njit.edu) and Peter Gerstoft (Univ. of California, San Diego, La Jolla, CA)

Source localization and geoacoustic inversion performance depends on the location of receiving phones, their spacing, their number, and the array aperture. Performance suffers when practical design issues limit the capabilities of the array. In this work, sparse arrays are augmented leading to the generation of virtual arrays for better sampling of the ocean and, thus, improved estimation performance. To that effect, Gaussian Processes are employed, which are shown to create high-fidelity field "measurements" at virtual, densely spaced hydrophones. Kernel functions are key building blocks in the implementation of Gaussian Processes, as they quantify the field coherence at neighboring spatial points. Functions of interest are the squared exponential, Matern, and plane wave kernels. We validate our method with application to synthetic data as well as data collected during the Seabed Characterization Experiment conducted in 2022. [Work supported by ONR.]

9:05

2aSP4. A modular neural network to quickly approximate the modal dispersion in coastal waters. Arthur Varon (GIPSA-Lab, 11 Rue des Mathématiques, Saint-Martin-d'Hères, Isère 38400, France, arthur.varon@gipsa-lab.grenoble-inp.fr), Jérome Mars (GIPSA-Lab, Saint-Martin-d'Hères, France), and Julien Bonnel (Woods Hole Oceanographic Inst., Woods Hole, MA)

Low-frequency acoustic propagation modeling in coastal waters usually relies on numerical models based on modal theory such as Kraken and Orca. These models compute the modal parameters (e.g., modal wavenumbers and depth functions) that can be used in the calculation of the acoustic field. Their repeated use in broadband applications, or for inversion purposes, comes with a notable computational cost. To mitigate this, a modular neural network (NN) was trained to approximate modal parameters for varying modes and frequencies, across diverse environments, with variable water sound speed profile and variable seabed geoacoustic parameters. The training dataset is generated using Kraken and the NN is evaluated on many environments not seen during training. Once trained, the NN can make broadband predictions without prior knowledge on the number of modes, even when the number of modes changes over the frequency-band of interest. This approach reduces computation time compared to the original forward propagation model, while maintaining high precision. The effectiveness of our method is demonstrated through transmission loss calculations and a simulated geoacoustic inversion scenario.

Contributed Papers

9:25

2aSP5. Frequency difference-wavenumber analysis for direction-of-arrival estimation in sparse vertical 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 (Korea Maritime and Ocean Univ., La Jolla, CA), and J. S. Kim (Korea Maritime and Ocean Univ., Busan, Korea (the Republic of))

Frequency–wavenumber (*f–k*) analysis can determine the direction-of-arrival (DOA) for broadband signals captured by a vertical array [M. J. Hinich, J. Acoust. Soc. Am., **69**, 732–737 (1981)]. The sparse vertical array generates numerous sidelobes in the *f–k* domain due to aliasing errors resulting from the spatial sampling. This presentation introduces the frequency difference-wavenumber ($\triangle f$–k) analysis, expanding the application of *f–k* analysis to the sparse vertical array by adopting the concept of frequency difference. The relationship between the frequency difference beamforming and the $\triangle f$–k analysis is also discussed. Experimental results verify the effectiveness of the proposed $\triangle f$–k analysis in estimating the DOA of snapping shrimp clicks (11–24 kHz) recorded using a sparse vertical array in a shallow water experiment. During the experiment, a vertical array with an element spacing of 3.75 m (i.e., design frequency = 200 Hz) is utilized, which is extremely sparse because it corresponds to 27.5 wavelengths at the lowest frequency.

9:40–9:55 Break

9:55

2aSP6. High-resolution matched autoproduct processing using sparse Bayesian learning for multiple source localization. Ze Yuan (State Key Lab. of Acoust., Inst. of Acoust., Chinese Acad. of Sci., Inst. of Acoust., Chinese Acad. of Sci., No. 21 North 4th Ring Rd., Haidian District, Beijing, China, Beijing 100190, China, yuanze@mail.ioa.ac.cn), Haiqiang Niu (State Key Lab. of Acoust., Inst. of Acoust., Chinese Acad. of Sci., Beijing, China), Zhenglin Li (School of Ocean Eng. and Technol., Sun Yat-sen Univ., Zhuhai, China), and Wenyu Luo (State Key Lab. of Acoust., Inst. of Acoust., Chinese Acad. of Sci., Beijing, China)

Matched autoproduct processing (MAP) is a nonlinear array-signal-processing technique employed for source location estimation within the framework of matched field processing (MFP). Operating by matching frequency-difference autoproducts instead of the pressure field, MAP induces a downshift in frequency. While offering reduced sensitivity to environmental mismatch, MAP presents drawbacks such as compromised spatial resolution, broadened mainlobe, and a diminished peak-to-sidelobe ratio within the ambiguity surface. These characteristics make the application of MAP challenging in scenarios involving multiple sources, especially when dealing with close proximity or a weak source alongside a stronger one. To address these limitations, MAP estimation is formulated as a sparse signal reconstruction problem, solved through sparse Bayesian learning (SBL). In the context of preserving frequency-difference robustness, the proposed method demonstrates a narrower mainlobe and diminished sidelobe levels compared to the original MAP. This improvement extends the applicability of MAP to multi-source localization scenarios. Finally, we validate the superior performance of the proposed method through both simulations and experimental data.

10:10

2aSP7. Artificial data augmentation using braided human features for underwater acoustics. Ananya Sen Gupta (Dept. of Elec. and Comput. Eng., Univ. of Iowa, 103 S Capitol St., Iowa City, IA 52242, ananya-sengupta@uiowa.edu), Andrew J. Christensen (Elec. and Comput. Eng., Univ. of Iowa, Iowa City, IA), Anjali Mathews (Dept. of Elec. and Comput. Eng., Univ. of Iowa, Iowa City, IA), Subhajit Das (Subhangik Dance Co., Bandel, West Bengal, India), and Ivars P. Kirsteins (NUWCDIVNPT, Newport, RI)

Underwater acoustics datasets provide rich environmental information that can potentially be mined using popular machine learning architectures. However, such knowledge discovery is typically limited by the high feature uncertainty, lack of robust ground truths, limited availability of public-domain data, and difficulty of reproducing experimental conditions in field experiments. This poses a data science challenge to study persistent target features, as they can only be learnt and classified robustly if large scale robust training and testing datasets are available. This "nano-data" problem can be potentially solved by augmenting the training and testing data repositories with a two-pronged approach: (i) physics-driven simulations of desired features, which offer interpretable ground truths but typically cannot emulate practical open-sea experiments; (ii) geometric-proxy augmentation using creatively constructed non-domain datasets that emulate the desired feature geometries and environmental effects. For example, a skilled dancer can emulate specific features geometries through geometric contortions of the human body. Such choreographed dance movements can be reproduced reliably against different types of stage environments, structured and unstructured, to emulate the oceanic environment under different conditions. We will present preliminary results in data augmentation from both approaches and discuss the trade-offs between physics-oriented simulations and geometry-driven proxy augmentation. [ONR grant N000142312503.]

10:25

2aSP8. Increasing the accuracy of ISO 354 and ASTM C423 through modal identification. Dario D'Orazio (Univ. of Bologna, Viale Risorgimento, 2, Bologna 40126, Italy, dario.dorazio@unibo.it), Andrea Zaccarini (Univ. of Bologna, Bologna, Italy), and Fiorella Falciano (Univ. of Bologna, Bologna, Italy)

Measuring the acoustic properties of materials is a highly challenging task. Previous studies have highlighted errors in the absorption coefficient above 200 Hz, leading to absorption coefficient values higher than one. These errors are often attributed to differences in modal decays with and without the specimen. This effect is amplified in the case of highly reactive materials, such as innovative materials, that significantly impact the acoustic field and, consequently, the modal behaviour of the room. This work aims to reduce the difference between measured reverberation time values, with and without the specimen, through Data Augmentation. Modal identification was performed on measured impulse responses of porous specimens, where the alpha is known, allowing the determination of modal decay times and reverberation times within the frequency range of 70–180 Hz. A statistical analysis of natural modes and numeric solutions enables the identification of outliers. In this way, even in a non-well-optimized reverberation room, it is possible to enhance the accuracy of acoustic absorption measurements.

Invited Paper

10:40

2aSP9. Frequency-difference beamforming and sparse processing. Yongsung Park (Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0238, yongsungpark@ucsd.edu) and Peter Gerstoft (Univ. of California, San Diego, La Jolla, CA)

Frequency-difference processing enables the estimation of the direction of arrival (DOA) for sources beyond the spatial aliasing frequency. The beamforming method takes advantage of the frequency difference between multiple frequencies, enabling processing at a lower frequency than the aliasing frequency. During this process, the number of DOAs we need to estimate becomes squared, and we require a high-resolution DOA estimator. Sparse processing has demonstrated high-resolution capabilities in DOA estimation, resulting in beamforming spectra with sharp peaks and enhancing resolution for accuracy. This presentation proposes the utilization of both frequency-difference processing and sparse processing. The suggested method is an aliasing-free beamformer for high-frequency sources using frequency-difference processing and achieves high-resolution capability using sparse processing.

Session 2pAAa

Architectural Acoustics and ASA Committee on Standards: Show Your Data: Architectural Acoustics Metrics II

Ana M. Jaramillo, Cochair
Ahnert Feistel Media Group, 8717 Humboldt ave N, Brooklyn Park, MN 55444

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Olson Sound Design LLC, 8717 Humboldt Avenue, N, Brooklyn Park, MN 55444-1320

Contributed Papers

1:00

2pAAa1. How can desks and ceilings influence room acoustic criteria in classrooms? Findings from measurements and wave-based simulations. Giulia Fratoni (Univ. of Bologna, Viale del Risorgimento 2, Bologna, BO 40136, Italy, giulia.fratoni2@unibo.it) and Dario D'Orazio (Univ. of Bologna, Bologna, Italy)

Sound diffusion is essential in achieving the required speech intelligibility criteria in classrooms ($T_{;30}$, $C_{;50}$, STI). Predictive formulas employed and recommended by international standards generally rely on the ideal diffuse-field theory, notwithstanding its applicability limitation in most feasible real-world scenarios. Yet, along with the sound-absorbing features of the surfaces, it is crucial to quantify any potential diffraction effect caused by furnishing elements and ceiling treatments. The present work uses acoustic measurements and wave-based numerical models to explore the impact of desk layouts and materials in false ceilings on sound diffusion. The single desk arrangement increases the sound diffusion, here quantified in terms of measured $T_{;30}$ standard deviation, more than other furniture layouts, e.g., desks arranged in circles. Moreover, preliminary results from acoustic measurements and finite-element analysis show that inhomogeneous treatments of the suspended ceiling – porous and perforated modules – increase sound diffusion through material discontinuity even at low frequencies. The match between predictive formulas, experimental results, and simulations confirms the reliability and accuracy of the acoustic design process when involving diffraction effects.

1:15

2pAAa2. Acoustical enhancement of a multi-purpose auditorium in an academic setting. Anthony Steven Garcia Mora (Escuela Superior Politécnica del Litoral, Guayaquil, Ecuador), Luis Bryan Vanegas Cruz (Escuela Superior Politécnica del Litoral, Guayaquil, Ecuador), Galo Durazno (Escuela Superior Politécnica del Litoral, Guayaquil, Ecuador), Carlos Yoong (Wood PLC, 2020 Winston Park Dr #700, Oakville, ON L6H6X7, Canada, carlos.yoong@woodplc.com), and Pedro Segovia Gonzalez (Universidad de las Artes, Guayaquil, Ecuador)

The Coastal Polytechnic School in Guayaquil, Ecuador is the leading engineering school in the country. Their Faculty of Mechanical Engineering currently have an auditorium that it is being used for different type of events such as conferences and artistic presentations. The design of the auditorium never considered acoustics, and the space currently presents issues with high reverberation time, deficient speech intelligibility and poor sound system. As part of a capstone project, a group of students from the Faculty conducted a detailed assessment of the space and develop solutions to enhance its acoustical properties for the required space functionality. Onsite measurements were taken and incorporated into a specialized software to calibrate the properties of the current space. Various solution scenarios were then evaluated computationally to find a solution suitable for a multi-purpose space.

1:30

2pAAa3. The investigation of sound and position in rehearsal rooms, the data is telling. Carolyn Dzul (Recording Arts & Sci., Peabody Inst. of The Johns Hopkins Univ., 606 Saint Paul St., P.O. 435, Baltimore, MD 21202, carolyn.dzul@gmail.com) and Ian B. Hoffman (Recording Arts & Sci., Peabody Inst. of The Johns Hopkins Univ., Baltimore, MD)

This paper includes the investigation of music rehearsal environments. Mainly focused at the high school level, it measures criteria that impact the health and performance of young musicians. This study goes beyond the ISO 3382: Measurements of room acoustic parameters and the Norwegian standard NS 8178 considerations. It assesses sound exposure and risks as they relate to room acoustics goals and data. The measurements of at least five rehearsal rooms in the Baltimore/DC area are conducted with orchestra and wind ensembles. A questionnaire to obtain observation data from music directors was considered in the analysis. The collected measured data focus on the actual sound exposure levels experienced by the music director and students. A consistent, repeated, triangulated setup of measurement locations allowed for both exposure and cross-room communication evaluations. The results revealed potential inconsistencies between healthy sound exposure and the goals/targets noted in the standards mentioned above.

1:45

2pAAa4. A survey of stage acoustic conditions for choral performances. Francesco Martellotta (Dept. Architecture, Construction and Design, Politecnico di Bari, Via Orabona 4, Bari 70125, Italy, francesco.martellotta@poliba.it), Chiara Rubino (Dept. Architecture, Construction and Design, Politecnico di Bari, Bari, Italy), and Stefania Liuzzi (Dept. Architecture, Construction and Design, Politecnico di Bari, Bari, Italy)

Choral performances, due to a very large number of non-professional groups often operating in small cities, frequently take place in non-dedicated spaces like churches, open atria, and other spaces with unusual acoustic conditions. This results in performances taking place in spaces without a real "stage" and consequently, in acoustic conditions that, as reported by several interviews, lack support and communication among singers. In order to better understand the effective acoustic conditions experienced in real spaces that frequently host choral performances, an on-site survey was carried out in six venues including one semi-open atrium and five churches of different styles and dimensions. Measurements were carried out according to ISO 3382-1 requirements, also taking into account the most recent proposals in terms of stage support to account for actual source directivity and presence of reflecting elements close to the source. Results showed that smaller spaces where vertical surfaces where more likely to be closer to the performers behaved better than large spaces where side volumes (like transepts or high domes) mostly withdrew acoustic energy yielding STearly values much lower than −12 dB.

2:00

2pAAa5. Acoustical investigations of multipurpose spaces at community facilities. Mandy Chan (2000 Argentia Rd., Plaza 1, Ste. 203, Mississauga, ON L5N 1P7, Canada, machan@hgcengineering.com), Michael Kundakcioglu (Mississauga, ON, Canada), and Alex Lorimer (Mississauga, ON, Canada)

Multipurpose spaces in community and recreational facilities hold significant value, catering to a variety of uses and accommodating users of diverse age groups. This article discusses investigations conducted at nine municipally owned facilities including community centres, arenas, and a senior centre related to reportedly poor acoustic conditions in multipurpose spaces. The acoustical measurements of the as-found conditions, relevant criteria, and analysis are outlined. The sound isolation, acoustical treatment for interior reverberation control, and background noise from HVAC are discussed as the contributing factors for the subjective acoustic concerns.

2:15

2pAAa6. Understanding diffusion versus scattering devices and how they work. Richard L. Lenz (RealAcoustix LLC, 2361 B Ave., Ogden, UT 84401, RL@RealAcoustix.com)

Nearly 50 years ago, Dr. Manfred Schroeder defined the word "Diffuser" as both a process and a product. Since then, many product manufacturers, acousticians and academics have come to use the term for many different products and designs. A careful reading of Dr. Schroeder's 1975 and 1978 papers reveals a lot of significant information as to why certain products are diffusers and why other products do not meet his criteria for diffusion. This paper will show, with the aid of Schroeder's interpretations and modern particle simulations, exactly what he had in mind when he, and others he was working with, created what ultimately became the Quadratic Residue Diffuser or QRD. Additionally, it will show his journey from primitive-root theory to his final quadratic designs. A discussion of the term "Scattering" will also be included to define how Schroeder uses the word and why we need to be careful as to how it is used as well. Test data showing phase models will be used to show the importance of phase manipulation in describing diffusion.

2:30

2pAAa7. Test method for diaphragmatic bass traps in a revberation chamber. Richard L. Lenz (RealAcoustix LLC, 2361 B Ave., Ogden, UT 84401, RL@RealAcoustix.com)

For decades the limitations of ASTM C423 and ISO 354 have been confined to frequencies above 100Hz. While many attempts have been made to provide low frequency test data using these standards, they all go beyond the actual standards. A significant problem with testing low frequencies in reverberation chambers has to do with using the diffuse field, or standard test area, of the lab. Low frequencies do not propagate in these areas to the level of the standard test range. Additionally, the devices, specifically diaphragmatic low frequency absorbers, are not used in the diffuse field of rooms. They are generally designed to be placed in, or near, corners of rooms, or modal areas, where low frequencies do propagate. This test method will show experiments and the process of determining how to test diaphragmatic low frequency devices using the modal zone of the chamber. These tests are intended to be presented to ASTM in order to develop an addendum to C423 for testing low frequencies and the lab limitations based on size and other factors.

2:45–3:00 Break

3:00

2pAAa8. An analysis and retrofit of the acoustics at Image Creators Health and Beauty Salon. Donna A. Ellis (Architectural Acoust., Lines by Nature LLC, 415 Riggs Ave., Severna Park Md. 21146, Severna Park, MD 21146, dellisdona@gmail.com)

This paper discusses the analysis and retrofit of the acoustics in a high-volume beauty salon in Severna Park, Md. The major issues in what was intended to be a serene environment are reverberation times of 1–1.68 s in the mid-to upper frequency range and Background noise levels ranging from 73.4 to 64.6 dB in the mid to low frequency range. Employee and customer complaints include ear pain and tinnitus due to prolonged exposure to high background levels, heightened stress, vocal strain, headaches and poor speech intelligibility. Existing analysis and the acoustical retrofit resolution will be demonstrated.

3:15

2pAAa9. A real-world Lexicon 960L reverberation chamber: Simulating a hardware reverberation unit in virtual acoustics. Aybar Aydin (Sound Recording Area, McGill Univ., McGill University, Dept. of Music Res., Montreal, QC H3A 1E3, Canada, aybar.aydin@mail.mcgill.ca), Vlad Baran, Kathleen Ying-Ying Zhang (Sound Recording Area, McGill Univ., Montreal, QC, Canada), Jack Kelly, Richard King, and Wieslaw Woszczyk (Music Res., McGill Univ., Montreal, QC, Canada)

The second generation of the Virtual Acoustic Technology (VAT) Laboratory at McGill University features a new real-time auralizer with a feedback canceller developed by CCRMA at Stanford University, allowing for the simulation of virtual acoustic environments with exceptionally high gain. This study is part of an ongoing research effort focused on integrating algorithmic reverberation tools designed for audio post-production into virtual acoustics at McGill University's VATLab. Previous work has been done using impulse responses (IRs) captured from various acoustic spaces. In contrast, this study focuses on using IRs captured from the legendary Lexicon 960L hardware reverberation unit and using them in the VATLab for recording sessions with musicians. Various 5.1 multichannel presets have been captured as IRs and 3 groups of related 5 channel IRs have been loaded into the existing 15 speaker system of VATLab to simulate a real world, physical Lexicon 960L "environment" through virtual acoustics. Objective measurements following the ISO 3382-1 and 3382-2 standards in the VATLab have been performed to measure the effect of the physical room and analyze the effects of changing different algorithmic reverb parameters such as Diffusion, Early Level Master Control, Early Rolloff, Early or Reflection Delays on the simulated acoustical environments.

3:30

2pAAa10. Musician-led performance perspectives in virtual acoustics. Kathleen Ying-Ying Zhang (Sound Recording Area, McGill Univ., McGill University, Dept. of Music Res., Montreal, QC H3A1E3, Canada, yingying.zhang@mail.mcgill.ca), Aybar Aydin, Vlad Baran, Richard King, and Wieslaw Woszczyk (Sound Recording Area, McGill Univ., Montreal, QC, Canada)

Musical performance is a complex task that involves juggling performance techniques, creative expression and auditory feedback, all of which in turn affect a musician's perception of their acoustic environment. Synthesized here are several recent studies conducted at McGill University in variable acoustic spaces: the Multimedia Room (with a Meyer Constellation system) and the Immersive Media Lab (utilizing a proprietary Virtual Acoustics Technology system). In studying the perception of room response, variable acoustic environments allow researchers to physically and/or virtually change the character of a space around a musician without the need for a physical venue relocation. These experiments explore different methodologies and approaches to studying the perception of room acoustic response of various vocalists and instrumentalists who completed both solo and group performance tasks. After contextualizing these studies with previous work done in this area, lessons recently learned point towards new methodologies for musician-led perspectives on room perception.

3:45

2pAAa11. Virtual acoustics in distributed performance over a closed audio network. Kathleen Ying-Ying Zhang (Sound Recording Area, Dept. of Music Res., McGill Univ., Montreal, QC H3A1E3, Canada, yingying.zhang@mail.mcgill.ca), Vlad Baran, Aybar Aydin, Michail Oikonomidis, Richard King, and Wieslaw Woszczyk (Sound Recording Area, McGill Univ., Montreal, QC, Canada)

We explore an application of active acoustics used to produce a shared virtual environment for live musical performance. As part of a concert given in the Immersive Media Lab at McGill University, musicians and audience members were located in adjacent but acoustically isolated spaces on the

same digital audio network. With computer generated effects and live instruments, the concert consisted of electroacoustic performances that utilized dynamic virtual environments produced by our Virtual Acoustic Technology (VAT) system as an improvisatory partner and a mixing device to blend the diffusion of electronic and acoustic musical sources. The performance, including its evolving virtual environment, was captured using spatial microphone techniques and distributed in real-time to the audience over an immersive loudspeaker system in an adjacent control room. Audience members were given the opportunity to visit the musicians' performance space in order to compare the reproduction to the original environment. Overall, the blending of computer-generated and acoustic sources created a specific use case for virtual acoustics, while the immersive capture and distribution method examined an avenue for producing a real-time shared experience. Future work in this area includes audio networks with multiple virtual acoustic environments and distributions.

4:00

2pAAa12. Acoustics of two Hindu temples in southern India. Shashank Aswathanarayana (Performing Arts, American Univ., Jack Child Hall, Rm 202, American University, Washington, DC 20016, shashank@american.edu) and Braxton Boren (Performing Arts, American Univ., Washington, DC)

Acoustically important aspects of Hindu worship include chants, bells, conch-shells, and gongs. Conch-shells and gongs are used at various times during puja rituals (Prasad and Rajavel, 2013), throughout which texts from the Vedas and other Sanskrit scriptures are chanted. These Vedic chants have phonetic characteristics such as pitch, duration, emphasis, and uniformity (Beck, 1995; Prasad, 2013). Traditional methods of acoustic characterization (e.g., for churches) are based on time domain characteristics like reverberation time and clarity. We term this as, "time domain soundscape of worship." This is vastly different from Hindu worship which involves extensive use of bells, conch-shells, and gongs in puja rituals, all of which produce unique sonic characteristics making frequency of sound very crucial which we term, "frequency domain soundscape of worship." Beck (1995, 2006, 2012) has explored the sonic aspects of Hindu tradition as they relate to the religion and drawn comparisons with religious practices in other cultures where pertinent. However, a comprehensive acoustic analysis of temples and the characteristics of these sounds within temples is yet to be done. In this paper, we analyze the impulse responses and decay curves measured at the Virupaksha and Vijaya Vittala temples in Southern India.

4:15

2pAAa13. Development of an acoustic design support tool for HVAC ventilation units. Malek Khalladi (MJM Acoust. Consultants Inc., 753 Rue Sainte-Hélène, Longueuil, QC J4K 1K5, Canada, mkhalladi@mjm.qc.ca) and Hong Tong (MJM Acoust. Consultants Inc., Longueuil, QC, Canada)

HVAC equipment is one of the main sources of noise inside or outside a building. Furthermore, many people are exposed daily to HVAC noise, which can lead to health-related troubles. Therefore, an HVAC mechanical unit must be selected to provide an acceptable sound level transmitted to the occupied spaces of a building and not disturb the community. This paper presents a prediction tool named Sound Prediction Tool (SPT) developed by MJM Acoustical Consultants Inc. in 2021 that allows the engineer (i) to design an HVAC unit with specific mechanical components (filter, coil, etc.) according to his input parameters and (ii) to predict and optimize the sound levels produced by the selected design through its casing and openings (inlet and outlet air). This tool has been validated firstly with acoustical tests ''*in situ*'' on several separate Air Handling Units (AHU) manufactured by our customer and secondly with Finite Elements Method (FEM) models using Comsol-Multiphysics. Several examples are presented to demonstrate the usefulness of the proposed tool.

4:30

2pAAa14. The acoustical consultant in the context of the construction of a building. Hong Tong (MJM Acoust. Consultants Inc., 753 Ste-Helene St., Longueuil, QC J4K 3R5, Canada, htong@mjm.qc.ca)

Acoustical engineering or consulting is a niche field. However, it touches on many aspects of building construction and will inevitably cross path with other disciplines such as the architect and the mechanical engineer. The purpose of this article is to expose the role of the acoustical consultant as an important stakeholder in the context of the construction of a building. From his or her role at the design phase of a project to the construction up to the reception. The article will also discuss about the different challenges and constraints as an acoustical consultant and will show the impacts of not implicating one. The technical aspect of the acoustical consultant is important, but there is also aspects of project management that needs to be considered. The moment there is a requirement for acoustics, an acoustical expert must be part of the project team at the earliest stages to deliver a successful project.

Session 2pAAb

Architectural Acoustics and Structural Acoustics and Vibration: Building Envelope Sound Isolation I

Joseph Keefe, Cochair
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Lucky S. Tsaih, Cochair
Dept. of Architecture, National Taiwan Univ. of Sci. and Tech., 43 Keelung Rd, Sec. 4, Taipei, 10607, Taiwan

Chair's Introduction—3:20

Invited Papers

3:25

2pAAb1. Planes, trains, and automobiles: Three case studies of building envelope design addressing transportation noise. Jessica S. Clements (Acoust. Studio, Newcomb & Boyd, LLP, 303 Peachtree Ctr. Ave. NE, Ste. 525, Atlanta, GA 30303, jclements@newcomb-boyd.com), John Garretson (Acoust. Studio, Newcomb & Boyd, LLP, Atlanta, GA), and Kirsten Barringer-Cook (Acoust. Studio, Newcomb & Boyd, LLP, Atlanta, GA)

This presentation will focus on three case studies of designs to limit transportation noise through the building shell to support the successful use of the building. The first project focuses on an office building located close to beautiful water views that desired a building shell made primarily of glass. However, it will be located directly under the flight path of a nearby US Airforce base. The second project considers a hotel conference center located directly adjacent to a heavy rail line. The third project address movie sound stages located adjacent to a major highway and directly under the flight path of Hartsfield Jackson International Airport, known as the busiest airport in the world. Site measurements were taken for each of these locations and a detailed review of the data used to determine which noise sources and frequencies of concern would be the focus of the design. The presentation will present examples of the collected data, the design strategies used, and discuss the challenges faced by the design team.

3:45

2pAAb2. Transit-oriented development: A case study of exterior-to-interior noise control for buildings near transit and highway noise sources. Leisa Nalls (Wilson Ihrig, 5900 Hollis St., Ste. T1, Emeryville, CA 94608, lnalls@wilsonihrig.com), Deborah Jue (Wilson Ihrig, Emeryville, CA), and Silas Bensing (Wilson Ihrig, New York, NY)

A case study of acoustical design for mixed-used, transit-oriented development is presented. The project is two, six-story mixed-used residential buildings near a rapid transit station and heavily traveled elevated highway in the San Francisco Bay Area. The buildings are metal frame construction over concrete podiums. The California Building Code requires the building shell to be designed to provide interior noise from exterior noise sources not to exceed 45 dBA Ldn. Exterior noise due to trains and buses serving the transit station, and traffic on the elevated highway, required high sound attenuating windows and acoustically rated walls and roofs. Initially, none of the exterior wall assemblies included batt insulation in the stud cavities, only continuous rigid insulation behind the façade. Modeling was done to compare the exterior wall assemblies with and without batt insulation in the stud cavities combined with various window configurations. Estimates of the building shell transmission loss are presented along with a summary of noise control recommendations for building roof, windows, and doors.

4:05

2pAAb3. Residential sound attenuation: Criteria and approaches. Joseph Keefe (Ostergaard Acoust. Assoc., 1460 US Hwy. 9 North, STE 209, Woodbridge, NJ 07095, jkeefe@acousticalconsultant.com)

A few case studies regarding residential building envelope sound isolation case studies will be presented. The case studies will focus on interior and exterior criteria, measured exterior sound exposure data, and prediction methods/strategies for achieving appropriate wall/window/door/roof sound attenuation values. The presentation will also discuss how to effectively communicate technical and subjective information to project stakeholders.

4:25

2pAAb4. Silencing the rolling thunder: An almost unthinkable acoustic isolation facade solution. Shane J. Kanter (Threshold Acoust., 141 W Jackson Blvd, Ste. 2080, Chicago, IL 60604, skanter@thresholdacoustics.com), Nicolaus T. Dulworth, and Carl P. Giegold (Threshold Acoust. LLC, Evanston, IL)

Situated along a popular motorcycle route, the Lindemann Performing Arts Center at Brown University posed a unique challenge to its design team — isolating the performance space from external road noise while adhering to the demanding RC 15 performance standard. This presentation provides a comprehensive exploration of the intricate development and testing of a robust yet relatively lightweight acoustic isolation façade system. The ultimate approach employed unconventional damping methods, ensuring compliance with dew point requirements. The paper delves into the nuances of the design process, with a focus on key elements such as glazing systems, roofing solutions, access doors, and gypsum buildups that push the boundaries of conventional construction, challenging the possible maximum length of a drywall fastener. Attendees will gain valuable insights into the innovative strategies implemented to achieve effective acoustic isolation without compromising the high-performance standards demanded by the Lindemann Performing Arts Center. This presentation serves as a valuable case study, offering a deeper understanding of the intersection between architecture, acoustics, and the practical challenges encountered in the design of performance spaces.

4:45

2pAAb5. Design condiserations for secondary glazing systems. Jennifer Levins (Acentech, 33 Moulton St., Cambridge, MA 02138, jlevins@acentech.com) and Benjamin E. Markham (Acentech, Cambridge, MA)

Addressing high noise levels from intermittent environmental sources, such as from train pass-bys, presents both acoustical and architectural challenges. One effective noise mitigation approach is to provide secondary storm glazing, but this is not always favored by architects, developers, and builders due to operational and aesthetic shortcomings, budget considerations, and installation challenges. These pitfalls are not always well-communicated among the design team, leading to a compromised solution. In this case study, we will discuss the concerns related to secondary glazing and the collaborative process used with the design team—informed by lessons learned on prior projects—to find a solution that addresses the various project goals.

5:05

2pAAb6. New York City noise code and open windows. David Manley (DLR Group, 6457 Frances St., Omaha, NE 68106, dmanley@dlrgroup.com) and Steph Ahrens (DLR Group, Omaha, NE)

The New York City Noise Code requires compliance with noise levels from stationary sources such as HVAC equipment to residential receivers to be measured inside the receiver location with an open window or patio door. The presentation will include the review of recent case studies completed by DLR Group with school renovations and calculated noise impact to adjacent properties. Discussion of the corrections used for window openings will be included.

Session 2pAB

Animal Bioacoustics, Computational Acoustics, Acoustical Oceanography, Underwater Acoustics, and Signal Processing in Acoustics: Data to Information, Navigating the Application of Acoustic Data for Conservation II

Megan F. McKenna, Cochair
Cooperative Institute for Research in Environmental Sciences, University of Colorado Boulder, 442 Gibson Ave., Pacific Grove, CA 93950

Carrie Wall, Cochair
University of Colorado, 325 Broadway, Boulder, CO 80305

Contributed Papers

2:45

2pAB1. Acoustic monitoring of marine environmental quality: An example from the Estuary and Gulf of St. Lawrence. Florian J. Aulanier (Fisheries and Oceans Canada, Pêches et Océans Canada, 850, Rte. de la Mer, P.O. Boîte 100, Mont-Joli, QC G5H 3Z4, Canada, florian.aulanier@dfo-mpo.gc.ca), Yvan Simard, Clément Juif (Fisheries and Oceans Canada, Mont-Joli, QC, Canada), and Samuel Giard (Pêches et Océans Canada, Mont-Joli, QC, Canada)

As a part of the Canada's Ocean Protection Plan, Fisheries and Oceans Canada has joined the efforts to better understand and monitor the effects of anthropogenic noise on marine environmental quality. Since 2017, underwater acoustic observatories were put in place across endangered whale habitats leading to the acquisition of big underwater acoustic dataset to process and analyze. In the Estuary and Gulf of St. Lawrence, underwater noise has been continuously monitored at 13 locations (6 to 10 simultaneously) between 2018 and 2023 at sampling rate up to 256 ksps in order to better understand the effect of shipping noise on marine environmental quality of the endangered St. Lawrence estuary beluga habitat. In this presentation, the data analysis pipeline from *in situ* sampling to processing is detailed, including recording schemes, data quality and control, soundscape cube, source separation, multi-scale statistics on noise levels and risk of impacts on habitat quality and visualization. These steps are used to identify, characterize and quantify daily to interannual spectral variability of underwater noise and their relationship with local environmental forcings such as shipping, wind, ice, tides, and currents at targeted locations. Ultimately, these results are used to provide support to (1) marine conservation and spatial planning initiatives from DFO and the Saguenay St. Lawrence Marine Parc; and (2) assess the predictability power of the outputs of soundscape modeling.

3:00

2pAB2. Soundscape characterization in the Central Arctic Ocean ecosystem (Svalbard) using acoustic indices. Andrea Lynn (Environ. Studies, Antioch Univ. New England, 40 Avon St., Keene, NH 03431-3516, alynn1@antioch.edu)

An increasingly warmer, less frozen Arctic is opening further to vessel traffic, transforming dominant ambient noise sources in the underwater soundscape. Chief sources may be shifting from wind to ship noise. Collecting data that help explain the changing composition of the soundscape may offer insights to guide regulation of noise in the ocean, furthering management and conservation efforts. Hydrophones connected to field recorders were used in this study to characterize the soundscape and to study marine mammal presence and ship noise. Hydrophones were deployed at 44 locations from a 13.8 m Ovni 445 sailing vessel between 9° E and 19° E and 69° N and 80° N in April 2023. Acoustic indices were utilized to assess soundscape composition. Vessel locations were confirmed using Automatic Identification System (AIS) marine traffic data. Wind, waves, and ice (geophony) dominated the soundscape's acoustic signature in remote locations, while human-caused sounds (anthrophony) were significant near Arctic shipping routes, fishing areas, and in fjords. Marine mammal vocalizations were detected near the ice edge, at fjord mouths, and in fjords. This acoustic characterization study provides a glimpse into the sonic sources and balance of sounds in the soundscape at present, essential data as the region rapidly transforms.

3:15

2pAB3. Bulk analysis of underwater sound spectra from vibratory pile driving of cylindrical steel piles in coastal waters. James Caplinger (Office of Protected Resources, National Oceanic and Atmospheric Administration (NOAA) Fisheries, 1315 East-West Hwy., Silver Spring, MD 20910, james.caplinger@NOAA.gov), Cara Hotchkin (Office of Protected Resources, National Oceanic and Atmospheric Administration (NOAA) Fisheries, Silver Spring, MD), Reny Tyson Moore (Contractor with Ocean Assoc. Inc., Office of Protected Resources, National Oceanic and Atmospheric Administration (NOAA) Fisheries, Silver Spring, MD), and Shane Guan (Div. of Environ. Sci., Bureau of Ocean Energy Management, Sterling, VA)

Vibratory pile driving is a common means of pile installation in both near-shore and offshore environments. Spectral knowledge of the noise produced by this activity is crucial in modeling and assessing the impacts to aquatic species. With the goal of providing generic representative spectra to better inform propagation modeling and related impacts, approximately 80 near-pile measurements of vibratory pile driving of cylindrical steel piles from more than 19 hydroacoustic reports were analyzed. Included pile diameters range from 24–48 inches and all measurements were of activities in U.S. coastal waters. Results include representative spectra based on the mean and median spectra binned by pile size and the presence of noise mitigation systems, an examination of the variation in the data, marine mammal hearing group weighted spectra, broadband sound pressure levels, and comparisons with general knowledge of vibratory pile driving spectra characteristics in previous literature. Finally, data availability and larger efforts aimed at a comprehensive underwater pile driving acoustic spectral database are discussed.

3:30

2pAB4. Using before-after control-impact methodology to quantify effects of full ship shock trial explosions on marine fauna. Kerri D. Seger (Appl. Ocean Sci., 2127 1/2 Stewart St., Santa Monica, CA 90404, kerri.seger.d@gmail.com), Shyam Madhusudhana (Curtin Univ., Perth, Western Australia, Australia), Holger Klinck (K. Lisa Yang Ctr. for Conservation Bioacoustics, Cornell Univ., Ithaca, NY), Kevin D. Heaney (Appl. Ocean Sci., Fairfax Station, VA), and John Boyle (Appl. Ocean Sci., Seattle, WA)

In the summer of 2021, the US Navy conducted a Full Ship Shock Trial (FSST) for the USS Gerald R. Ford. This involved three large underwater explosions off the coast of Florida, USA. We collected underwater acoustic recordings, using low-sensitivity recorders, for the Naval Undersea Warfare Center to validate their underwater acoustic propagation models. We also deployed SoundTraps on the shallow moorings to collect additional acoustical biologics data in hopes of measuring any acoustic responses of marine fauna to the explosions. The acoustic energy of the explosions did not propagate up the continental slope and were hence not captured by the SoundTraps on the shallow moorings. However, our analyses did yield some notable changes in acoustic behavior after as compared to before the explosions. Here, we describe how our field plan was designed for before-after control-impact (BACI) hypothesis testing and discuss our analyses and findings of the few significant cases. These results lend insight for improving the impact assessments and conducting behavioral response studies during a future FSST or other large underwater explosions.

3:45

2pAB5. Acoustic monitoring of harbor porpoises in Hood Canal, WA: Efforts to improve detection of a highly cryptic cetacean. Asila Ghoul Bergman (Res. and Undersea Test Ranges, U.S. Navy, NUWC Div., Keyport, Res. & Undersea Test Ranges, Code 214, Bldg 1074L, Keyport, WA 98345, asila.m.bergman.civ@us.navy.mil) and Dawn Grebner (Res. and Undersea Test Ranges, U.S. Navy, NUWC Div., Keyport, Keyport, WA)

Harbor porpoises (*Phocoena phocoena*) are the smallest, most abundant cetaceans occurring within the waters of the US Navy's Northwest Training and Testing Ranges. The ability to detect harbor porpoise and monitor their presence over time is essential to assessing potential impacts of Navy activities. This can be difficult due to their highly cryptic nature exhibited in their surface behavior (vessel avoidance) and acoustic repertoire (producing only high-frequency clicks). This presents significant challenges for both visual and acoustic (PAM) survey methods. Here, we discuss the Navy's efforts to improve harbor porpoise monitoring capability using an automated acoustic detection system. We tested multiple detection algorithms and evaluated their performance in identifying and extracting harbor porpoise clicks from acoustic recordings collected in Hood Canal, WA. Using this detection data combined with data collected from concurrent visual surveys of Hood Canal, we explored how acoustic and visual methods may be used in concert to improve harbor porpoise monitoring. Hood Canal provides a unique opportunity to do this, as it is the only known US Navy range with a resident population of harbor porpoise. These efforts will also inform future behavioral response studies needed to measure the impact of Navy sounds to individual animals.

4:00–4:15 Break

4:15

2pAB6. Automated detection of fin whale calls recorded with distributed acoustic sensing. Quentin Goestchel (School of Oceanogr., Univ. of Washington, 1503 NE Boat St., Seattle, WA 98115, qgoestch@uw.edu), William S. Wilcock (School of Oceanogr., Univ. of Washington, Seattle, WA), and Shima Abadi (Univ. of Washington, Seattle, WA)

As the global biodiversity crisis intensifies, wildlife monitoring has become a necessity. From an oceanographic perspective, the tracking of baleen whales allows studies of their habitat use and can support call density estimates, which are indicative of ocean health. Distributed Acoustic Sensing (DAS) is a promising observational technique that measures strain rates along a spare optical fiber with a spatial resolution of tens of meters to distances of $\sim$100 km. Over 4-days in November 2021, a public domain DAS dataset was collected on the submarine cables of the Ocean Observatories Initiative Regional Cabled Array off the coast of central Oregon. The experiment recorded the acoustic signals from tens of thousands of fin whale calls as well as blue whale calls, ship signals and T-phases. We will describe preliminary efforts to develop automated methods to detect and localize fin whale calls using small representative examples of the DAS data. Our longer-term goal is to extend this approach to the full data set to understand the sensitivity of DAS to fin whale calls as function of cable geometry and seafloor characteristics and to study the distribution of calling fin whales in coastal waters off central Oregon. [Work supported by ONR.]

4:30

2pAB7. Integrating ocean soundscape modelling and mapping into marine environment quality assessment and marine spatial planning. Florian J. Aulanier (Fisheries and Oceans Canada, Pêches et Océans Canada, 850, Rte. de la Mer, P.O. Boîte 100, Mont-Joli, QC G5H 3Z4, Canada, florian.aulanier@dfo-mpo.gc.ca), Patrice Lebel (Ismer, UQAR, Trois-Rivières, QC, Canada), and Yvan Simard (Fisheries and Oceans Canada, Rimouski, QC, Canada)

As on land, underwater anthropogenic noise and its potential impacts on marine ecosystems have been a growing concern in the past three decades. Initially focusing on louder noise sources, acute physical and behavioral impacts on marine mammals and commercial fish, governmental agencies started to integrate soundscapes into marine spatial planning. However soundscape science has to deal with a large number of metrics and variables (time, space, frequencies, species, types of impacts, sound sources,…) and uncertainties to be able to bring a scientifically robust and reliable support to decision making process. This is a real challenge to integrate and communicate to a vast diversity of stakeholders. To address this challenge, we present a study of the impact of shipping noise on the St Lawrence Estuary and Gulf ecosystems and in particular on endangered marine mammals. The methodology uses probabilistic underwater acoustic modelling to produce 3D-maps of acoustic-field statistics and risk of impacts at daily, weekly, monthly and annual scale over few year-cycles. Those maps are then fed into a web application capable of handling terabytes of geospatial raster's which allows to produce statistics on user-defined area interactively in order to explore and support collaborative decision making process between marine spatial planner and stakeholders. Details on probabilistic methodology and practical examples will be shown, with concluding remarks on gaps and remaining challenges.

4:45

2pAB8. Moving cargo, keeping whales: Investigating solutions for ocean noise pollution. Vanessa M. ZoBell (Scripps Inst. of Oceanogr., 812 Ocean Surf Dr., Solana Beach, CA 92075, vmzobell@ucsd.edu), John Hildebrand, and Kait Frasier (Scripps Inst. of Oceanogr., La Jolla, CA)

Human activities introduce high levels of noise into the ocean. Commercial shipping, in particular, has increased to the point that ships make a larger contribution to ocean noise than natural noise sources for most ocean locations and over a broad range of frequencies. Primeval ocean noise levels, those that would have been experienced before the advent of human-made noise in the ocean, are largely unknown. Ocean noise monitoring efforts began post-industrialization, leaving baseline sound levels under which marine organisms evolved unclear. This study modeled primeval (wind-driven) ocean noise levels and modern (ship noise plus wind noise) ocean noise levels in the Santa Barbara Channel off Southern California. The modern noise levels were validated with acoustic measurements from two sites equipped with High-frequency Acoustic Recording Packages. There was good agreement between the modern noise level models when compared to measured levels for high frequencies, and at a site shielded by islands from long range sound propagation. The lower frequency acoustic environment, modeled at 50 Hz, was more degraded than the higher frequency noise levels, modeled at 1000 Hz. This model can be used to identify target regions and times for noise reduction efforts, as well as model future scenarios for noise reduction to identify techniques with the greatest potential for conservation.

5:00

2pAB9. Masking as a means to quantify impact of increased commercial shipping on southern resident killer whales in the Salish Sea. Rianna Burnham (Dept. of Fisheries and Oceans, Inst. of Ocean Sci., Sidney, BC V8L 5T5, Canada, rianna.burnham@dfo-mpo.gc.ca), Svein Vagle, and Maximilian Lauch (Dept. of Fisheries and Oceans, Victoria, BC, Canada)

The push to report underwater soundscape measures in standardized forms allows for comparison between sites and through time. This identifies regions most impacted by anthropogenic noise sources, yet may not fully address how the characteristics of these sources affect the marine life that these areas host. In risk assessments, a move away from the use of noise thresholds encourages the use of tools such as the "maskogram" or consideration of "active" or "listening" space changes. Here, we propose the use of masking metrics, whereby the reduced efficacy of communication and echolocation signals of marine species, can be quantified as a loss of effective range. We show that this may be an effective way to consider potential impacts of noise in a more species-centric way. Using data on southern resident killer whales (SRKW, *Orcinus orca*) in waters of the coast of British Columbia, we consider how near-future scenarios in vessel presence, including increased number and size of vessels to support growing commerce and a port expansion, will alter both the soundscape and SRKW acoustics use. We will then show how masking metrics can be used to understand the impact to SRKW, and better direct management measures to lessen disturbance.

5:15

2pAB10. Using passive acoustic data to understand sustainable vessel operations. Megan F. McKenna (Cooperative Inst. for Res. in Environ. Sci., Univ. of Colorado Boulder, 442 Gibson Ave., Pacific Grove, CA 93950, megan.mckenna@noaa.gov), Lindsey Peavey Reeves (Office of National Marine Sanctuaries, National Oceanic and Atmospheric Administration, Silver Spring, MD), Timothy Rowell (Southeast Fisheries Sci. Ctr., National Marine Fisheries Service, National Oceanic and Atmospheric Administration, Beaufort, NC), Sofie Van Parijs (Southeast Fisheries Sci. Ctr., National Oceanic and Atmospheric Administration, Woods Hole, MA), Carrie Wall (Cooperative Inst. for Res. in Environ. Sci., Univ. of Colorado, Boulder, CO), and Leila Hatch (Office of National Marine Sanctuaries, National Oceanic and Atmospheric Administration, Scituate, MA)

Marine vessels support diverse ocean economic sectors. As society works towards a sustainable ocean economy, a variety of vessel initiatives are emerging. Passive acoustic monitoring (PAM) offers a method to evaluate these initiatives. Here we review efforts related to vessel noise management in marine protected areas (MPA) and benefits of vessel speed reduction programs (VSRs). Monitoring marine vessel activity and related underwater noise across a network of protected areas, like the U.S. National Marine Sanctuary system, helps managers ensure the quality of habitats used by a wide range of species. Network-wide comparisons of vessel noise revealed a spectrum of conditions, providing robust metrics to help prioritize management and inform condition assessments. These same vessel noise metrics provided insight on noise reduction related to changes in vessel operations during multiple VSRs. The metrics are complementary to other noise reduction metrics currently used to evaluate VSRs. With the growth in VSRs, there is a need to communicate noise reduction at scales relevant to the targeted vessels, typically larger than individual VSRs. Coordinated efforts are advancing to meet these needs. With strategic, systematic, and sustained efforts, PAM can continue to provide key insight on efforts to realize sustainable marine vessel operations.

Session 2pAO

Acoustical Oceanography: Acoustical Oceanography Prize Lecture

David R. Barclay, Chair
Department of Oceanography, Dalhousie University, PO Box 15000, Halifax, B3H 4R2, Canada

Chair's Introduction—1:00

Invited Paper

1:05

2pAO1. Solutions to muddy (geoacoustic inversion) problems. Julien Bonnel (Woods Hole Oceanographic Inst., 266 Woods Hole Rd., MS# 11, Woods Hole, MA 02543-1050, jbonnel@whoi.edu)

Waveguide propagation is a ubiquitous topic in ocean acoustics. This presentation focuses on low-frequency ($f < 500$ Hz) sound propagation in coastal oceans (water depth shallower than 500 m). These environments are highly dynamical and complex, and act as dispersive waveguides, bounded by the sea-surface and the seafloor. The underwater acoustic field can thus be described by a set of modes that propagate with frequency-dependent speeds. Propagation of transient acoustic signals to a distant receiver (range > 1 km) accrues multi-modal dispersion information about the propagation medium. To extract relevant information about the oceanic environment from the acoustic recording, physics-based processing methods must be used. In this presentation, I will briefly review modal propagation and time-frequency analysis. I will then show how these approaches can be combined into a non-linear signal processing method dedicated to extracting modal information from a single receiver: this information is the foundation of a transdimensional inversion method used to characterize the oceanic environment. This method will be used to perform geoacoustic inversion on the New England Mud Patch. Using several datasets collected under different oceanographic conditions, I will notably demonstrate a successful estimation of the seafloor properties, consistent with geophysical core measurements and other inversion studies, even when the water column is dynamical and mostly unknown.

Session 2pBAa

Biomedical Acoustics: General Topics in Biomedical Acoustics: Microbubbles

Brandon Helfield, Chair
Physics, Concordia University, 7141 Sherbrooke Street West, L-SP 365.04, Montreal, H4B 1R6, Canada

Contributed Papers

1:00

2pBAa1. *In vitro* comparison of subharmonic-aided pressure estimation sensitivity among microfluidic monodisperse microbubbles, sonazoid, and definity. Ga Won Kim (Radiology, Thomas Jefferson Univ., Henderson Hall (HND), 1013 NE 40th St., Seattle, WA 98105, gxk181@jefferson.edu), Lisa te Winkle, Wim van Hoeve (Solstice Pharmaceuticals, Enschede, the Netherlands), Kausik Sarkar (Mech. and Aerosp. Eng, George Washington Univ., Washington, DC), Flemming Forsberg, and John Eisenbrey (Radiology, Thomas Jefferson Univ., Philadelphia, PA)

The subharmonic response of microbubble-based ultrasound contrast agents (UCAs) typically exhibits an inverse linear relationship with the surrounding ambient pressure. Although SHAPE has been clinically successful, the sensitivity of SHAPE at lower pressures is suboptimal. While previous studies optimized SHAPE using commercially available UCAs without considering their size distribution, in this study, two SHAPE-specific monodisperse microbubble (MMB) UCAs with different bubble diameters were compared with commercial polydisperse UCAs as well as their buoyancy-separated subpopulations. UCAs were imaged in a hydrostatic tank containing 350 ml of deionized water. The pressure within the tank was increased from 0–100 mmHg. Using a modified Logiq E10 scanner (GE HealthCare) with acoustic power optimization, SHAPE data was acquired using subharmonic imaging mode with a C1-6 curvilinear probe at 2.5 MHz (receiving at 1.25 MHz). After identification of the optimal acoustic output power, MMB with a mean diameter 1.66 mm demonstrated a mean SHAPE sensitivity of −0.207 dB/mmHg ($r^2 = 0.94$). The second MMB with a mean diameter 3.45 mm demonstrated a mean SHAPE sensitivity of −0.300 dB/mmHg ($r^2 = 0.92$). In contrast, this was 2.5–4 times more sensitive than the two polydisperse UCAs (slope range of −0.081 to −0.074 dB/mmHg) and 1.45–5 times more sensitive than the buoyancy-separated UCAs (ranging −0.061 to −0.142 dB/mmHg). These results suggest that SHAPE-optimized MMB are more sensitive to measuring changes in pressure.

1:15

2pBAa2. Size and pressure dependent activity of lipid coated bubbles and finite element simulation of the propgation of focused ultrasound through bubbly media. Amin Jafarisojahrood (Phys., Toronto Metropolitan Univ., 77 harbour square, Apt. 2103, Toronto, NS M5J 2S2, Canada, amin.jafarisojahrood@ryerson.ca), Carly Pellow (Physical Sci., Sunnybrook Health Sci. Ctr., Toronto, On, Cayman Islands), Agata A. Exner (Dept. of Radiology, Case Western Reserve Univ., Cleveland, OH), David Goertz (Physical Sci., Sunnybrook Health Sci. Ctr., Toronto, ON, Canada), and Michael Kolios (Phys., Toronto Metropolitan Univ., Toronto, ON, Canada)

Use of nanodroplets as alternative to bubbles is limited to high pressure applications due to their high vaporization threshold (>1 MPa). For low pressure applications (e.g <800 kPa) and stable bubble activity, size isolated micron or sub-micron bubbles may be used to tackle the pre-focal bubble activity and attenuation. Numerical simulations of the Marmottant model were ran for bubble sizes of 0.45, 1, 2, and 4 μm in response to 1 MHz ultrasound with pressures between 10 and 700 kPa. All agents were volume matched to 20 μl/kg of Definity considering inter-bubble interactions. The pressure-dependent attenuation and the total acoustic power (TAP) were calculated for each population. Finite element simulations (FEMS) were run by taking account the pressure dependent attenuation and sound speed. Using size isolated bubbles an experimental passive cavitation case study was performed for the same exposure conditions and sizes. Numerical results show that TAP and attenuation of the bubbles are size dependent. Bigger bubbles have stronger responses at lower pressures. However, smaller agents exhibit a size-dependent pressure threshold behavior (PT) above which their attenuation and TAP grow stronger than their bigger counterparts in qualitative agreement with experiments. FEMS show that the PT of oscillations may be used to reduce pre-focal attenuation for ultrasound propagation with minimal loss.

1:30

2pBAa3. Microstreaming profile of a phospholipid-coated wall-attached microbubble undergoing shape oscillation. Hongchen Li (Erasmus MC, P.O. Box 2040, Deparment of Biomedical Eng., Rotterdam 3000 CA, Netherlands, h.li@erasmusmc.nl), Yuchen Wang, Ruisheng Su (Erasmus MC, Rotterdam, Netherlands), Christian Cierpka (Technische Universität Ilmenau, Ilmenau, Germany), Michel Versluis (Univ. of Twente, Enschede, Netherlands), Antonius F. van der Steen (Erasmus MC, Rotterdam, Netherlands), Martin D. Verweij (Imaging Phys., Delft Univ. of Technol., Delft, Netherlands), Nico de Jong, and Klazina Kooiman (Erasmus MC, Rotterdam, Netherlands)

Ultrasound-activated microbubbles induce shape oscillation and microstreaming. However, a thorough understanding remains elusive. This study investigated the 3D microstreaming profile of clinically relevant lipid-coated microbubbles undergoing shape oscillation when bound to a wall. Size-controlled biotinylated microbubbles (radius 3–8 μm) were produced by a flow-focusing device and bound to a streptavidin-coated glass. Insonification spanned 85–425 kPa at 25,000 cycles and 1.25 MHz. Two cameras, operating at 5 Mfps and 10 kfps, coupled to a microscope, captured microbubble shape oscillation and cavitation microstreaming, respectively. Astigmatic particle tracking velocimetry measured 3D particle trajectories of 500-nm beads. In the dataset (n = 79), four microstreaming profiles were observed: quadrupole, dipole, radial, and patternless. Modal decomposition revealed that quadrupole patterns resulted from self-interaction of the predominant shape mode, while dipole patterns arose from strong interactions of two nearby modes at the same oscillation frequency. Microstreaming reached 0.002–0.01 m/s at varying acoustic pressures. Quadrupole microstreaming generally induced higher shear stresses than dipole and radial patterns at identical acoustic pressure. A lower shape mode induced higher shear stresses than a higher mode at the same acoustic pressure. The unique microstreaming characteristics yield diverse outcomes in mechanical impact, which could aid efficient therapeutic applications of ultrasound-activated microbubbles.

1:45

2pBAa4. Focused ultrasound pulse repetition frequency impacts bubble activity and tracer delivery during blood-brain barrier opening. Stecia-Marie Fletcher (Radiology, Brigham and Women's Hospital/Harvard Med. School, 221 Longwood Ave., EBRC 515b, Boston, MA 02115, sfletcher4@bwh.harvard.edu), Amanda Chisholm (Radiology, Brigham and Women's Hospital/Harvard Med. School, Boston, MA), Yongzhi Zhang (Radiology, Brigham and Women's Hospital/Harvard Med. School, Boston, MA), and Nathan J. McDannold (Radiology, Brigham and Women's Hospital/Harvard Med. School, Boston, MA)

The impact of pulse repetition frequency (PRF) on microbubble activity and blood-brain barrier (BBB) opening, for a fixed total number of pulses, is unclear. We studied bubble response to a range of PRFs (0.125, 0.25, 0.5, 1, and 2 Hz) by monitoring bubble emissions during focused ultrasound (FUS) sonications of the rat brain (274.3 kHz). When a location was treated with a test PRF and a control PRF of 1 Hz, at the same pressure, the change in the harmonic amplitude compared to the control decreased with increasing PRF, with a median change of 73.8% at 0.125 Hz and −38.3% at 2 Hz. Significant changes were not observed with repeated sonications at the same PRF. Furthermore, between PRFs of 0.25 and 1 Hz, no difference was observed in the threshold for broadband emissions, which are linked to the potential for tissue damage. Fluorescence imaging was used to estimate the concentration of Trypan Blue (TB) dye following a fixed-pressure 75-pulse exposure for PRFs of 1 and 0.25 Hz in rats. A 0.25 Hz PRF led to a 68.2% increase in the mean concentration measured at the target, with a 53.9% increase in the mean harmonic sum compared with a 1 Hz PRF. Finally, a harmonic emissions-based controller at a PRF of 0.25 Hz yielded similar TB delivery, with fewer instances of petechiae observed through histology, compared to the same controller at 1 Hz. These results may be adapted to improve the clinical safety margin of FUS-mediated BBB opening and to improve the sensitivity to detecting small harmonic signals from cavitating microbubbles.

2:00

2pBAa5. An *in vitro* investigation into Lumason's utility for subharmonic-aided pressure estimation with direct comparison to Sonazoid and Definity. Hailee Mayer (Radiology, Thomas Jefferson Univ., 1411 Walnut St., Apt. 706, Philadelphia, PA 19102, hailee.mayer@jefferson.edu), Ga Won Kim, Priscilla Machado, John Eisenbrey, Trang Vu (Radiology, Thomas Jefferson Univ., Philadelphia, PA), Kirk Wallace (GE HealthCare, Niskayuna, NY), and Flemming Forsberg (Radiology, Thomas Jefferson Univ., Philadelphia, PA)

Subharmonic-aided pressure estimation (SHAPE) with ultrasound contrast agents (UCAs) has shown promising results *in vitro* and *in vivo*. Most commercial UCAs show an inverse linear relationship between subharmonic signal and hydrostatic pressure. However, conflicting trends have been reported for Lumason. Thus, this study investigated the subharmonic (i.e., SHAPE) response of Lumason and directly compare to Sonazoid and Definity in a static and a dynamic *in vitro* system using a clinical Logiq E10 scanner. SHAPE signals were acquired (in triplicate) using a C1-6 probe (transmit/receive: 2.50/1.25 MHz) in increments of 10 mmHg from 0 to 200 mmHg in the static tank, and peak pressures of 10, 20, 30, and 45 mmHg in the dynamic system. Sonazoid and Definity maintained consistent inverse linearity in both static and dynamic conditions, with average sensitivities of −0.07 dB/mmHg (r = −0.88) and −0.04 dB/mmHg (r = −0.97) in the static tank, and −0.14 dB/mmHg (r = −0.86) and −0.10 dB/mmHg (r = −0.85) in the dynamic system, respectively. Lumason exhibited a triphasic behavior; from 0–90 mmHg, SHAPE increased with increasing hydrostatic pressure (0.07 dB/mmHg; r = 0.98), while from 90 to 140 mmHg, the response plateaued (r = 0.32), before decreasing with increasing pressure from 140 to 200 mmHg (−0.14 dB/mmHg; r = −0.95). The subharmonic response of Sonazoid and Definity continues to be well-understood, but further investigations into Lumason's SHAPE response is needed before clinical translation.

2:15

2pBAa6. The effect of flow rate on sonoporation within individual *ex vivo* mesenteric arteries. Stephanie He (Biology, Concordia Univ., 7141 Sherbrooke St. West, Loyola Campus SP-501.06, Montreal, QC H4B 1R6, Canada, stephanie.he@mail.concordia.ca), Davindra Singh (Biology, Concordia Univ., Montreal, QC, Canada), and Brandon Helfield (Phys., Concordia Univ., Montreal, QC, Canada)

This study investigates the impact of vascular flow rate on the efficiency of sonoporation in viable mesenteric arteries ((391 ± 35) μm in intralumenal diameter), shedding light on the applications of ultrasound and microbubble-mediated (USMB) therapies. Isolated viable rat mesenteric arteries, which retained physiological functionality, exhibited reversible responses to vasoactive agents when mounted on a pressure myograph throughout the experiments. We then subjected these vessels to ultrasound (2.25 MHz) and microbubble treatment (Definity) under varied acoustic and flow conditions and assessed sonoporation through propidium iodide (PI) fluorescence microscopy. Results disclosed a strong correlation between microbubble flow rate, duty cycle, and sonoporation efficiency. Higher flow rates and duty cycles were associated with increased PI-positive cell counts, signifying more efficient cellular permeability. Post-treatment viability assays affirmed vessel integrity. These findings underscore the pivotal role of vascular flow rate in shaping therapeutic efficacy within individual vessels. The implications extend to refining USMB therapies in diverse disease scenarios, emphasizing the necessity for meticulous parameter selection to ensure both effectiveness and safety. Overall, the study furnishes valuable insights for enhancing the success and applicability of USMB-based therapeutic approaches in cardiovascular and oncological contexts.

2:30

2pBAa7. Dosage dependent subharmonic response and ambient pressure sensitivity of sonazoid microbubbles for subharmonic aided pressure estimation. Mehmet Yapar (Mech. and Aerosp. Eng, George Washington Univ., Sci. & Eng. Hall, 800 22nd St NW, Washington, DC 20052, myapar23@gwu.edu), Roozbeh H. Azami (Mech. and Aerosp. Eng, George Washington Univ., Washington, DC), Flemming Forsberg, John Eisenbrey (Dept. of Radiology, Thomas Jefferson Univ., Philadelphia, PA), and Kausik Sarkar (Mech. and Aerosp. Eng, George Washington Univ., Washington, DC)

Microbubbles are micron-sized contrast-increasing agents for diagnostic ultrasound imaging. At half of the excitation frequency due to nonlinear oscillations, they generate a subharmonic response, the magnitude of which changes by the ambient pressure. Subharmonic Aided Pressure Estimation (SHAPE) is a noninvasive method to measure local *in vivo* pressure utilizing subharmonic emission from microbubbles using medical ultrasound. In this study, we investigated the *in vitro* dosage effect of Sonazoid, a commercially available microbubble, on SHAPE. We subjected Sonazoid microbubbles to acoustic excitations ranging from 100 to 700 kPa peak negative pressure (PNP) and at a frequency of 3 MHz. We increased the ambient pressure up to 20 kPa and investigated the change in subharmonic signal during eight pressurizing-depressurizing cycles. We observed two different behaviors of subharmonic responses over investigated parameters. Subharmonic response increased significantly with increasing ambient pressure under low PNP excitations for sufficiently high concentrations. Meanwhile, increased ambient pressure at higher PNPs drastically reduced the subharmonic response for sufficiently low concentrations. Reported findings are crucial for characterizing the SHAPE performance of Sonazoid microbubbles, offering valuable insights for improved, consistent, and cost-efficient practice.

2:45–3:00 Break

3:00

2pBAa8. Extracellular matrix stiffness affects microbubble-assisted endothelial permeabilization under flow. Zoe D. Katz (Biology, Concordia Univ., 921 rue du couvent, Home, Montreal, QC h4c 2r7, Canada, katzdaniela15@gmail.com), Elahe Memari (Concordia Univ., Montreal, QC, Canada), and Brandon Helfield (Phys., Concordia Univ., Montreal, QC, Canada)

Cancer immunotherapy has faced challenges in the treatment of solid cancers due to the complex tumor microenvironment (TME), including physical barriers that prohibit immune cell infiltration. Ultrasound (US) stimulated microbubbles present a novel way to potentiate cancer immunotherapy in tumors by permeabilizing the tumor vasculature, however the effect of individual TME parameters on treatment efficacy has not yet been elucidated. Here, we focus on one biophysical parameter, showing that an increase in matrix stiffness increases US-assisted membrane permeabilization. Using a novel setup that allows for real-time visualization under flow, HUVECs seeded on polyacrylamide hydrogels of different stiffnesses (800 and 1600 Pa) showed an increase in sonoporation rates. Collagen models corroborate this trend at two different flow rates for a range of stiffnesses (5, 25, and 75 Pa): for both 5 ml/min and 30 ml/min, there is a relative increase in sonoporation from the stiffer substrate. For a given collagen substrate, there is a significant increase in sonoporation efficiency with increasing flow rate. These data can be used to fine-tune treatments according to different TMEs; further, our findings have applications in designing US parameters for targeted drug delivery and other clinical contexts that consider a variety of tissue stiffnesses.

3:15

2pBAa9. Influence of the liquid ionic strength on the resonance frequency and estimated shell parameters of lipid-coated microbubbles. Amin Jafarisojahrood (Phys., Toronto Metropolitan Univ., 77 Harbour Square, Apt. 2103, Toronto, NS M5J 2S2, Canada, amin.jafarisojahrood@ryerson.ca), Celina Yang (Phys., Toronto Metropolitan Univ., Toronto, ON, Canada), Claire Counil, Pinuta Nittayacharn (Dept. of Radiology, Case Western Reserve Univ., Cleveland, OH), David Goertz (Physical Sci., Sunnybrook Health Sci. Ctr., Toronto, ON, Canada), Agata A. Exner (Dept. of Radiology, Case Western Reserve Univ., Cleveland, OH), and Michael Kolios (Phys., Toronto Metropolitan Univ., Toronto, ON, Canada)

Measurement of the resonance frequency and shell properties of coated microbubbles (MBs) is important in understanding and optimizing their response to ultrasound. In many applications MBs are in ionic liquids; however, the influence of the medium charges on the MB behavior is not well investigated. This study aims to measure the medium charge interactions with MBs by measuring the frequency-dependent attenuation of the same size MBs in mediums of varying charge density. In-house lipid-coated MBs were size isolated to a mean size of 2.35 μm using differential centrifugation. MBs were suspended in distilled water (DW), Phosphate-Buffered Saline solution (PBS1×) and PBS10×. The frequency-dependent attenuation of the MBs solutions was measured using a system of two aligned 100% bandwidth transducers with 10 MHz center frequency. The MB shell properties were estimated by fitting the linear equation to experiments. With increasing salinity, the frequency of the peak attenuation decreased (13, 7.5 and 6.25 MHz in DW, PBS1× and PBS-10×, respectively) and attenuation peak increased (~140%). Estimated MBs shell elasticity decreased by 64% between DW and PBS-1× and 36% between PBS-1x and PBS-10x. The estimated shell viscosity reduced by ~40% between DW and PBS-1× and 42% between PBS-1× and PBS-10×. The significant reduction in the fitted stiffness and viscosity is possibly due to the formation of a densely charged layer around the shell and requires proper inclusion in the related MB models.

3:30

2pBAa10. Controlling the stability of monodisperse lipid-coated microbubbles by tuning their buckling pressure. Michel Versluis (Phys. of Fluids, Univ. of Twente, 7522 NB, Enschede, the Netherlands, m.versluis@utwente.nl), Benjamin van Elburg, Guillaume Lajoinie (Phys. of Fluids, Univ. of Twente, Enschede, the Netherlands), and Tim Segers (BIOS Lab-on-a-Chip group, Univ. of Twente, Enschede, the Netherlands)

Through recent years, major hurdles associated with the microfluidic production of phospholipid-coated monodisperse microbubble have been overcome which has granted them a long shelf life. This has heaved the hope of using their unique properties *in vivo* for improved contrast, controlled therapy, or noninvasive pressure measurement. However, these bubbles are also very sensitive to small changes in ambient pressure which compromises their clinical translation: upon intravenous injection, physiological pressures will cause bubble dissolution, degrading their uniformity. Here, we demonstrate the direct relation between shell buckling and bubble dissolution by acoustically measuring the buckling pressure and the response of bubbles to controlled pressure changes. We show that the concentration of PEGylated lipid, necessary for microfluidic operation, can be used to tune the buckling pressure, and thereby increase bubble stability. We show that this concentration can be changed either directly during bubble production or, more conveniently, by heating the microbubbles after production. The proposed heating step exploits the phase-change of the phospholipids within the shell to selectively expel the PEGylated lipids. Doing so, the bubble buckling pressure can be increased from 0 kPa to 27 kPa which is above physiological pressures.

3:45

2pBAa11. Ultrasound and microbubble mediated T cell modulation in peripheral blood mononuclear cells. Ana G. Baez (Biology, Concordia Univ., 2240 Ave. Madison, Apt. #30, Montreal, QC H4B2T6, Canada, anabaezc1i@gmail.com), Davindra Singh, Stephanie He (Biology, Concordia Univ., Montreal, QC, Canada), Mehri Hajiaghayi, Fatemeh Gholizadeh, Peter J. Darlington (Dept. of Health, Kinesiology & Appl. Physiol., Concordia Univ., Montreal, QC, Canada), and Brandon Helfield (Phys., Concordia Univ., Montreal, QC, Canada)

Ultrasound (US)-stimulated microbubbles are emerging as a revolutionary therapeutic technique, notably within the field of cancer immunotherapy. While this approach has been shown to modify cell permeability and to trigger local anti-tumor effects, the interactions between vibrating microbubbles and T-cells is not well understood. Here, we explore the biophysical impact of microbubbles on T-cell permeability and viability, setting the stage for potential advancements in cellular immunotherapy. Following the activation of fresh human peripheral blood mononuclear cells (PBMCs), cells were incubated with FITC-dextran (10kDa)—used here as a surrogate drug– and exposed in the presence of Definity^{TM}(1:217–cell:bubble ratio) US (1MHz, $N = 1000$, $PRF = 5$ ms for 2 minutes) over a range of peak-negative pressures (208-563kPa). Viability was assessed immediately and post-treatment concurrently with FITC macromolecule uptake using flow cytometry. T-cell population was identified using CD3 and CD4 antibodies (60% CD3 + and 20% CD4+). All conditions examined resulted > 95% viability. Results indicated a significant increase in viably permeated PBMCs with higher acoustic pressures, ranging from 0% to 30% (sham-corrected). When analyzed separately both CD4 + and CD4- T-cells exhibited similar permeability rates (0%–37.5%). The 563 kPa condition yielded the highest permeability with minimal viability loss. These findings open avenues for enhancing the efficacy of cellular immunotherapy for solid tumors.

4:00

2pBAa12. Possible physical mechanisms of the high echogenicity of lipid coated nanobubbles. Amin Jafarisojahrood (Phys., Toronto Metropolitan Univ., 77 Harbour Square, Apt. 2103, Toronto, NS M5J 2S2, Canada, amin.jafarisojahrood@ryerson.ca), Agata A. Exner (Dept. of Radiology, Case Western Reserve Univ., Cleveland, OH), Michael Kolios (Phys., Toronto Metropolitan Univ., Toronto, ON, Canada), and David Goertz (Physical Sci., Sunnybrook Res. Inst., Toronto, ON, Canada)

Lipid coated nanobubbles (NBs) have attracted a great level of interest as ultrasound (US) contrast agents due to their ability to extravagate through leaky tumor vasculature. Their linear resonance frequency is in the range of ~50 M Hz–200 MHz, leading to confusion over their observed strong contrast in diagnostic US frequencies. By solving the Marmottant model, the dynamics of uncoated and lipid coated NBs and microbubbles (MBs) are studied over the frequency and pressure ranges (6–12 MHz, 0.1–1.2 MPa) generally used in diagnostic US. A novel bifurcation analysis in tandem with the analysis of the frequency component of the scattered pressure are conducted. Results show that despite the increased linear resonance frequency and viscous damping due to the lipid shell, buckling and rupture of the shell enhances the generation of the 2nd and 3rd harmonic resonances at pressures as low as 0.2 MPa, not observed with uncoated NBs. The generation of the harmonic resonances are concomitant with an abrupt increase in the 2nd and 3rd harmonic frequency component of the scattered pressure with their pressure threshold (PT) increasing with decreasing NBs size. For the same gas volume, and above the PT, the maximum non-destructive 2nd and 3rd harmonic powers of NBs can become higher than the 2–4 μm MBs. Similar to the lower subharmonic pressure threshold of MBs, the dynamic variation of the NBs effective surface tension due to buckling and rupture may be the potential reason behind the observed harmonic echogenicity.

TUESDAY AFTERNOON, 14 MAY 2024 ROOM 212, 1:00 P.M. TO 4:40 P.M.

Session 2pBAb

Biomedical Acoustics, Physical Acoustics, Signal Processing in Acoustics, and Structural Acoustics and Vibration: Ultrasound Beamforming and its Applications II

Jian-yu Lu, Chair
Bioengineering, The University of Toledo, 2801 West Bancroft Street, Toledo, OH 43606

Invited Papers

1:00

2pBAb1. A high-resolution ultrafast beamformer for surgical microtransducers. Jeremy Brown (Biomedical and Elec., Dalhousie, 5790 University Ave. rm. 247, Halifax, NS B3L1V7, Canada, j.brown@dal.ca)

In recent years, ultrasound imaging has become an indispensable tool for intra-surgical guidance. Most neuro and spine surgeries, however, are now trending towards minimally invasive approaches. In these surgeries, the entire surgery is performed using endoscopic instruments inserted into a small surgical pathway. Consequently, surgical imaging technologies such as ultrasound, must also adapt to be compatible with these new approaches where they are confined to narrow surgical corridors. We have recently developed a novel, high-resolution, endoscopic ultrasound system specifically for guiding these minimally invasive surgeries. The entire system including the probe, the electronics, and software has been designed and fabricated from the ground up. This imaging platform has recently been approved for a preliminary patient imaging study has already produced promising, first-of-its kind data during brain tumour resection and spinal cord surgeries. This presentation will discuss the challenges associated with developing high frequency (30 MHz) miniaturized micro-arrays and the associated electronic beamformer. The beamformer we have developed is based on parallelized FPGAs, can process data at rates higher than 10Tb/s, and generate frame-rates higher than 1kHz. The beamforming architecture incorporates a hybrid imaging mode that interleaves ultrafast imaging for Doppler, and line-by-line focusing for B-Mode. The Doppler mode is overlayed on the BMode image in real-time.

1:20

2pBAb2. Volumetric beamforming in real-time using commodity hardware. Sebastian K. Præsius (Dept. of Health Technol., Tech. Univ. of Denmark, Ørsteds Plads, Bldg. 349, Rm. 226, Kgs. Lyngby 2800, Denmark, sebka@dtu.dk), Lasse T. Jørgensen, and Jørgen A. Jensen (Dept. of Health Technol., Tech. Univ. of Denmark, Kgs. Lyngby, Denmark)

Ultrasound imaging is widely used in medicine for its safety and affordability. However, it demands large transducer arrays because the image resolution is proportional to the number of elements, *N*, typically around 128 for 2D imaging. Three-dimensional imaging requires N^2 audio channels (≈350 GB/s of data) if the equivalent 2D matrix array is used, which is practically impossible to process. To solve this issue, row-column arrays (RCAs) aggregate rows and columns of elements, reducing data rate and processing demands by a factor of N. A novel dual-stage beamforming algorithm further lowers the beamforming operations by N/2, with negligible impact on the image quality. For N = 128, the processing is 8192 times faster than with a matrix array, and it is hypothesized the 3D RCA beamforming can be done in real-time using a commodity graphics card. The beamforming rate of an NVIDIA RTX 4090 GPU was measured for *in-vivo* data from a rat kidney, achieving 1394 full volumes per second, which is over 150 times faster than previous implementations. Combining RCAs with the new beamforming algorithm and GPU processing thus enables volumetric beamforming to be done affordably at the bedside in real-time using a standard scanner and PC.

1:40

2pBAb3. Tunable liquid-based lenses for ultrasonic beamforming. Sina Rostami (Phys. and Astronomy, Univ. of MS, 115 Ashley Way, Oxford, MS 38655, srostami@go.olemiss.edu) and Joel Mobley (Phys. and Astronomy, Univ. of MS, University, MS)

Quasi-planar stepped lenses (QSL's) based on phasing have proven to perform closely to conventional refractive lenses for the generation of both focused and limited diffraction beams. However, since QSLs are passive and made from solid materials, each must be designed with a specific frequency and beam type in mind. In this presentation, we report on a tunable lens with a planar aperture that employs liquid channels to perform phase-based beamforming. The desired phase patterns are produced using specific speed-of-sound profiles and can be flexibly configured for a range of frequencies and beam types without modification. In this talk, we report on the results of both experimental and numerical results for this class of lenses and discuss their potential applications.

1:55

2pBAb4. Modulating ultrasound field phase and amplitude using foam gratings as an alternative to acoustic lenses. Luke A. Richards (Univ. of Oxford, Oxford, United Kingdom), Eleanor P. Stride (Univ. of Oxford, Inst. of Biomedical Eng., Oxford OX3 7DQ, United Kingdom, eleanor.stride@eng.ox.ac.uk), and Robin O. Cleveland (Univ. of Oxford, Oxford, United Kingdom)

There is an increasing number of applications for therapeutic ultrasound, including high intensity focused ultrasound surgery, neuromodulation and drug delivery. For many of these applications, accurate focusing of the ultrasound field at tissue depths of several cm is essential and thus patient-specific acoustic lenses have been widely investigated as an economical alternative to phased array transducers. Lenses can be cast or 3D printed from a range of polymers offering good acoustic impedance matching to human tissue. Disadvantages of polymer lenses, however, include their relatively large size and the difficulties involved in eliminating gas bubbles during production. In this study an alternative approach is investigated using a "grating" comprising a sheet of polymer foam, perforated with a series of pores that act as wave guides. The grating is placed in front of a transducer with the pores arranged to modulate the phase and amplitude of the sound field as required. The gratings can be cheaply and rapidly fabricated, are < 5 mm thick and can achieve phase changes of more than 180° were successfully achieved while maintaining a transmission amplitude of more than 50%.

2:10

2pBAb5. Separate emission/reception transducers for 3D ultrafast ultrasound imaging. Alexis Carrion, Ibrahima Touré, Tamara Krpic, Maxime Bilodeau, Patrice Masson (Université de Sherbrooke, Sherbrooke, QC, Canada), and Nicolas Quaegebeur (Université de Sherbrooke, Mech. Eng. Dept., 2500 blvd Universite, Sherbrooke, QC J1K 2R1, Canada, Nicolas.Quaegebeur@USherbrooke.ca)

Recent advancements in ultrafast 3D ultrasound imaging have revolutionized the field of echography, enabling its extension to novel applications such as cerebral dynamics, cardiac electrophysiology and the quantitative imaging of intrinsic mechanical properties of tumors. To facilitate these advancements, two primary transducer strategies are employed in 3D imaging. The first involves dense 2D probes equipped with a large number of elements, typically exceeding 1024. The second strategy uses row-column addressing, which simplifies the electronic control of the probe elements. Despite their effectiveness, these methods entail complexities in design and fabrication. Addressing these challenges, our study introduces an innovative transducer configuration that distinctly separates emission and reception functionalities. This separation not only simplifies the overall transducer design but also significantly reduces the system's complexity. A sparse array of PVDF transducers, which have been laser micro-machined to ensure acoustic transparency, is used at the reception. For the emission aspect, we employ a specialized acoustic concentrator to emulate point-like emission. The paper presents the detailed design requirements, assembly process, and the operational principles of this novel transducer. Furthermore, an experimental validation is conducted using a CIRS 040GSE phantom model. This validation is crucial to demonstrate the practical applicability and reliability of our transducer in real-world medical imaging scenarios.

2:25

2pBAb6. Miniaturized acoustic concentrators for ultrafast 3D ultrasound imaging. Ibrahima Touré, Maxime Bilodeau (Université de Sherbrooke, Sherbrooke, QC, Canada), and Nicolas Quaegebeur (Université de Sherbrooke, Mech. Eng. Dept., 2500 blvd Universite, Sherbrooke, QC J1K 2R1, Canada, Nicolas.Quaegebeur@USherbrooke.ca)

Ultrafast 3D ultrasound imaging is a rapidly growing research field that enables real-time imaging at a high frame rate and in a non-invasive manner. Currently, to achieve ultrafast 3D ultrasound, plane or diverging waves from virtual sources are used. These virtual sources can be considered as focal points during emission, allowing for improved transmission of ultrasonic energy. However, they have limitations in terms of transmitted energy, complex electronics, and timing issues, which impact the quality of the resulting images. To address this, a study was conducted to design and experiment with new miniaturized ultrasound transmitters based on cylindrical waveguides with tapered sections. These transmitters, with a diameter of approximately 100 μm, can replace the virtual sources. A numerical study based on 2D axisymmetric Finite Element Modeling (FEM) is first performed to model the transmission and reflection of longitudinal ultrasonic waves through various cylindrical rods. The results demonstrated that the transmission depends on the diameter ratio, frequency and longitudinal mode order (L0, L1, L2, etc.). An experimental study is then conducted on a stainless steel waveguide instrumented with 6mm piezo discs. The mechanical energy, estimated using Laser Doppler Vibrometer is transmitted through a 150 μm diameter rod, allowing mechananical focusing in the frequency range between 0.5 and 5 MHz. The application of this method to separate emission/reception transducers is then demonstrated for 3D ultrafast imaging.

2:40–2:55 Break

2:55

2pBAb7. Transmit beam design for tissue harmonic volume imaging of muscular tissues with a row-column array. Maryam Satarpour (Dept. of Bioengineering, Univ. of Pittsburgh, Pittsburgh, PA), Zhiyu Sheng (Dept. of Medicine, Univ. of Pittsburgh, Pittsburgh, PA), John M. Cormack (Dept. of Medicine, Univ. of Pittsburgh, Div. of Cardiology, Dept. of Medicine, University of Pittsburgh, Pittsburgh, PA, jmc345@pitt.edu), and Kang Kim (Dept. of Bioengineering, Univ. of Pittsburgh, Pittsburgh, PA)

Row-Column-addressed Arrays (RCAs), which have recently become commercially available, transmit along the rows and receive along columns of a 2D array in order to produce 3D volume images with low channel count. Imaging of the inherently three-dimensional, layered, and fibrous structure of muscular tissues in 3D using an RCA can improve evaluations of muscle function and disease compared to conventional 2D cross-sectional imaging, but volume images obtained using an RCA suffer from low lateral spatial resolution, low contrast, and grating lobe artifacts. Tissue harmonic imaging (THI) employs nonlinear acoustic effects to image at twice the transmit frequency, and can yield increased spatial resolution, increased contrast, and decreased artifacts from grating lobes. Here we investigate transmission considerations for THI in muscular tissues using a commercially available RCA (Vermon RC6gV). The parameter space is explored efficiently with frequency domain simulations of the 3D Westervelt equation. Pulse inversion imaging is applied experimentally using both steered plane wave and focused transmits to evaluate THI with an RCA in tissue phantoms and in muscular tissue. Pulse inversion volume images are compared to conventional images in terms of signal-to-noise, contrast, and ability to identify layers and fibrous structures in muscle in 3D. [Work supported by NIH HEAL Initiative R61AT012282.]

3:10

2pBAb8. Transducer module apodization for reducing bone heating during focused ultrasound uterine fibroid ablation. Sobhan Goudarzi (Physical Sci. Platform, Sunnybrook Res. Inst., 2075 Bayview Ave., Toronto, ON M4N 3M5, Canada, sobhan.goudarzi@sri.utoronto.ca), Ryan M. Jones, Yin H. W. Lee, and Kullervo Hynynen (Physical Sci. Platform, Sunnybrook Res. Inst., Toronto, ON, Canada)

During MR-guided focused ultrasound (MRgFUS) for uterine fibroids, thermoablation of tissue near spine/hips is challenging due to bone heating that can cause patient pain and potentially damage nerves. Here we investigate transducer module apodization for maximizing the focal-to-bone heating ratio ($\Delta T_{;\mathrm{ratio}}$) *in silico* using a 6144-element flat fully populated phased array operating at 0.5 MHz (Arrayus Technologies, Inc.). Acoustic and thermal simulations were performed using anatomies of ten patients who underwent MRgFUS ablation for uterine fibroids with this device as part of a clinical trial (NCT03323905). Transducer modules (64 elements/module) whose beams intersected no-pass regions were identified, their amplitudes were reduced by varying blocking percentage levels, and the resulting temperature field distributions were evaluated across multiple sonications per patient. For all simulated sonications transducer module blocking improved $\Delta T_{;\mathrm{ratio}}$ compared to no blocking. In 42% of sonications, full module blocking maximized $\Delta T_{;\mathrm{ratio}}$, with mean improvements of $97\% \pm 55\%$ and $47\% \pm 36\%$ in hip and spine compared to no blocking, at the cost of increased focal thermal volumes and acoustic power levels. In the remaining sonications, partial module blocking provided increased $\Delta T_{;\mathrm{ratio}}$ values ($39\% \pm 45\%$ in hip, $19\% \pm 15\%$ in spine targets) relative to full blocking. The optimal blocking percentage varied depending on the specific treatment geometry.

3:25

2pBAb9. Investigation of HIFU beam propagation through acoustic holograms in water. Timofey Krit (Inst. for Diagnostic and Interventional Radiology, Univ. Hospital of Cologne, Kerpener Str. 62, Cologne 50937, Germany, Timofey.Krit@uk-koeln.de), Luisa Brecht, Nina Reinhardt, Johannes Lindemeyer, and Holger Grüll (Inst. for Diagnostic and Interventional Radiology, Univ. Hospital of Cologne, Cologne, Germany)

We investigated variations in the HIFU beam of a clinical system as it traverses acoustic holograms—small artificial objects of different types—within a tank that was filled in with the degassed deionized water. The pressure distribution within the HIFU beam was assessed in multiple planes perpendicular to the beam axis. The ultrasound transducer with the focal length of 14 cm inside the Sonalleve table top (Profound Medical, Mississauga, ON, Canada) generated the focusing ultrasonic beam, which propagated through the water. Each hologram was positioned 3 cm below the focal plane, with precise adjustments of positions and angles for each measurement. Pressure measurements were conducted using the needle hydrophone HNA-0400 (ONDA Corp, Sunnyvale, CA, USA) mounted on a 3D positioning system. The motors of the positioning system were controlled, and pressure values were recorded using a GUI generated in LabVIEW. The displacement step created by the motors on each axis was 0.01 mm. The properties of the holograms were reconstructed, considering the experimental data obtained. Our findings demonstrated that the shape, size, elasticity, and placement of each acoustic hologram significantly influenced the observed field pattern. This work was supported by the German Federal Ministry of Education and Research (“MR-HIFU-Pancreas”, FKZ:13GW0364D).

3:40

2pBAb10. Experimental and numerical comparison of multiple passive beamformers for separating intra- and extra-canal cavitation activity during transvertebral spinal cord therapy. Andrew P. Frizado (Sunnybrook Res. Inst., 2075 Bayview Ave., Toronto, ON M4N 3M5, Canada, andrew.frizado@mail.utoronto.ca) and Meaghan O’Reilly (Sunnybrook Res. Inst., Toronto, ON, Canada)

Distinguishing intra-canal cavitation activity during transvertebral focused ultrasound sonication of the spinal cord is a challenge due to the strong prefocal cavitation emissions in the spinalis musculature overwhelming emissions originating in the canal. To achieve mapping of all cavitation sources simultaneously, two methods for enhancing detection sensitivity are investigated: (1) multiple dynamic ranges within a reconstructed volume and (2) utilizing alternative beamformers to delay-and-sum (DAS) during map reconstruction. The performance of DAS beamforming is compared to a delay-multiply and-sum beamformer (DMAS), with and without a paired multiplicative compounding method (pDMAS). Experiments and simulations were performed on a 128-element, dual-aperture transvertebral array, through stacks of *ex vivo* human vertebra. Numerically and experimentally obtained point spread functions were compared in 3D, producing voxel-wise cross correlation values of 0.84, 0.89, 0.97 ($N = 1$) for the beamformers listed above, respectively, in water. Experimental, transvertebral localizations of canal sources in isolation produced localization error of 2.8 ± 1.2, 2.9 ± 1.4, 2.7 ± 1.3 mm for a single vertebral target ($N = 30$ sonications), respectively. A large numerical data set investigating the prefocal cavitation problem ($N > 160$) is presented and compared with ($N = 9$) experimental data demonstrating enhancement of intracanal sensitivity in cavitation maps.

3:55

2pBAb11. Quantitative ultrasound in synthetic transmission aperture ultrasound imaging. Yuan Xu (Toronto Metropolitan Univ., 350 Victoria St., Dept. of Phys., Toronto, ON M5B2K3, Canada, yxu@ryerson.ca), Na Zhao, Khalid Abdalla, and Keyan Sheppard (Toronto Metropolitan Univ., Toronto, ON, Canada)

Conventional ultrasound imaging mainly provides qualitative information, and the imaging results depend strongly on the operator of the scanners. Quantitative ultrasound (QUS) imaging can provide specific numbers related to tissue properties that can increase the specificity of image findings, leading to improvements in diagnostic ultrasound. Most of the QUS systems use B-mode imaging. B-mode QUS usually has poor spatial resolution. Some of them need a reference phantom, which makes the clinical implementation slow. In addition, different QUS methods use different data acquisition protocols or imaging modes. So far, no *in vivo* system can extract the attenuation coefficient, speed of sound, and microstructure information of biological tissues at a good spatial resolution from one data set in a clinically applicable device. This paper presents our study on developing QUS in STA (Synthetic Transmit Aperture Ultrasound Imaging). STA is shown to have richer information in the raw data than B-mode imaging. Therefore, it has the potential to improve spatial resolution and accuracy and provide richer feature information in QUS. The quantitative information includes, but is not limited to, attenuation coefficient, speed of sound, and tissue microstructure information from one frame of image data in a clinically applicable device.

4:10

2pBAb12. Low-intensity pulsed ultrasound (LIPUS) mediated on-demand release of cannabidiol from the surface of gold nanoparticles. Anshuman Jakhmola (Toronto Metropolitan Univ., Toronto, ON, Canada), Farshad Moradi Kashkooli (Toronto Metropolitan Univ., 350 Victoria St., Toronto, ON, Toronto, ON M5B 2K3, Canada, fmoradik@torontomu.ca), Tyler Hornsby, Michael Kolios (Toronto Metropolitan Univ., Toronto, ON, Canada), Kevin Rod (Toronto Polyclinic, Toronto, ON, Canada), and Jahangir (Jahan) Tavakkoli (Toronto Metropolitan Univ., Toronto, ON, Canada)

Emerging evidence has suggested that natural drugs like Cannabidiol (CBD) have the potential to selectively destroy cancer cells in the body. In this study, we designed gold nanoparticles (GNPs) loaded with CBD molecules and employed a patented custom-designed handheld 1-MHz low-intensity pulsed ultrasound (LIPUS) device to selectively release CBD from the surface of gold nanoparticles in an *ex vivo* tissue model. A $60 \times 73 \times 25\,\mathrm{mm}^3$ *ex vivo* tissue sample was placed in a 3D-printed plastic sample holder, while0.5 ml solution of CBD-loaded GNPs was pipetted into a separate 3D-printed GNP holder. The 3D-printed GNP holder was then inserted into the tissue center, and the combined holders were submerged in a 37 °C acrylic water tank with a water heater for temperature control. The LIPUS device was held at the top of a water tank which directed ultrasound downward towards the region of interest in tissue. Initial results displayed the highest fluorescence intensity at a LIPUS exposure of 4.1 W acoustic power, a 50% duty cycle, and a total exposure time of 5 min. The release of CBD was measured by a dye-based fluorescence assay, and it was found to be significantly higher when applying LIPUS compared to a bulk heating-only treatment using the same temperature profile. It is concluded that both thermal and non-thermal effects caused by LIPUS has a significant impact on CBD release from gold nanoparticles with 42% and 58% for thermal and non-thermal effects of LIPUS, respectively.

4:25

2pBAb13. Real time cavitation monitoring during ultrasound-mediated drug delivery in normothermically perfused human tumour-bearing livers. Michael Gray (Inst. of Biomedical Eng., Univ. of Oxford, Inst. of Biomedical Eng., University of Oxford, Oxford OX3 7DQ, United Kingdom, michael.gray@eng.ox.ac.uk), Brian Lyons, David Johnson (Inst. of Biomedical Eng., Univ. of Oxford, Oxford, United Kingdom), Brannon Nicholls (Medical Sci. Doctoral Training Ctr., John Radcliffe Hospital, Oxford, United Kingdom), Alex Gordon-Weeks (Nuffield Dept. of Surgical Sci., John Radcliffe Hospital, Oxford, United Kingdom), Robert Carlisle, and Constantin Coussios (Inst. of Biomedical Eng., Univ. of Oxford, Headington, Oxford, Oxfordshire, United Kingdom)

In the fight against a broad spectrum of human diseases, cavitation techniques show great promise for overcoming physical barriers that lead to suboptimal uptake of passively administered therapeutics. However, small animal testing of candidate therapies remains a poor predictor of clinical success. Here we demonstrate ultrasound-mediated drug delivery in normal and tumour-bearing human livers infused with protein-based cavitation nuclei (PCaN). Whole and partial human livers were obtained immediately from hepatectomy surgeries and were normothermically sustained using a clinically approved perfusion system (OrganOx Metra). Ultrasound was applied using a 0.5 MHz focused source (Sonic Concepts H107) and was monitored with a calibrated linear array (ATS L7-4) for real time structural and cavitational imaging implemented on an array controller (Verasonics Vantage 256). Specifically, the therapy process was monitored using passive acoustic mapping (PAM) of broadband cavitation emissions, employing a non-adaptive beamformer that deconvolves the array point spread function. Levels of fluorescently labelled drugs incorporated in and co-administered with the PCaN were quantified in blood and tissue samples collected during and following treatment, respectively. This presentation highlights PAM observations of broadband cavitation persistence and drug delivery in untargeted and targeted tissues, including the first ever experiments in tumour-bearing human livers.

Session 2pCA

Computational Acoustics, Animal Bioacoustics, Signal Processing in Acoustics, Biomedical Acoustics, and Architectural Acoustics: Application of Model Reduction Across Acoustics

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

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

Benjamin M. Goldsberry, Cochair
Applied Research Laboratories at The University of Texas at Austin, 10000 Burnet Road, Austin, TX 78758

Chair's Introduction—1:00

Invited Papers

1:05

2pCA1. Modal reduction for exterior acoustic and vibroacoustic problems. Steffen Marburg (School of Eng. and Design, Tech. Univ. of Munich, Boltzmannstr. 15, Garching 85748, Germany, steffen.marburg@tum.de)

Modal analysis, mode superposition and modal reduction are considered standard approaches for interior acoustic problems as well as for pure vibration problems. Things are different for exterior problems. Only a few methods are known to formulate a linearized eigenvalue problem for unbounded acoustic problems. Even fewer techniques on modal superposition and reduction are found in the literature. This talk will review such techniques which are either based on conjugate Astley-Leis infinite elements or on boundary elements for the unbounded region. While the results appear to be promising, a number of open questions need to be solved in this context. Among others, it is yet unclear which modes are actually required for a modal reduction and how to find a method to just evaluate these modes and the related eigenvalues. The situation is different for vibroacoustic radiation problems with clear resonances. In such cases, a modal reduction is able to substantially accelerate the solution process. Future work might lead to modal reduction using modes from non-linear eigenvalue problems which are known from the numerous papers on the boundary element methods published over the last decade.

1:30

2pCA2. Latent space representation method for structural acoustic assessments. Gregory A. Banyay (Appl. Res. Lab., Penn State Univ., State College, PA) and Andrew S. Wixom (Appl. Res. Lab., Penn State Univ., P.O Box 30, M.S. 3220B, State College, PA 16801, axw274@psu.edu)

When targeting structural acoustic objectives, engineering practitioners face epistemic uncertainties in the selection of optimal geometries and material distributions, particularly during early stages of the design process. Models built for simulating acoustic phenomena generally produce vector-valued output quantities of interest, such as autospectral density and frequency response functions. Given finite compute resources and time we seek computationally parsimonious ways to distill meaningful design information into actionable results from a limited set of model runs, and thus aim to use machine learning to perform model order reduction. Unlike time series data for which recurrent neural networks can learn from prior time steps to inform subsequent steps, frequency-dependent data demands a different machine learning paradigm. We thus evaluate the utility of autoencoders to represent structural acoustic results with a low dimensional latent space to enable such objectives as surrogate modeling for design optimization. We demonstrate the accuracy of autoencoder based methods of constructing a manifold representation for frequency dependent functions of varying modal density and damping, and discuss the predictive capability thereof.

1:55

2pCA3. Blind source extraction using Kronecker product structured mixing model. Zbyněk Koldovský (ITE, Tech. Univ. of Liberec, Studentska 2, Liberec 46117, Czechia, zbynek.koldovsky@tul.cz), Jaroslav Čmejla, and Aamir Farooq (ITE, Tech. Univ. of Liberec, Liberec, Czechia)

Blind Source Extraction (BSE) aims at extracting a source-of-interest (SOI) from a mixture of signals observed through multiple sensors. In this work, we consider a parameter-reduced mixing model suitable for sensor arrays structured in a grid. The parameter vector involving weights with which the sensors receive the SOI is factorized into the Kronecker product of two sub-vectors. In addition to the description of the new model, we present an algorithm designed to identify the mixing parameters based on the assumption of independence of the SOI from the other signals in the mixture. In simulations, the performance of the algorithm is compared with that of the fully parameterized counterpart and is compared with the corresponding Cramer-Rao-induced bounds on the achievable Interference-to-Signal Ratio.

2:20

2pCA4. A hybrid formulation for determining the input power in a stiffened plate over a wide frequency range. Nickolas Vlahopoulos (Univ. of Michigan, 2600 Draper Dr., Ann Arbor, MI 48108, nickvl@umich.edu), Sungmin Lee, and Jonmarcos Diaz (Univ. of Michigan, Ann Arbor, MI)

A hybrid analytical/numerical formulation is presented for determining the input power in a stiffened plate over a wide frequency range. A Component Mode Synthesis (CMS) approach is used for developing the hybrid formulation. The structure is divided into a local substructure in the vicinity of the excitation and non-local substructures representing the remaining system. It is considered that the excitation is applied at a small number of discrete locations, therefore the local substructure is a small portion of the overall system. The computational efficiency originates from using an analytical approach and periodic structure theory for determining the dynamic modes and natural frequencies for the non-local substructures. Finite elements are used for determining the static modes for the non-local substructures and also for both the dynamic and static modes of the local substructure. The CMS matrices for the non-local substructures are condensed to their common interface degrees of freedom with the local substructures. In this manner, the non-local substructures are eventually represented as boundary conditions on the local substructure. Results from the hybrid formulation are compared with results from dense finite element models in order to demonstrate the validity and the efficiency of the hybrid method.

2:45

2pCA5. Modeling high-frequency backscattering from an impedance surface using Kirchhoff approximation. Pei-Tai Chen (Ctr. of Oceanic Eng., National Taiwan Ocean Univ., 2 Pei-Ning Rd., Keelung, Taiwan 886, Taiwan, ptchenline@gmail.com)

The Kirchhoff approximation is based on planar wave impinging on a planar surface, which is approximated for high-frequency wave scattering from a curved object. The Kirchhoff approximation is used to assess backscattering from a curved, rigid surface when exposed to high-frequency incident waves. In this presentation, we expand the rigid surface Kirchhoff approximation to accommodate impedance surfaces. The derivation is performed on a planar wave's incident onto an impedance flat surface, resulting in a reflection coefficient, which in turn determines both surface pressure and surface normal velocity. When these two surface quantities are integrated into a Helmholtz integral formula, it allows for the evaluation of backscattering acoustic field pressures. The BeTTSi model, featuring an impedance surface, serves as an illustration for comparing the current Kirchhoff approximation with the solution obtained by a Fast Multipole Expansion Boundary Element Method (FMBEM). The latter is a numerical solution of the corresponding Helmholtz surface integral equation. The numerical discretization involves nearly a million points, representing the degrees of freedom necessary for modeling the BeTTSi under high-frequency incident waves. The comparison involves these two numerical evaluations across various incident wave directions, encompassing scenarios with single reflection and multiple reflections due to surface geometry.

3:10–3:25 Break

Contributed Papers

3:25

2pCA6. Information geometry approach to model reduction. Jay C. Spendlove (Dept. of Phys. and Astronomy, Brigham Young Univ., C341 ESC, Brigham Young University, Provo, UT 84602, jayclark24@gmail.com), Mark K. Transtrum, and Tracianne B. Neilsen (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Information geometry combines the fields of information theory and differential geometry. In information geometry, a model is interpreted as a Riemannian manifold, referred to as the model manifold, where distance on the model manifold corresponds to statistical distinguishability. Boundaries on the model manifold are reduced-order models where a parameter or parameter combination is taken to an extreme limit and drops out of the model. Therefore, by locating the model manifold boundaries, simpler models with fewer parameters can be obtained. A model reduction algorithm which exploits this insight is the Manifold Boundary Approximation Method, or MBAM, which locates manifold boundaries by calculating geodesics on the surface of the model manifold. This can be done iteratively to find increasingly simple reduced models. We demonstrate this method for model reduction using the Pekeris waveguide transmission loss model. A machine learned surrogate model for the Pekeris waveguide is constructed so that the necessary derivatives for calculating geodesics can be easily obtained using automatic differentiation. [Work supported by the Office of Naval research Grant N00014-21-S-B001.]

3:40

2pCA7. Comparison of substructuring methods with interface reduction. Matthew Luu (Penn State, 446 Bluecourse Dr (Apt907), State College, PA 16803, mbl5743@psu.edu), Jon Young (Appl. Res. Lab, State College, PA), and Andrew S. Wixom (Appl. Res. Lab., Penn State Univ., State College, PA)

For complex dynamical systems, substructuring techniques can be applied to simulate the large system more efficiently by handling it as a collection of smaller components. In order to couple these components, the description of the interface between components is very important. Classic substructuring techniques, such as the Craig-Bampton method, reduce the interior modes of the system, but don't reduce the interface description. This can cause unnecessary degrees of freedom kept in the final assembled model. While there have been a few substructuring methods published that do include some amount of interface reduction, they are much more rare and even more rarely applied in practice. Therefore, the goal of this work is to survey these techniques and compare them with a new, line-based orthogonal polynomial interface reduction technique. These methods will be applied to a set of example problems in order to determine their validity and convergence compared to the full model.

3:55

2pCA8. Accurate and computationally efficient basis function generation using physics informed neural networks. Nathan Cloud (Appl. Res. Labs. at The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX, nathan.cloud@arlut.utexas.edu), Benjamin M. Goldsberry, and Michael R. Haberman (Appl. Res. Labs. at The Univ. of Texas at Austin, Austin, TX)

Basis functions that can accurately represent simulated or measured acoustic pressure fields with a small number of degrees of freedom is of great use across various applications, including finite element methods, model order reduction, and compressive sensing. In a previous work [B. M. Goldsberry, J. Acoust. Soc. Am. 153, A193 (2023)], basis functions were derived for an element in a given mesh using a combination of interpolation functions defined on the boundaries of the element and the Helmholtz-Kirchhoff (HK) integral. This forms a new interpolatory basis set that efficiently and accurately represents the interior of the element. However, the previous analysis was limited to a two-dimensional rectangular element. In this work, physics informed neural networks (PINN) are investigated as a means to generate HK basis functions for general element shapes. PINNs have been previously shown to accurately learn solutions to parameterized partial differential equations. The element geometry parameterization and the boundary interpolation functions are given as inputs to the PINN, and the output of the PINN consists of the physically accurate basis functions within the element. Details on the implementation and training requirements on the PINN to achieve a desired accuracy will be discussed. [Work supported by ONR.]

4:10

2pCA9. An overview of symmetry-based techniques used for sound power application with illustrative case studies. Ian C. Bacon (Phys. & Astronomy, Brigham Young Univ., 333 W 100 S, Provo, UT 84601, icbacon@byu.edu), Micah Shepherd, and Scott D. Sommerfeldt (Phys. & Astronomy, Brigham Young Univ., Provo, UT)

This work investigates the growing application of vibration-based sound power (VBSP) measurements in structural acoustics. The VBSP method involves multiplying a measured velocity vector by a geometry-dependent radiation resistance matrix to compute the frequency-dependent sound power. However, computational challenges arise during matrix construction, even for simple geometries. To address this, inherent symmetries in the radiation resistance matrix for both baffled and unbaffled geometries are exploited. Multi-layered Toeplitz symmetries emerge for baffled conditions, while centrosymmetric symmetries are manifest for unbaffled conditions. These symmetries enable an efficient compression and subsequent reconstruction of the resistance matrix. Simple case studies demonstrating the temporal costs for flat, curved, and arbitrarily curved plates in unbaffled conditions will be presented. [Funding for this work was partially provided by the National Science Foundation (NSF).]

TUESDAY AFTERNOON, 14 MAY 2024 — ROOM 204, 1:00 P.M. TO 4:05 P.M.

Session 2pEA

Engineering Acoustics, Biomedical Acoustics and Physical Acoustics: Things That Go Boom: High Amplitude Acoustic Sources

Thomas E. Blanford, Chair

University of New Hampshire, University of New Hampshire, Durham, NH 03824

Invited Papers

1:00

2pEA1. History of high-amplitude underwater sound source research at the University of Texas Applied Research Laboratories. Preston S. Wilson (Walker Dept. Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, 204 East Dean Keeton St., Mail Stop: C2200, Austin, TX 78712-0292, pswilson@mail.utexas.edu), Andrew R. McNeese, Kevin M. Lee, and Robert L. Rogers (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

The Applied Research Laboratories has an extensive history of supporting underwater acoustic research through the development of novel high-amplitude underwater sound sources, such as the Plasma Sound Source (PSS), the Combustive Sound Source (CSS) and the Rupture Induced Underwater Sound Source (RIUSS). These sources generate broadband acoustic pulses capable of long-range propagation and seabed penetration, and can be viewed as alternatives to explosive sources. The PSS is based on the discharge of electrical energy stored in capacitors which results in a plasma bubble. The CSS consists of a submersible combustion chamber, open to the water, which is filled with a combustive mixture that is ignited via spark. Upon ignition, the combustive mixture is converted into high temperature combustion byproducts which expand and ultimately collapse, thereby radiating an acoustic pulse. In this talk, the motivation for alternative impulsive sources is discussed and the PSS is briefly discussed. Next, the early development of CSS, through the end of the 1990’s, is described, which led to the basic understanding of the CSS and the parameters required to modify its output. In the following talk further development of CSS, and the invention of RIUSS, are discussed. [Work supported by ONR and NAVO.]

1:20

2pEA2. Further development of high-amplitude underwater sound sources at The University of Texas Applied Research Laboratories. Andrew R. McNeese (Appl. Res. Labs., The Univ. of Texas at Austin, 10,000 Burnet Rd., Austin, TX 78758, mcneese@arlut.utexas.edu), Preston S. Wilson (Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Kevin M. Lee (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Following the early research on the Combustive Sound Source (CSS), described in the previous talk, additional development occurred in the 2000's and beyond, which included work on both the CSS and a new source, the Rupture Induced Underwater Sound Source (RIUSS). Development of the CSS in this era included maximizing the output of the source, investigating the use of CSS in arrays, and towing the source. The CSS was also deployed in surveys conducted during Shallow Water '06 and the Seabed Characterization Experiment. In part due to lessons learned during those surveys, the RIUSS was conceived and developed. The RIUSS functions by placing a rupture disk over an evacuated chamber and mechanically breaking the disk (either by striking on demand or via hydrostatic pressure) at a specified depth to initiate a volume collapse that produces an impulsive acoustic waveform. The development and use of each sound source will be provided along with high-speed underwater video, examples of signatures, and examples of the efficacy of the sources in ocean acoustics research. [Work Supported by ONR and NAVO].

1:40

2pEA3. Controls on the acoustic expression of buried and surface chemical explosions. Daniel Bowman (Sandia National Labs., Albuquerque, NM), Amrit K. Bal (Sandia National Labs., PO Box 969, M.S. 1377, Livermore, CA 94551-0969, akbal@sandia.gov), and Fransiska Dannemann Dugick (Sandia National Labs., Albuquerque, NM)

Surface and buried chemical explosions in the 1-50 ton TNT equivalent range can generate acoustic waves capable of traveling thousands of kilometers. Planned explosive campaigns in this size range have been used to develop acoustics-based yield determination techniques, investigate how acoustic propagation paths vary over short time scales, and examine how ground motion induces sound above the epicenter, among others. Here, we describe several test campaigns involving buried and surface chemical explosions. We discuss the combinations of explosive yield and burial depth that produce laterally propagating infrasound waves, as well as instances where diffuse sound from violent topographic shaking is observed. We also give an overview of a test series specifically aimed at investigating acoustic signal variation at tens to hundreds of kilometers range using co-located surface explosions fired tens of seconds apart. This will provide context for acoustic data interpretation from existing test series and aid in the design of future ones. [SNL is managed and operated by NTESS under DOE NNSA Contract No. DE-NA0003525.]

2:00

2pEA4. Energy density source levels of underwater explosive charges. Ross Chapman (Univ. of Victoria, University of Victoria, 3800 Finnerty Rd., Victoria, BC V8P5C2, Canada, chapman@uvic.ca)

This paper presents experimental measurements of energy density source levels of small explosive charges for high frequencies to 18 kHz. The experiments were carried out using Signals Underwater Sound (SUS) charges deployed at a deep water site in a coastal fiord north of Vancouver BC. The waveform of an underwater explosion is characterized by a high intensity shock wave signal followed by a series of bubble pulses of decreasing intensity. The bubble pulses dominate the SUS energy density spectrum at frequencies below 1 kHz, but at higher frequencies the spectrum is dominated by the characteristics of the decay envelope of the shock wave signal. The explosive material in SUS charges is packed in a thin cylindrical cavity, resulting in a non-uniform decay envelope with secondary pulses when detonated, and a non-spherical distribution of the radiated sound. The impact of the cylindrical charge shape on energy density source level was investigated experimentally. Data were recorded for SUS deployments that generated in-line and broadside sound propagation to the receiver with respect to the axis of the cylindrical charge. Experimental measurements indicate significant differences in the high frequency SUS energy density spectrum depending upon the SUS and receiver geometry.

2:20

2pEA5. Acoustic source characterization for chemical explosions in air. Keehoon Kim (Geophysical Monitoring Program, Lawrence Livermore National Lab., 7000 East Ave., L-103, Livermore, CA 94550, kim84@llnl.gov)

Chemical explosions in the air generate large pressure disturbances. These pressure waves propagate as non-linear shock waves near the source and transition into acoustic waves in the far-field. Since low-frequency acoustic waves propagate long distances without significant loss of energy, acoustic signals induced by explosions are often used to determine explosion energies of the events in terms of an explosion yield. In order to estimate the explosion energy accurately, the relationship between acoustic energy and explosion yield must be understood. However, explosion yields are often measured by non-linear shock waves in the near-field, and it is not clear how much acoustic energy accounts for the explosion energy. In addition, acoustic signals typically have lower-frequency contents than shock waves in the near-field, and hence frequency-dependent explosion energy should be understood to accurately infer explosion yields based on acoustic observations. In this study, we investigate the relationship between acoustic energy and explosion yield based on ground-truth explosion data. A standard acoustic source waveform will be determined by acoustic observations, and frequency-dependent energy will be explored for yield estimation. We will demonstrate that this frequency-dependent acoustic source characterization can improve the accuracy and confidence of explosion yield estimation.

2:40–2:55 Break

2:55

2pEA6. Modeling and development of a narrowband gas-combustion infrasound source. Chad M. Smith (Appl. Res. Lab., The Penn State Univ., State College, PA 16804, chad.smith@psu.edu) and Thomas B. Gabrielson (Penn State Univ., State College, PA)

An invaluable tool in characterization of any receiver, propagation path, or detection system, is a source with known and repeatable signal characteristics. This talk will discuss development of a well-controlled narrowband infrasound source with frequency capabilities over the 0.1 to 10.0 Hz band. Design of a transportable sound source within this band is a difficult engineering challenge. The simple source equation, which will govern any portable infrasound source due to the long wavelengths, shows this fundamental difficulty. As frequency decreases, volume displacement must increase by the squared inverse factor of frequency in order to maintain an equal pressure at equal range. To combat this, the authors have developed a source system that uses gas combustion to displace large volumes in the open atmosphere using the gaseous and highly heated combustion constituents of propane and atmospheric oxygen. Measurements have verified the capability of generating narrowband signals with reasonable signal-to-noise ratio (SNR) over the full band at close range. Signals at 1 Hz have been measured with reasonable SNR to ranges greater than 2 km. Development of the infrasound source prototype, experimental measurements, and modeling of the acoustic pressure output will be discussed.

3:15

2pEA7. Echo sounding with the shock wave generated by the implosion of an instrument pressure housing. Scott Loranger (Ocean Sci., Kongsberg Discovery, 86 Water St., Woods Hole, MA 02543, sloranger@whoi.edu), David R. Barclay (Oceanogr., Dalhousie Univ., Halifax, NS, Canada), and Michael Buckingham (Scripps Inst. of Oceanogr., San Diego, CA)

In December 2014, what was intended to be a passive-acoustic experiment at the Challenger Deep, suddenly became an active-acoustic experiment. Two free-falling, passive-acoustic instrument platforms, each with a suite of hydrophones, a CTD, a sound speed sensor and a glass-sphere pressure housing containing the electronics, were deployed from the R/V Falkor. One of the platforms imploded at a nominal depth of 9000 m. The highly energetic, broadband shock wave created by the implosion was recorded by the surviving platform. The shock wave reflected multiple times from the seafloor and the surface and the arrival times of the multiple reflections were used to obtain a highly constrained estimate of the Challenger Deep: 10 983 $\pm$ 6 m. We will discuss the signal processing methods used and propagated uncertainty in this highly constrained estimate. We will also discuss how this estimate of the depth of the deepest part of the ocean compares with other estimates of the depth of the Challenger Deep, including an update on estimates obtained following the publication of this study.

Contributed Papers

3:35

2pEA8. Dissecting recorded gunshot sounds. Steven D. Beck (Beck Audio Forensics, 14101 Hwy. 290 West, Bldg 1700, Ste. A, Austin, TX 78737, stevendbeck@alumni.rice.edu)

Gunshot audio recordings are composites of multiple sound sources, including blast, ballistic, mechanical, and environmental sounds. The existence of each sound and its distinguishing features depend on the source characteristics, the propagation environment, and the recording system. Controlled recordings show the situational dependence of these sound sources and the recording system. Knowledge of the physics and these dependences is critical in forensic analysis and the ability to dissect, separate, and identify sources in recorded gunshots. Multiple blast sources (muzzle blast, gas jet, mechanical impact, and primer explosion), ballistic sources (shockwave and subsonic flow), and environmental sources (reflections and terminal impact) will be shown from both controlled experiments and uncontrolled forensic recordings.

3:50

2pEA9. Source characterization of afterburning laboratory-scale jet noise with vector acoustic intensity. Michele L. Eggleston (Phys. and Astronomy, Brigham Young Univ., Brigham Young University, Provo, UT 84604, megglest@byu.edu), Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), Logan T. Mathews, Tyce W. Olaveson, Hunter J. Pratt (Phys. and Astronomy, Brigham Young Univ., Provo, UT), Ashwin Kumar M. G. (Mech. Eng., Virginia Tech, Blacksburg, VA), and Joseph W. Meadows (Mech. Eng., Virginia Tech, Blacksburg, VA)

This paper presents an intensity-based characterization of noise sources in a highly heated laboratory-scale jet with a 3-inch exit diameter. Run conditions had nozzle pressure ratios between 2.7 and 3.5 and temperature ratios of $\sim$7. Measurements were performed using a microphone scanning rig containing 8 four-microphone intensity probes. Probes were separated by 1.5 nozzle diameters and the microphones were spaced 0.5 nozzle diameters apart. Each scan mapped out over 3,000 measurement locations to produce an intensity field map. Vector intensity was processed over a 200 Hz to 20 kHz bandwidth using the phase and amplitude gradient estimator (PAGE) method [D. C. Thomas, *et al.*, *J. Acoust. Soc. Am.* 137, 3366–3376 (2015)]. Results show the dominant source region as a function of frequency and are compared to the acoustical holography source reconstruction of the T7-A afterburner engine condition [L. T. Mathews and K. L. Gee, AIAA J., in press (2024)]. Additionally, the intensity-derived source locations are connected to the jet's supersonic and subsonic flow regions. [Work supported by ONR Grant No. N00014-21-1-2069].

Session 2pED

Education in Acoustics, Engineering Acoustics, Musical Acoustics, and Physical Acoustics: Acoustics Education: A Potpourri of Classical and Unusual Materials and Demonstrations

Olivier Robin, Cochair
Université de Sherbrooke, 2500, bd de l'université, Faculté de Génie - Dpt Génie Mécanique, Sherbrooke, J1H1L2, Canada

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

Invited Papers

1:00

2pED1. Revisiting a classic experiment in violin acoustics as a classroom demonstration or laboratory activity. Andrew C. Morrison (Natural Sci., Joliet Junior College, 1215 Houbolt Dr., Joliet, IL 60431, amorriso@jjc.edu)

In 1982, Carleen Hutchins built a violin to conduct experiments largely to learn about the role of air cavity resonances on the overall sound produced by the violin. This violin was of a standard design except for 65 holes 5 mm in diameter drilled into the ribs. The holes could be selectively plugged with corks for experimentation. Carleen Hutchins was a skilled luthier, and this violin, playfully called "Le Gruyère," was a finely crafted musical instrument. For an introductory acoustics general education laboratory course, a version of Hutchins' Swiss cheese violin was created by drilling holes in the ribs of a factory-made student-model violin. Students in the course have the opportunity to replicate classic acoustics experiments or this violin can be used for class demonstration purposes. In this presentation, the advantages and challenges of implementing this activity as a laboratory or classroom demonstration are discussed.

1:20

2pED2. Flame suppression due to acoustic streaming by using time reversal. Jay M. Cliftmann (Phys. & Astronomy, Brigham Young Univ., BYU, Dept. of Phys. & Astron., N284 ESC, Provo, UT 84602, jc897@byu.edu), Brian E. Anderson (Phys. & Astronomy, Brigham Young Univ., Provo, UT), and Brian D. Patchett (Phys., Utah Valley Univ., Orem, UT)

Using high amplitude time reversal focusing to extinguish small flames is a novel demonstration of transient acoustic streaming. Typically, others have placed a low frequency acoustic source near the flame in order to generate a high enough particle displacement to extinguish it. In this study, we move the sources far from the flame and demonstrate that the time reversal method can focus transient waves to the flame. The peak acoustic overpressure level needed to extinguish a candle flame in free space is 191 dB peak when using a frequency range of 300 to 15000 Hz. This momentary focus causes acoustic streaming at the flame location after the focusing, which extinguishes the flame. By tracking the flame in high-speed video, we show the displacement of the flame due to the passing acoustic wave and subsequently due to the acoustic streaming. We further show that acoustic streaming extinguishes the flame, not the acoustic particle displacement.

1:40

2pED3. Teaching the basics of acoustics using smartphones. Olivier Robin (Université de Sherbrooke, 2500, bd de l'université, Faculté de Génie - Dpt Génie Mécanique, Sherbrooke, QC J1H1L2, Canada, olivier.robin@usherbrooke.ca)

This communication proposes a basis for offering short introductions to the basic principles in acoustics, like adding decibels and the behavior of Helmholtz or quarter-wavelength resonators. The proposed approach combine mobile and experiential learning approaches using smartphone sensors and applications. The objective of these mobile laboratories is to complement, but not replace, traditional laboratories, which allow for more advanced learning in particular topics. However, the proposed series of experiments can be carried out autonomously; students can explore scientific knowledge in real-life situations and without time or place constraints. Indeed, generic or everyday places can be considered, contrary to traditional laboratories that have to be held in dedicated or specific premises.

2:00

2pED4. Pedagogical strategies to enhance learning and awareness of acoustics within our engineering school community. Olivier Doutres (Mech. Eng., ETS (Ecole de technologie superieure), 1100, rue Notre-Dame Ouest, Montreal, QC H3C 1K3, Canada, olivier.doutres@etsmtl.ca), Kévin Rouard (Mech. Eng., ETS (Ecole de technologie superieure), Montreal, QC, Canada), and Maël Lopez (Mech. Eng., ETS (Ecole de technologie superieure), Montreal, QC, Canada)

Acoustics is taught at the École de technologie supérieure (ÉTS, Montreal, Canada) in a single advanced specialization course during the final year of the mechanical engineering bachelor's program. This course aims to equip students with the skills needed to measure and reduce noise based on the theoretical foundations of industrial acoustics and associated experimental techniques. The fact that the science of acoustics is not well-known among engineering students, coupled with the optional nature of this course, results in an average enrollment of only about thirty students each year (across two distinct sessions), a number further reduced since 2020 due to the unfortunate impact of the pandemic. Paradoxically, Quebec lacks engineers trained in this discipline and often recruits them from abroad. This presentation will aim to showcase various strategies and pedagogical tools that have been used and experimented with in recent years (e.g., flipped classroom, in-class experiments, cellphone measurements, community service-oriented semester projects). The goal is to ensure the quality and enjoyment of student learning and to contribute to raising awareness about acoustics and noise-related issues within the ÉTS community.

2:20

2pED5. Educational uses of a microflown p-u probe. Daniel A. Russell (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg, University Park, PA 16802, dar119@psu.edu)

Two of the most useful transducers I have ever purchased for my research lab are a B&K Type 8001 mechanical impedance head and a Microflown p-u matchstick probe acoustic impedance head. This talk will illustrate several educational demonstrations one can perform using a Microflown p-u probe. One demonstration will show the transition from the near-field of a spherical source (where pressure and velocity are in quadrature) to the far-field (where pressure and velocity are in phase). Another demonstration will illustrate the relative phase differences for pressure and velocity upon reflection from an open or closed end of a pipe. A third demonstration will show how a measurement of acoustic impedance can be used to determine the undamped natural frequency of a Helmholtz resonator, which in turn may be used to accurately predict the length correction at the open end of the resonator mouth. Additional demonstrations may be shown if time permits.

2:40–2:55 Break

Contributed Papers

2:55

2pED6. Converting acoustical demonstrations into in-class experiments. Kurt R. Hoffman (Phys., Whitman College, 345 Boyer Ave., Hall of Sci., Walla Walla, WA 99362, hoffman@whitman.edu)

Teaching acoustical principles to non-science students invites extensive use of classroom demonstrations to help clarify concepts. Ideally, a separate laboratory course opens up many more opportunities for active learning for improved student understanding. However, observing and hearing a demonstration is distinct from performing the activity directly. Moreover, the addition of a laboratory course impacts departmental teaching load and may impact enrollment. In this paper I will discuss laboratory and demonstration activities that have been converted into group activities that are implemented as a classroom activity. The opportunity to add these elements to the course hinge on being able to build or purchase a large number of replicas of equipment needed for the planned activities. The two examples I will discuss in detail are the use of two stringed monochords and pvc trumpets. In addition, I will share a simple mechanical demonstration based on vibrating bars to visualize the location dependent resonances of the basilar membrane.

3:10

2pED7. ROC curve measurements using euro-rack modules (and some DIY). Murray S. Korman (Phys., U.S. Naval Acad., Phys. Dept., U.S. Naval Acad., 572 C Holloway Rd., Annapolis, MD 21402, korman@usna.edu) and Kevin L. Mcilhany (Phys., U.S. Naval Acad., Annapolis, MD)

Acoustic signal processing projects encourage DIY electronics as well as hardware designed to expose principles found either in nature or from man-made sources. Searching within the Eurorack format of the modular synthesis community, functional equivalents to more common DIY units are identified. Some signal generators and measurements are still underrepresented commercially and require DIY Eurorack modules to be built. One goal is to use existing Eurorack components where DIY modules are designed as necessary in this format. A Receiver Operating Characteristic curve (ROC) is calculated for three cases using analog energy detection and probability density function (pdf) circuitry. The hypothesis for noise (N) is a fixed Gaussian noise distribution. The three hypotheses for the signal plus noise (S+N) have signals characterized as; a constant sinusoid with random phase, a broadband unknown Gaussian signal, and an unknown narrow band Gaussian signal utilizing a narrow band carrier frequency. Our classroom demonstration involves two loudspeakers, one for noise, the other for signal. The detected microphone's voltage is bandpass filtered, amplified, squared, integrated and then fed into the pdf circuitry where pulses are counted for 4 seconds in each voltage window. Increasing the signal amplitude will increase the detection index.

3:25

2pED8. Design and construction of an in-air sonar for signal processing education. Kyle S. Dalton (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, ksd5377@psu.edu), Robert Nelson, Steven J. Todd, Jonah Warner, Lily R. Hetherington, John A. Case, Mitchell J. Swann (Graduate Program in Acoust., Penn State Univ., State College, PA), and Constantinos S. Kandias (Aerosp. Eng., Penn State Univ., University Park, PA)

Sonar is an acoustic sensing method commonly used in underwater imaging and navigation. Students are often introduced to sonar theory, but few gain hands-on experience due to the cost and logistical difficulties associated with in-water experimentation. Alternatively, a sonar system can be designed for in-air use to mitigate some of the practical and operational challenges of acquiring sonar field data. Inspired by previous in-air sonar projects, the Pennsylvania State University student chapter of the Acoustical Society of America acquired $5,000 to construct an in-air sonar for exploring signal processing and mechanical design concepts. For a cost-effective solution, consumer audio equipment and additive manufacturing were leveraged. This presentation will walk through the sonar design and build process, discuss the project's current status, and present initial results. Plans for project development and potential inclusion into the curriculum will be discussed. Advice will be offered to those who may want to pursue similar projects.

3:40

2pED9. Demonstration of nonlinear tuning curve vibration response using a homemade analog sweep spectrum analyzer. John Paulenich III (Phys., U.S. Naval Acad., 641 Wayward Dr., Annapolis, MD 21401, jpaulenich@gmail.com) and Murray S. Korman (Phys., U.S. Naval Acad., Annapolis, MD)

A super-heterodyne sweep spectrum analyzer has been designed (using analog circuitry), built and tested to cover the 5 to 1000 Hz range. The analyzer employs a linear ramp voltage versus time, driving a voltage-controlled oscillator VCO that generates a sweep from 32.768 kHz to 33.768 kHz in 70 s. An analog AD734 multiplies the VCO signal by a sinusoidal signal (generated by a watch crystal 32.768 kHz oscillator). The "mixer" signal is low-pass filtered and amplified to generate a 5-1000 Hz swept tone using a 2-in. speaker. The accelerometer (mounted on a thin circular clamped acrylic plate) vibration response involves multiplying this signal by the VCO, then filtering this "mixer" signal using a 4-stage watch crystal ladder filter with a 1 Hz bandwidth. This signal is squared and low-pass filtered to generate the time (converted to frequency) vs. mean-squared voltage response on an oscilloscope. In the demonstration, 300 g of 6 mm diameter glass beads are supported by the 11.4 cm diam, 3.2 mm thick plate and upper rigid cylindrical wall column. The nonlinear tuning curve vibration response is recorded for various incremental drive amplitudes to demonstrate that the tuning curve resonant frequency significantly decreases with increasing drive amplitude.

3:55

2pED10. Exploring comb filtering effects in the recording studio: A lesson on room acoustics. Wesley Bulla (Audio Eng., Belmont Univ., Nashville, TN), Lisa LaFountaine (Belmont Univ., 1900 Belmont Blvd, Nashville, TN 37212, Lisa.LaFountaine@bruins.belmont.edu), and Raymond Plasse (Belmont Univ., Nashville, TN)

A common practice during a live music-recording session is to position microphones, so-called, "room mics" to capture the ambient characteristics of the recording space. An in-class, hands-on experiment was conducted at Belmont University's Historic Columbia Studio A in order to measure and observe the effect of comb filtering on the signal captured by an on-axis room microphone placed at various heights from the floor. A total of seven positions were recorded, spanning from just above the floor to twelve feet in the air. Sine sweep measurements and musical samples from a live virtual jazz-quartet were taken at each position. Students were then asked to listen back and discuss their preferences for each microphone height, comparing their observations to the matching frequency response analysis. A majority of students enjoyed the recording with the microphone positioned at the floor corresponding with the least amount of comb filtering. Although listener preference data was not collected, this experiment demonstrated the use of interdisciplinary examples as an educational tool in the field of acoustics.

4:10

2pED11. Building a guitar amplifier: Accessible, engaging, and project-based audio instruction for students of all academic backgrounds. Benjamin R. Thompson (Elec. & Comput. Eng., Univ. of Rochester, 120 Trustee Rd., Rochester, NY 14620, bthomp23@ur.rochester.edu) and Sarah R. Smith (Elec. and Comput. Eng., Univ. of Rochester, Rochester, NY)

A challenge in designing first-year engineering curriculum is meeting the needs of a group of students with diverse academic backgrounds. Educators in the field of audio and acoustical engineering are often tasked with simultaneously preparing majors to be successful in their subsequent coursework while delivering a course that is both accessible to and exciting for *all* students. At the University of Rochester, we offer a survey course that introduces students to fundamental concepts in acoustics, signals, and audio electronics in which we leverage students' existing experiences with sound and music in a project-based environment. Students complete a series of laboratory exercises where they interact with the science of audio by building a guitar amplifier circuit from scratch. Analyzing a schematic, soldering components, solving equations, seeing waveforms, and HEARING the results of their work, allows all students to engage in the content more deeply than any of these activities in isolation. In this presentation, we will share the specifics of our approach, and provide open-source curriculum in the form of lab manuals, supporting files, and sample PCBs.

4:25

2pED12. Teaching concepts of acoustics in air—Part 3, reverberation. William J. Gastmeier (HGC Eng., 12 Roslin Ave. South, Waterloo, ON N2L2G5, Canada, bill@gastmeier.ca)

This paper has been written to participate in the "Education in Acoustics" session at the 2024 joint conference of the Canadian Acoustical Association and the Acoustical Society of America. It is written to build on Parts 1 and 2 of this series and similarly contains materials extracted from 30 years of teaching to Architects at the University of Waterloo and Dalhousie University in Halifax. Part 1 dealt primarily with sound propagation in air and the concepts of longitudinal wave motion, speed, frequency and wavelength and related effects which relate to what we perceive as pitch. Part 2 expanded on those concepts by discussing superposition and the definition and measurement of sound pressure, decibels and the decibel scale and how to manage decibels, all of which we perceive as loudness. Part 3 deals with reverberation, which is arguably the most important aspect of how sound propagates in indoor spaces. It affects both our ability to communicate and how we experience and perceive the quality of the acoustical environment. Practical demonstrations are provided to enhance learning of the concept of Reverberation. These include "hands on" demonstrations including physical experimentation, an audio demonstration and written materials to advance the concept.

Session 2pMUa

Musical Acoustics: Winds Instruments II

Gary Scavone, Cochair
Music Research, McGill University, 555 Sherbrooke Street West, Montreal, H3A 1E3, Canada

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

Invited Papers

1:00

2pMUa1. Electronic wind instruments for mobility-restricted musicians. David S. Whalen (EMPAC, Rensselaer Polytechnic Inst., Troy, NY), Henry Lowengard (AUMI Consortium, Kingston, NY), and Jonas Braasch (School of Architecture, Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, braasj@rpi.edu)

One of the earliest applications of analog music synthesizers was the simulation of orchestral instruments. Electronic wind instrument sounds were played through piano-like keyboards or interfaces with breath controllers that mimicked physical wind instruments. Synthesizers are also often the best alternative for artists with disabilities to play a musical instrument. This talk focuses on methods and experiences the authors were involved with to design electronic alternatives to wind instruments for people who lost control of their arms. The Jamboxx is a USB controller that was specifically designed as a head-only interface using a breath controller to simulate airflow but also operates as a suck/puff device to activate menus and buttons, a slider to manipulate pitch, and a tilt-sensor to control expression parameters. The Jamboxx also addressed musicians seeking a harmonica-type interface to augment their guitar performance. The AUMI instrument is a touchless device based on camera tracking. This talk will address different sound synthesis methods to achieve realistic wind instruments, the ability to integrate these instruments with other off-the-shelf products (e.g., VST plugins), and special requirements for artists with special needs to play these instruments without too much fatigue. [Work supported by the Craig H. Neilsen Foundation.]

1:20

2pMUa2. A theoretical framework for estimating wind instrument bell reflection and transmission functions. Tamara Smyth (Music, UC San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0021, trsmyth@ucsd.edu)

This work examines the bell reflection/transmission by using a combination of a piecewise waveguide model and an open-end reflection (along with the assumed amplitude-complementary transmission) function, modeled as a first-order shelf filter. The wind instrument bell is first modeled by deriving the chain scattering matrix for a piecewise cylindrical model, constructed using reflection coefficient vectors that have a useful structure of being interleaved with zeros (due to the round-trip delay of two samples in each section). The matrix is then used to yield the instrument (or instrument bell) transfer function which is shown to have both poles and zeros if the open-end reflection/transmission is made frequency dependent. If the open-end is modeled as a first-order filter and incorporated into the instrument model using matrix convolution, filter coefficients can be estimated from values in the reflection coefficient vectors that would otherwise be zero.

1:40

2pMUa3. Resonators in free reed instruments: Good, bad, and unusual effects. James P. Cottingham (Phys., Coe College, 1220 First Ave. NE, Cedar Rapids, IA 52402, jcotting@coe.edu)

Western free reed instruments such as the accordion and the harmonica do not typically use pipe resonators. Yet even without pipes there are two sorts of resonators employed. In the case of mouth blown instruments, free-reed or not, the resonances of the players vocal tract are quite significant. Less often recognized as resonators are the reed chambers necessary to provide a secure mounting for the reed and to properly direct the airstream. The acoustical properties of these may or may not be beneficial. Because of their different reed construction, the Asian free-reed mouth organs require pipe resonators in to function. This paper will present examples of ways in which reed chamber and pipe resonances can enhance or interfere with tone production, as well as some interesting special effects produced by intentional mismatches between reed and resonator.

2:00

2pMUa4. The effects of vowel, pitch, and dynamics on the emotional characteristics of the Mezzo-Soprano, Countertenor, and Baritone Voices. Bing Yen Chang (Comput. Sci. and Eng., Hong Kong Univ. of Sci. and Technol., Hong Kong, Hong Kong), Hiu Ting Chan (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., TClear 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)

Previous research on the soprano, alto, tenor, and bass voices have shown that their emotional characteristics change with different vowel, pitch, and dynamics. This work continues the investigation with the mezzo-soprano, countertenor, and baritone voices. Listening tests were conducted whereby listeners gave absolute judgements on the voice tones over ten emotional categories, with the data analyzed using logistic regression. High-arousal categories were stronger for loud tones, whereas low-arousal categories were stronger for soft tones. The categories Happy, Heroic, Romantic, and Comic had an upward trend across the pitch range, whereas Calm, Mysterious, Shy, and Sad had a downward trend. Angry and Scary had different trends among the voices. For most categories, all five vowels were mostly similar in terms of emotional expressiveness, with exceptional cases for the baritone voice. Overall, pitch had the strongest effect and was almost twice as strong as dynamics and vowel, with dynamics slightly stronger than vowel. Vowel O had the largest strength-of-expressiveness overall, closely followed by vowels U and A, and finally vowels I and E last. These results give a quantified preliminary perspective on how vowel, pitch, and dynamics shape emotional expression in the voices across a wide range of voice types.

Contributed Papers

2:20

2pMUa5. Study on the source of overtone series in a harmonica note. Kuiliang Li (Tufts Univ., 640 BOSTON Ave., Unit 509, Medford, MA 02155-1330, likuiliang123456@gmail.com) and Chris Rogers (Tufts Univ., Medford, MA)

The harmonica reed is a cantilever beam and therefore does not produce integer harmonics. High-speed camera video of the reed and vibrometer measurements both demonstrate this expected behavior at the reed tip when being played. However, the acoustical recording shows an integer overtone series with large amplitudes ($<$10 dB from the fundamental) all the way past 22 kHz. To understand this phenomenon, we eliminated resonant cavities and other possible acoustic feedback systems by plucking the reed with only the reed plate attached. The acoustical and vibrometer data showed the same discrepancy in this case, but only when the reed enters the reed slot in the reed plate. The overtones don't show up when the reed is vibrating completely away from the reed slot. In this paper, we will present these data and show how this phenomenon could be a result of vorticity produced by the sharp edge of the plate and the small air layer gap between plate and reed.

2:35–2:50 Break

2:50

2pMUa6. Abstract withdrawn.

3:05

2pMUa7. One-dimensional acoustic modeling of the şimşal considering the external Interaction of the open finger holes. Diako Kaboodi (Faculty of Graduate and Postdoctoral Studies, Laval Univ., 2345, allée des Bibliothèques, QC, QC G1V 0A6, Canada, diako.kaboodi.1@ulaval.ca), Denis Laurendeau (Faculty of Sci. and Eng., Laval Univ., Québec, QC, Canada), Gary Scavone (Music Res., McGill Univ., Montreal, QC, Canada), and Aaron Liu-Rosenbaum (Faculty of Music, Laval Univ., Québec, QC, Canada)

A one-dimensional model in Python was developed to analyze the acoustic behavior of the Kurdish wind instrument, the şimşal. This model was developed using the Transmission Matrix Method (TMM), which was refined by taking into account the external interactions between the open tone holes. This research focuses on the acoustic resonator part of the şimşal with the lattice of finger holes in different cross-fingering patterns. The results were validated by measurements using an impedance probe, CapteurZ, which was developed by LAUM[1] and CTTM[2] The results are in good agreement with the measurement and despite the possible sources of error, which are discussed, the deviations are within an acceptable range. [1] The Acoustics Laboratory of the University of Le Mans [2] Le Mans Technology Transfer Center.

3:20

2pMUa8. Acoustic impedance measurement head design and evaluation. 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 Res., McGill Univ., Montreal, QC, Canada)

Acoustic input impedance measurements are a useful tool for characterizing the behavior of a wind instrument. Methods for measuring acoustic input impedances include the two-microphone three-calibration (TMTC) technique (Gibiat and Laloë, 1990) and the multi-microphone method (Jang and Ih, 1998). We present and evaluate the design of a new impedance measurement head based on the multi-microphone method. We will discuss practical aspects such as impedance head calibration, measurement stimulus, and leak detection.

3:35

2pMUa9. Numerical modeling of a saxophone and comparison with experimental measurements. Marie L. Jeanneteau (Res. and development, Henri Selmer Paris, ICA, 3 rue Caroline Aigle, Toulouse, Occitanie 31400, France, marie.jeanneteau@insa-toulouse.fr), Paul Oumaziz, Jean-Charles Passieux, Vincent Gibiat (Metrology Identification Control and Monitoring, Clément Ader Inst., Toulouse, Occitanie, France), and Jonathan Cottier (Res. and development, Henri Selmer Paris, Paris, France)

Saxophones making has long relied on craftsmanship, associated with an empirical knowledge of their acoustic functioning. The design process is now mainly based on the study of the input impedance, accessible experimentally or computationally. The numerical approaches are currently limited either by the accuracy (analytical models) or by the computation time (numerical resolution of PDEs). The challenge is to propose a numerical method combining both, to access the acoustic pressure and velocity fields in the instrument. Our objective was to develop a high-performance parallel numerical tool based on FEM to model accurately (a few cents on the resonances) the acoustic behavior of the resonator under playing conditions. The computation time was reduced by an order of magnitude compared to the brute force approach, using different modeling strategies specific to wind instruments: First, the modeling of the visco-thermal losses at the walls; then, the reduction of the size of the linear systems and finally, the reuse and model reduction to limit the effect of the multiple resolutions imposed by the wide frequency range of interest. The model is validated by comparison with experimental measurements. An impedance measurement system allowing acquisitions under controlled flow has been developed to reproduce realistic playing conditions.

3:50

2pMUa10. Sounding saxophones like flutes. Jonas Braasch (School of Architecture, Rensselaer Polytechnic Inst., School of Architecture, 110 8th St., Troy, NY 12180, braasj@rpi.edu)

The saxophone can be modified with alternative non-single-reed tone generators, including double and free reeds and brass-instrument mouthpieces. It can also be played as a rim flute on the neck, which is a known extended technique for the saxophone. Flutes differ fundamentally from the other tone-generator types as they do not act like a valve, and their flutes are so-called open-open resonator instruments, while reeds and the lips in brass instruments effectively close one end of the resonator. As a practical consequence, all open-close hole key combinations result in fundamentally different pitches known from the regular saxophone. The fact that the saxophone is conically shaped, while the flute is typically cylindrical or inverse conical, further complicates the matter. Approaches to playing the saxophone as flute using alternative fingering combinations will be discussed along with measures of pitch accuracy, timbral, and level balance. While the achievable range aligns with many orchestral wind instruments, complex cross-fingerings make it difficult to play fast chromatic lines. The large bore of the saxophone gives the flute sound a dark character, more like a Native American flute or shakuhachi than a Western concert flute or recorder.

2p TUE. PM

Session 2pMUb

Musical Acoustics: Native American Flute Concert

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

Chair's Introduction—4:30

Invited Paper

4:35

2pMUb1. A concert by Theresa "Bear" Fox. Theresa. Fox (Mohawk Nation - Akwesasne, Akwesasne, ON, Canada), Jonas Braasch (School of Architecture, Rensselaer Polytechnic Inst., School of Architecture, 110 8th St., Troy, NY 12180, braasj@rpi.edu), and Paula Conlon (School of Music, Univ. of Oklahoma, Ottawa, ON, Canada)

Theresa "Bear" Fox is a prolific singer and songwriter from the Mohawk Nation at Akwesasne. During this concert, she will present songs from her eight CD albums highlighting Akwesasne culture. The ancestral land of the Akwesasne community of about 12,000 people spreads across both banks of the St. Lawrence River and intersects Quebec, Ontario, and New York State. Bear Fox's music is deeply rooted in her living indigenous tradition. Following Akwesasne culture, Bear Fox's songs often have a specific function. Some songs are healing songs, and others are provided as gifts to family members. Bear Fox writes her lyrics in both Mohawk language and English, which will give the audience the opportunity to listen to the authentic sound of the Akwesasne music tradition and directly experience the meaning of words in the larger context of Native American songs. During the concert, Bear Fox will explain how each song has a story that is grown from a planted seed. Songs are often written to honor Mother Earth, mothers, water, wind, Grandmother Moon, and our newborns. Bear Fox believes in the positive message that can come from songs, which is a central part of the Mohawk culture.

Session 2pNSa

Noise, Computational Acoustics and Psychological and Physiological Acoustics: Advanced Air Mobility Noise: Noise from New Air Transportation in Urban and Underserved Communities II

Matthew Boucher, Chair
NASA Langley Research Center, 2 N. Dryden St., M/S 463, Hampton, VA 23681

Chair's Introduction—1:00

Invited Papers

1:05

2pNSa1. Reduced-order equivalent source modeling of aeroacoustic sources. Christian Dreier (Inst. for Hearing Technol. and Acoust., RWTH Aachen Univ., Kopernikusstrasse 5, Aachen 52074, Germany, christian.dreier@akustik.rwth-aachen.de), Xenia Vogt, Wolfgang Schröder (Inst. of Aerodynamics, RWTH Aachen Univ., Aachen, Germany), and Michael Vorlaender (Inst. for Hearing Technol. and Acoust., RWTH Aachen Univ., Aachen, Germany)

The acoustic far-field prediction of high-fidelity aerodynamic simulation data is still not accurately possible when accounting for the surface reflections from an aircraft's fuselage or wings. By presenting a general method for order-reduced equivalent source modeling of aeroacoustic sound sources, this work is bridging the gap between high-fidelity aerodynamic simulations and its integration to auralization frameworks. At the example of a numerically simulated jet, this study shows the computation of an equivalent source model with minimal complexity by means of spherical harmonic (SH) coefficients. The minimal source extension, in which all acoustic sources of a flow field are confined, is computed by using spherical Hankel extrapolation of sound pressure data from virtual concentric microphone arrays. The result of the SH transform shows that the dominant energy contribution can be associated to nine elementary sound sources. The resulting equivalent source model of jet noise provides a convenient date interface between large-scale computational fluid dynamics and acoustics simulations.

1:25

2pNSa2. Acoustic measurements of full-scale electric vertical takeoff and landing aircraft. Shane V. Lympany (Blue Ridge Res. and Consulting, 29 N Market St., Ste. 700, Asheville, NC 28801, shane.lympany@blueridgeresearch.com) and Juliet A. Page (Blue Ridge Res. and Consulting, Asheville, NC)

The Advanced Air Mobility industry is progressing rapidly towards the goal of revolutionizing transportation by connecting urban and underserved communities with electric vertical takeoff and landing (eVTOL) aircraft. To achieve widespread use, eVTOL aircraft must generate acceptable noise in surrounding communities. Many eVTOL aircraft currently under development have novel configurations of rotors, propellers, and wings that may produce rotor-rotor and rotor-airframe interaction noise at certain flight conditions. Acoustic measurements of full-scale eVTOL aircraft are critical to understand the complex noise-generation mechanisms and to validate aeroacoustic models for these novel aircraft designs. In this presentation, we present acoustic measurements of Hexa, an 18-rotor, single-passenger eVTOL aircraft developed by LIFT Aircraft, Inc. We apply acoustic source characterization techniques using simultaneous acoustic measurements and vehicle telemetry data to provide a better understanding of the noise-generation mechanisms. One such analysis technique is the Vold-Kalman filter, which we apply to auralize the tonal waveforms of individual rotors. Finally, we discuss plans for upcoming acoustic measurements of a full-scale, multi-passenger eVTOL aircraft. These measurements and analyses provide the data and tools required to better understand the noise-generation mechanisms and to validate aeroacoustic models for full-scale eVTOL aircraft

1:45

2pNSa3. Onboard sound emission measurements of unmanned aerial vehicles for psychoacoustic experiments. Jonas Jaeggi (Lab. for Acoust. / Noise Control, Empa, Swiss Federal Labs. for Material Sci. and Technol., Ueberlandstrasse 129, Dübendorf 8600, Switzerland, jonas.jaeggi@empa.ch), Jonas Meister, and Reto Pieren (Lab. for Acoust. / Noise Control, Empa, Swiss Federal Labs. for Material Sci. and Technol., Dübendorf, Switzerland)

With the prospect of small UAVs being more commonly used for tasks like surveillance or parcel transport within inhabited areas, demand for noise regulations arises. The noise assessment of UAV operations can be investigated through laboratory listening experiments using auralized sound stimuli. Sound recordings of source signals for dynamic flight operations have to be performed outdoors where measurements with stationary microphones impose uncertainties in backpropagation and suffer from bad signal-to-noise ratio due to the weak source and the presence of ambient sounds. To overcome these difficulties, this contribution presents an approach for acquiring UAV source signals using onboard microphones. A lightweight setup for time synchronous recordings of sound using MEMS microphones, the UAV position and the rotational speed of each rotor was developed and applied to two quadcopters of 0.9 and 6.3 kg mass.

Source directivity and propagation effects were added to the source signals to obtain clean sound pressure signals at virtual listener positions. The resulting stimuli were successfully used in a listening experiment on short-term noise annoyance.

2:05

2pNSa4. Sound from unmanned aerial vehicles in ambient acoustic environments: Influences of context, vehicle characteristics and flight operations on human perception and response. Michael J. Lotinga (Acoust. Res. Ctr., Univ. of Salford, Newton Building-Salford, Salford M5 4BR, United Kingdom, m.j.lotinga@edu.salford.ac.uk), Marc C. Green (Acoust. Res. Ctr., Univ. of Salford, Salford, United Kingdom), and Antonio J. Torija Martinez (Acoust. Res. Ctr., Univ. of Salford, Manchester, United Kingdom)

The potential opportunities for unmanned aerial vehicles (UAVs) to offer wide societal benefits are accompanied by risks of introducing novel sources of community noise impact. Accordingly, it is important to develop a greater understanding of the subjective perception and response to UAV sound. Suitable evidence-based strategies for managing the potential risks to public health and wellbeing can then be devised in the form of flight path optimisation tools, to support initial technology deployment efforts. The 'REFMAP' project aims to develop these tools, which will include a component enabling noise constraints to be considered. In support of this objective, a laboratory listening experiment has been undertaken to study psychoacoustic aspects of UAV sound exposure. The experiment design incorporated influences of contextual auditory and soundscape factors, by embedding spatially rendered UAV sound events within ambisonic recordings of urban acoustic environments. The UAV renderings comprised varying flight operations and numbers of flight events. The experiment was focussed on determining both perceptual noticeability and affective responses to UAV sound, including noise annoyance. Initial results from the experimental data analysis are discussed, in the context of future planned work within the REFMAP research project.

2:25

2pNSa5. A psychoacoustic test on the effect of masking on annoyance to urban air mobility vehicle noise. Matthew Boucher (Structural Acoust. Branch, NASA Langley Res. Ctr., 2 N. Dryden St., M/S 463, Hampton, VA 23681, matthew.a.boucher@nasa.gov), Andrew Christian (Structural Acoust. Branch, NASA Langley Res. Ctr., Hampton, VA), Tyler Tracy (Structural Acoust. Branch, NASA Langley Res. Ctr., Cambridge, MA), Siddhartha Krishnamurthy, Kevin Shepherd (Structural Acoust. Branch, NASA Langley Res. Ctr., Hampton, VA), Durand Begault (Human Systems Integration Div., NASA Ames Res. Ctr., Moffett Field, CA), and Stephen A. Rizzi (NASA Langley Res. Ctr., Hampton, VA)

Urban Air Mobility (UAM) vehicles have a large range of designs and configurations that lead to new noise characteristics and potentially different perceptual responses when compared to traditional aircraft. In addition, UAM vehicles are expected to operate around and within densely populated regions where the presence of ambient background noise is often present. Strategically, this can be leveraged to inform vehicle design and operations to partially or completely mask UAM noise, allowing for mitigation of negative responses and an increased number of allowed operations. A psychoacoustic test was conducted to investigate how masking effects can influence the annoyance response to a low frequency harmonic tone complex (80-320 Hz). To do this, five test subjects compared their annoyance response to the low frequency tonal noise with a higher frequency broadband noise (10 dB down bandwidth between 300 and 2000 Hz), with and without a masking noise present. Detection thresholds were also measured for both sounds to help fit a model to the data. Although the effect of masking on annoyance is complex, results indicate that for some individuals, masking leads to a lower annoyance than the sound level alone would predict.

Session 2pNSb

Noise and Architectural Acoustics: Soundscape—Focus on Applications II

Brigitte Schulte-Fortkamp, Cochair
HEAD Genuit Foundation, Ebert Straße 30 a, Herzogenrath, 52134, Germany

Bennett M. Brooks, Cochair
Brooks Acoustics Corporation, 49 N. Federal Highway, #121, Pompano Beach, FL 33062

Contributed Papers

1:00

2pNSb1. Harmony in confinement: Exploring acoustic and thermal comfort in the Sandiaoling eco-friendly tunnel. Chan N. Truong (Dept. of Architecture, National Taiwan Univ. of Sci. and Tech., 43 Keelung Rd. Sec. 4, Taipei 106, Taiwan, M11213805@mail.ntust.edu.tw), Clarissa Averina, Gabriela Niederberge, Juliana Manuela Muet, Nikita Grace Manullang, Stijn Zeger van Brug, Roxana Ghadiri, Ni Made Putri Indriyani, Phoa Angela Grace Wibowo, Khaing Thinzar, Tuan Sanh Diep, Shiang-I Juan, and Lucky S. Tsaih (Dept. of Architecture, National Taiwan Univ. of Sci. and Tech., Taipei, Taiwan)

The Sandiaoling Eco-friendly Tunnel in New Taipei City, Taiwan, a renowned tourist destination for foot and bicycle travel, features intentionally designed elements such as illumination, sustainable materials, and low-carbon construction. In contrast to open public spaces, the tunnel's conditions are confined by a closed atmosphere, limited direction, and unclear vision. However, the consistent temperature and intermittent natural sounds within the tunnel could contribute to a sense of comfort. To evaluate acoustic comfort at the site, the study utilizes the soundscape study method outlined in ISO 12913. This involves documenting sound sources, paths, and levels, along with a questionnaire to understand visitor preferences. Additionally, infrared thermography images, relative humidity, and wind speed for each measured position are documented and analyzed to explore potential correlations between acoustic and thermal comfort. Preliminary results indicate that the pleasantness level at the site during measurements is not necessarily affected by thermal conditions. Visual and environmental freshness in the tunnel appear to be dominant factors for pleasantness despite the constant occurrence of manmade sounds. A comprehensive understanding of the relationship between thermal conditions and the acoustic environment would necessitate year-round data collection, providing valuable insights into the factors influencing the overall comfort experience for visitors.

1:15

2pNSb2. Unveiling the impact of auditory salience and sources identification on perceived pleasantness in environmental soundscapes. Nicolas Misdariis (Sound Percept. and Design Group, Ircam STMS Lab, 1, Pl. Igor Stravinksy, Paris F-75004, France, nicolas.misdariis@ircam.fr), Baptiste Bouvier (Sound Percept. and Design group, Ircam STMS Lab, Paris, France), Catherine Marquis-Favre (Univ Lyon, ENTPE, Ecole Centrale de Lyon, CNRS, LTDS, UMR5513, Vaulx-en-Velin, France), and Patrick Susini (Sound Percept. and Design group, Ircam STMS Lab, Paris, France)

On the basis of initial results related to modulations of auditory salience—the ability to capture attention—by timbre attributes (especially, brightness and roughness), the present study investigates auditory salience as a predictor of perceived pleasantness in environmental sound scenes. A new paradigm was set up to measure continuous pleasantness while observing the impact of specific events, in a corpus of 11 various soundscapes. Specific events were found to affect perceived pleasantness, in a source-dependent direction in line with the literature. In addition, causality analyses examined the impact of temporal salience predictions on pleasantness ratings, along with other indices (equivalent sound level, loudness, or timbre attributes). They showed that salience was the best predictor of pleasantness, ahead of loudness, level or brightness and roughness. Nonetheless, the last two turned out to play, as well, a significant role in the pleasantness percept. Therefore, auditory salience is confirmed as a relevant indicator for assessing the perception of soundscapes, and the identification of sound sources that compose them is also confirmed to be crucial in this assessment.

1:30

2pNSb3. Towards soundscape management of protected natural areas using the ISO 12913: A field study. Tin Oberman (Inst. for Environ. Design and Eng., Univ. College London, Central House, 14 Upper Woburn, London, United Kingdom, t.oberman@ucl.ac.uk), Simone Torresin (Dept. of Civil Environ. and Mech. Eng., Univ. of Trento, Trento, Italy), Arianna Latini (Dept. of Construction, Civil Eng. and Architecture, Università Politecnica delle Marche, Ancona, Italy), Giacomo Gozzi (Silenzi in Quota, Trento, Italy), Francesco Aletta (Univ. College London, London, United Kingdom), and Jian Kang (Inst. for Environ. Design and Eng., Univ. College London, London, United Kingdom)

Human perception of soundscapes in protected natural areas like national parks is crucial for their protection. At popular scenic spots, visitors themselves often contribute to noise pollution. Decibel-based systems (such as LAeq or Lden) do not fully explain human reactions to this phenomenon, necessitating a more holistic approach to allow for an effective management strategy. A mixed-methods soundscape approach based on the ISO 12913 series, developed mostly in urban soundscape studies, was tested in four protected natural ex-urban areas in the Dolomites (Italy) and Cairngorms (United Kingdom). During five soundwalks (7-12 km long), conducted by adopting the Method A of ISO/TS 12913-2, a total of 443 questionnaire responses were gathered across 28 evaluation points, alongside corresponding binaural measurements. A range of acoustic environments as quiet as LAeq = 31 dBA and as loud as LAeq = 76 dBA were observed, eliciting perceptions ranging from very calm to chaotic. A Linear Mixed-Effects Model was computed to analyse the impact of sound source dominance, psychoacoustic and environmental acoustic indices on perception. Presence of human sounds proved to be a major factor driving the perception of chaotic soundscapes.

1:45

2pNSb4. Relationship between greenery and the health of Madrid's citizens. Guillermo Rey-Gozalo (Física Aplicada, Universidad de Extremadura, Av. Universidad s/n, Cáceres 10003, Spain, guille@unex.es), Juan Miguel Barrigón Morillas, David Montes González, Rosendo Vílchez-Gómez (Física Aplicada, Universidad de Extremadura, Cáceres, Spain), Carlos Iglesias-Merchán (Universidad Politécnica de Madrid, Madrid, Spain), Silvia Merino-de-Miguel (Universidad Politécnica de Madrid, Madrid, Spain), Pierre Aumond (Université Gustave Eiffel, Nantes, France), Laura Muñoz-Bermejo (Universidad de Extremadura, Mérida, Spain), José Manuel Pérez Pintor (Universidad de Extremadura, Cáceres, Spain), and Celia Moreno González (Física Aplicada, Universidad de Extremadura, Cáceres, Spain)

Several studies have demonstrated the relationship between the presence of green spaces and a decreased likelihood of experiencing health issues or exposure to environmental pollutants, including noise. The increase in green areas will be one of the interventions in future cities to reduce pollution and enhance the quality of life for their citizens. However, it is crucial to identify the variables within green spaces that have the most significant impact on well-being and health. Different green areas in the city of Madrid were analyzed in this study, and simultaneously with the recording of environmental variables, their users were surveyed. The study's hypothesis aimed to determine whether the use of green areas has a greater impact on health than their mere presence. The quality of green areas is an influential factor in these relationships; therefore, the quality of the acoustic environment was considered. The results indicate that the acoustic environment is a significant factor in the development of various activities and the overall perception of the park. The visibility of greenery is a variable with psychological benefits, but the use of green spaces with good environmental quality also has a direct relationship with health status.

2:00

2pNSb5. A soundscape perspective for biophilia hypothesis: Theoretical considerations and conceptual approach. Enkela Alimadhi (Interior Architecture and Environ. Design, Bilkent Univ., Faculty of Art, Design and Architecture, Ankara 06800, Turkey, enkelaalimadhi@bilkent.edu.tr) and Semiha Yilmazer (Interior Architecture and Environ. Design, Bilkent Univ., Ankara, Turkey)

This study proposes a conceptual framework to consider the integration of the soundscape perception into the Biophilia Hypothesis paradigm and to further investigate the positive psychological outcomes such as affection or cognition indicators, and physiological outcomes or neurophysiological outcome indicators based on brain signal analysis on building occupants. The biophilia hypothesis claims that a natural content environment has positive psychological and physiological outcomes identified as restorative. The established soundscape framework, which is still under exploration in indoor environments, will help frame the rationale of this study to understand the soundscape perception and consider how to integrate it into the biophilic theoretical claims. In this line of thought, the conceptual framework emphasizes the significance of not just the visual realm, as suggested by the Biophilia hypothesis paradigm through empirical, experimental, and theoretical claims. Additionally, incorporating sound can contribute to a more holistic approach. To do so, based on an literature review and also on our very preliminary study this conceptual framework is an extension of the soundscape theoretical framework. It proposes the integration of complementary theories such as Attention Restoration, Stress-Recovery Theory, Biophilia hypothesis, and the soundscape framework to be further implemented for healthy designed environments.

2:15–2:55
Panel Discussion

Session 2pNSc

Noise and ASA Committee on Standards: Honoring the Life and Legacy of Tony F. W. Embleton

Bennett M. Brooks, Cochair
Brooks Acoustics Corporation, 49 N. Federal Highway, #121, Pompano Beach, FL 33062

Nicholas Sylvestre-Williams, Cochair
Aercoustics Engineering Ltd., 165-50 Ronson Drive, Toronto, M9W 1B3, Canada

Chair's Introduction—3:05

Invited Papers

3:10

2pNSc1. Tony Embleton—Distinguished researcher, successful manager, consummate acoustician. Bennett M. Brooks (Brooks Acoust. Corp., 49 N. Federal Hwy., #121, Pompano Beach, FL 33062, bbrooks@brooksacoustics.com)

Tony F. W. Embleton was a distinguished researcher in acoustics. His early career focused on practical noise control projects and experimental non-linear acoustics. Later, Tony made important contributions to the understanding of sound propagation outdoors. Tony spent much of his career at the National Research Council of Canada (NRCC). His professional homes were the ASA and the CAA. Tony received numerous awards from the ASA for his work, including the R. Bruce Lindsay Award in 1964, the ASA Silver Medal in Noise in 1986, and the ASA Gold Medal in 2002 "For fundamental contributions to understanding outdoor sound propagation and noise control and for leadership in the Society." Tony recognized the value of professional organizations and served the ASA as Vice-President (1977–78) and President (1980–81). He also understood the importance of practical applications in acoustics serving as ASA Standards Director (1993–97), navigating the organization through a tough period, and growing the program toward a solid future. Most importantly, Tony was known for his kind and generous spirit, and his outreach and inspiration to newer colleagues. Many of us are the grateful beneficiaries of his gentle enthusiasm.

3:40

2pNSc2. Tony F. W. Embleton in the Acoustics Section, Physics Division, of the National Research Council of Canada (NRCC). Anthony J. Brammer (Medicine, Univ. of Connecticut Health, Farmington, CT, brammer@uchc.edu), Gilles Daigle (Gatineau, QC, Canada), Michael Stinson (Carlsbad Springs, ON, Canada), and Floyd Toole (Ottawa, ON, Canada)

Tony completed his PhD in three years at Imperial College (London) studying under Dr. R.W.B. Stephens. He then joined George Thiessen and Edgar Shaw at NRCC to form the core of what became arguably the most influential and productive research group in acoustics in Canada from the 1960s to the 1980s. An important activity was service to industry, a role Tony embraced by collaborating with industry associations, participating in committees and directing his research activities. Working with George Thiessen, he succeeded in reducing the noise of couch rolls, a major source of noise in paper making, by randomizing the pattern of holes through which air was sucked to dry the paper. Other successful noise control projects included staggered stator blades for gas turbine engines and mufflers for rock drills. Tony made seminal contributions to outdoor sound propagation and refined condenser microphone calibration. In addition to his research and outreach, he found time for professional service to the ASA and CAA, including serving as founding editor of what is now Canadian Acoustics, and to develop standards and mentor younger scientists, He is remembered for his cheerfulness and willingness to provide advice and engage in conversation on any and all topics.

4:10

2pNSc3. The 1952 Ph.D. dissertation of T.F.W. Embleton on nonlinear acoustic propagation and reflection. Victor W. Sparrow (Grad. Prog. Acoust., Penn State, 201 Appl. Sci. Bldg., University Park, PA 16802, vws1@psu.edu)

In his doctoral research Tony Embleton studied nonlinear acoustics experimentally using techniques available in the Physics Department of Imperial College, University of London. Originally developed by V. Timbrell, the interferometric equipment was substantially improved by Tony Embleton, enabling his Ph.D. findings. Studying propagation in a tube and reflection by both closed and open ends, Tony Embleton was able to substantially confirm the theory of the time, developed by Stokes, Earnshaw, Rankine, Rayleigh, and Taylor, many pieces confirmed for the first time. He also found discrepancies regarding the attenuation coefficients regarding the first and second half cycle of each pulse as well as a shortening of a pulse at reflection from a closed end. These findings are still worthy of further investigation. The results of reflection of pulses from the open end of a tube, forming a pressure release boundary condition, still are very interesting to the present author. A pulse propagating toward the open end steepens, but the reflected wave "unsteepens" which is a unique nonlinear acoustic phenomenon. [Work supported by the Penn State College of Engineering and its United Technologies Corporation Professorship.]

4:40

2pNSc4. Tony Embleton and the ASA Standards Program. Bennett M. Brooks (Brooks Acoust. Corp., 49 N. Federal Hwy., #121, Pompano Beach, FL 33062, bbrooks@brooksacoustics.com) and Stephen J. Lind (LindAcoustics LLC, Onalaska, WI)

Tony F. W. Embleton had a distinguished career as a researcher in acoustics, and as a leader in the ASA and CAA. Yet, he was keenly aware of the importance of standardization to practical applications in acoustics. Tony served as ASA Standards Director from 1993 to 1997, when the organization was going through a tough period, even facing abolishment by the ASA Executive Council. Tony reinvigorated the ASA Standards Program, including increasing its connection to the Technical Committees, as well as its administrative operations [T. F. W. Embleton, "Standards and the Acoustical Society," J. Acoust. Soc. Am. 98, 691–693 (1995); S. Blaeser and C. J. Struck, "A history of ASA standards," J. Acoust. Soc. Am. 145 (1), (2019)]. Tony also was instrumental in founding CAA's Standards program. Tony's legacy of engagement with the ASA technical and administrative leadership is still evident today. An overview of how Tony's ideas remain a guiding force in the current ASA Standards Program is given.

5:10–5:30
Panel Discussion

TUESDAY AFTERNOON, 14 MAY 2024 ROOM 202, 1:00 P.M. TO 5:30 P.M.

Session 2pPA

Physical Acoustics, Structural Acoustics and Vibration and Engineering Acoustics: Resonant Ultrasound Spectroscopy For Characterizing Material Property and Structure

Christopher M. Kube, Cochair
Engineering Science and Mechanics, The Pennsylvania State University, 212 Earth and Engineering Sciences Bldg, University Park, PA 16802

Matthew Cherry, Cochair
Air Force Research Laboratory, 2230 10th St., Fairborn, OH 45433

Rasheed Adebisi, Cochair
UDRI, University of Dayton, 141 Firwood Drive, Shroyer Park Center, Dayton, OH 45419

Jeff Rosin, Cochair

Invited Papers

1:00

2pPA1. Resonant ultrasound spectroscopy for textured materials. Julian D. Maynard (Phys., The Penn State Univ., 104 Davey Lab, Box 231, University Park, PA 16802, maynard@phys.psu.edu)

Resonant ultrasound spectroscopy (RUS) is a method in which the least-squares fitting of a sample's measured resonance frequencies is used to determine the sample material's elastic constants. Difficulties arise when RUS is applied to textured materials, which are composed of crystallites that are neither completely aligned nor randomly oriented; such materials would have effective elastic tensors with 21 independent elastic constants, and this would overburden the conventional RUS analysis process. In any case, the elastic nature of some textured materials may be represented as the weighted average of the elastic tensors of the constituent crystalline material rotated in the directions of the individual crystallites. The collection of weights is the orientation distribution function (ODF). Now the least-squares fitting of a sample's measured resonance frequencies may be used to determine not the sample's elastic constants but instead its ODF. This is not trivial since the weights must be positive and sum to one, but it may be done with a simple modification of the conventional RUS software.

1:20

2pPA2. Characterization of effective elastic constants and anisotropy directions for Wire Arc Additive Manufactured steel samples using RUS. Florian Le Bourdais (DIN, Commissariat à l'Énergie Atomique (CEA), CEA Ctr. de Saclay, Gif sur Yvette 91191, France, florian.lebourdais@cea.fr), Audrey Gardahaut, and Nicolas Leymarie (DIN, Commissariat à l'Énergie Atomique (CEA), Gif sur Yvette, France)

In recent years, Resonant Ultrasound Spectroscopy (RUS) has been extensively applied to objects produced by additive manufacturing to characterize elastic material properties, detect defects or geometrical deviations. In this talk, we analyze samples that were produced using the wire-arc additive manufacturing (WAAM) process using different grades of steel wires. Resonance spectra were obtained and allowed to classify samples as either elastically isotropic or anisotropic. Detailed investigation on anisotropic samples (produced with 316L wire) under an orthotropy hypothesis showed that the samples were markedly softer along the layer deposition direction. Subsequent investigation using EBSD confirmed the results obtained with RUS. They also allowed to quantitatively model the elastic constants using the Voigt-Reuss-Hill averaging theory, which were in good agreement with the ones obtained using the RUS inverse problem.

1:40

2pPA3. Resonant ultrasound spectroscopy experimental design and optimization. Joshua Ward (Univ. of Dayton Res. Inst., 300 College park, Dayton, OH 45469, joshua.ward@udri.udayton.edu), Mark Obstalecki, and Matthew Cherry (Air Force Res. Lab., Fairborn, OH)

Resonant Ultrasound Spectroscopy (RUS) is a nondestructive material characterization technique that utilizes solid body resonances to estimate elastic moduli. This method is advantageous over traditional mechanical testing due to the significantly lower number of required samples to estimate the same number of moduli. The goal of this project is to develop a high-throughput RUS system capable of measuring experimental resonances and mode shapes in a fraction of the time of traditional RUS measurements. A new system was designed which leverages data/image analysis algorithms to aid in sample alignment and data collection. A standardized sample geometry was also analyzed by performing a geometric sensitivity study and optimizing the aspect ratio of lengths of a parallelepiped. The results show that the optimized sample geometry is consistent between materials with varying elastic properties. In order to validate the design of the new RUS system a comparative study to an existing system was conducted. The results show that the newly designed system produces comparable resonant frequency and mode shape measurements.

Contributed Papers

2:00

2pPA4. Modeling resonant ultrasound of synthetic polycrystalline microstructures produced by Dream3D. Christopher M. Kube (Eng. Sci. and Mech., The Penn State Univ., 212 Earth and Eng. Sci. Bldg, University Park, PA 16802, kube@psu.edu)

Resonant ultrasound spectroscopy (RUS) is established for determining elastic constants of homogeneous solids. Often applied to polycrystalline materials like metals, RUS presumes homogeneity based on the microstructure's scale. This presumption is challenged with large grain sizes and higher-order modes, complicating homogeneity testing in real samples due to the need for 3D metallurgical analysis. An alternative is RUS modeling on digitized synthetic microstructures. This presentation investigates the RUS variational model with Rayleigh-Ritz solutions on microstructures created using Dream3D. It starts with a proof for solutions that minimize the Lagrangian in heterogeneous solids. Subsequently, Rayleigh-Ritz volume integrals are numerically evaluated for synthetic polycrystalline microstructures, as spatial heterogeneity precludes analytical solutions. No homogeneity assumption is made. The 3D integrals, requiring advanced GPU processing for a detailed mesh, lead to forward model results. These will be compared against common homogenization schemes, highlighting new insights when using RUS for material characterization of polycrystalline materials.

2:15

2pPA5. Constrained anisotropy using spectral structures of elastic tangent tensors. Jim W. Colovos (Vibrant Corp., 8440 Washington St. NE, STE B, Albuquerque, NM 87113, jcolovos@vibrantndt.com)

Geomaterials and biomaterials that are assumed to have symmetry about a single preferred direction have five independent transversely isotropic elastic constants. Spectral decompositions of the anisotropic elastic tangent stiffness tensor have led to an elegant framework for a variety of symmetry classes known to be of importance for characterizing material property and structure. The fourth-order eigenprojectors can be combined to form the isotropic and anisotropic basis tensors with various spatial symmetries and computationally motivated constraints. We consider the mathematically motivated decoupling of tensorially linear and non-linear functions of a texture tensor. Neglecting the non-linear components, as often done for cracked-rock models, imposes a constraint that the lateral shear modulus depends on the remaining elastic constants. We also consider the mathematically motivated decoupling of purely spherical and purely deviatoric components, often used in the field of biomechanics. When the mixed spherical-deviatoric components are neglected, the axial and lateral Poisson's ratios are constrained to become dependent on the two tensile moduli and a new independent bulk modulus type parameter. The described constraints reduce the number of independent elastic constants from five to four. The potential for using similar techniques to investigate elastic constants is discussed for resonant ultrasound spectroscopy, specifically additively manufactured materials, with the goal of increasing robustness to local minima.

Invited Papers

2:30

2pPA6. Abstract withdrawn.

2:50

2pPA7. Enabling resonant ultrasound spectroscopy in extreme environments. Christopher A. Mizzi (National High Magnetic Field Lab., Los Alamos National Lab., Los Alamos, NM) and Boris Maiorov (National High Magnetic Field Lab., Los Alamos National Lab., Los Alamos Nat. Lab, MS E536, Los Alamos, NM 87545, maiorov@lanl.gov)

Elastic moduli are fundamental thermodynamic susceptibilities that connect to thermodynamics, electronic structure, and mechanic properties. Thus, determining the changes of elastic moduli as a function of time or temperature is a powerful tool to study several thermodynamic and materials phenomena. Resonant Ultrasound Spectroscopy (RUS) determines elastic constants with high accuracy and precision from a single measurement of the mechanical resonances of a sample. Conventionally, the quantitative extraction of elastic moduli with RUS assumes free boundary conditions but lead to unstable positioning of the sample making it incompatible with extreme environments like high magnetic fields. In practice, even holding the sample produces contact forces that deviate from free-boundary conditions shifting resonant frequencies. In this talk, we show that, we can reduce the free-boundary conditions (by gluing/adhering the sample to the transducer), while still being able to obtain the full elastic tensor by a simple model of the sample-transducer interaction. This means elastic constants can be determined to within the uncertainty of conventional RUS, but with significant improvements in sample stability and control of sample orientation. We show the results of studying of several magnetic and structural phase-transitions using this new method.

3:10–3:30 Break

3:30

2pPA8. Phonon dispersion at long and short wavelength as seen by ultrasound and neutron spectroscopy. Raphael Hermann (Mater. Sci. and Technol. Div., Oak Ridge National Lab., 1 Bethel Valley Rd., Oak Ridge, TN 37830-8050, hermannrp@ornl.gov)

Lattice excitations are fundamental energy carriers responsible for thermal transport in every solid. Whereas ultrasound spectroscopy typically probes these excitations in the kHz to MHz "sound" regime corresponding to mm to μm wavelengths, inelastic scattering of neutrons or x-rays probe the THz regime and Å wavelength. We will discuss a few examples of materials with peculiar sound and phonon properties, such as chiral α-$TeO_{;2}$ and the $AgSbTe_{;2}$-PbTe (LAST) solid solution. Analysis of resonant ultrasound data of α-$TeO_{;2}$ requires the use of code that supports chiral space groups, such a RUSCal; intriguingly sound is extremely anisotropic, and in some directions the speed of sound is highly frequency dependent in the 0–1 THz range. LAST exhibits nanoscale inhomogeneities where the speed of sound is different in the matrix and the nanoinclusions. Work supported by the US Department of Energy (DOE), Office of Basic Energy Science (BES), and by Laboratory Directed Research at Oak Ridge National Laboratory; this research used resources at the Spallation Neutron Source a facility supported by DOE, BES, Scientific User Facilities Division. J. Torres, A. Flores-Bettancourt, M. Manley, B. Winn, G. Yumnam, I. Sergeev, P. Bauer-Pereira and A. Jafari are gratefully acknowledged for their collaboration.

Contributed Paper

3:50

2pPA9. Using resonant ultrasound spectroscopy to characterize thermally conditioned high explosive materials. Jordan Lum (Lawrence Livermore National Lab., 7000 East Ave., Livermore, CA 94550, lum21@llnl.gov), David M. Stobbe, Paul Mirkarimi, William Shaw, Henry Reinstein (Lawrence Livermore National Lab., Livermore, CA), Rebecca Lindsey (Chemical Eng., Univ. of Michigan, Ann Arbor, MI), and Richard Gee (Lawrence Livermore National Lab., Livermore, CA)

The ability to nondestructively quantify changes in mechanical properties of granular high explosive materials due to thermal conditioning is of importance for a myriad of civil and defense applications and could lead to better understanding of environmental aging-related effects for explosive material performance and safety. In this study, we report the first demonstration of using resonant ultrasound spectroscopy (RUS) to quantify the bulk elastic properties of granular high explosive materials at different bulk pressing densities and degree of thermal conditioning. Monitoring elastic property changes in granular explosive pressings has not yet been demonstrated using RUS, which is an appealing nondestructive characterization tool since it requires only dry point contact with the explosive material and can be applied to explosives with small form factors. Here, explosive material was studied in the form of binderized plastic bonded explosive pressings and unbinderized neat pressings, both of which are used commercially. Elastic stiffness coefficients calculated from the RUS measurements on the binderized and neat pressings show a significant increase for the post-conditioned samples compared to the pre-conditioned samples. This trend of increasing elastic properties with thermal conditioning was consistent for different density pressings and different thermal exposure conditions. [This work was performed by LLNL under Contract No. DE-AC52-07NA27344 and document release number LLNL-ABS-858868.]

Invited Papers

4:05

2pPA10. Microcavity-enhanced photoacoustic vibrational spectroscopy of single particles. Yun-Feng Xiao (School of Phys., Peking Univ., No. 5, Yiheyuan Rd., Haidian District, Beijing 100871, China, yfxiao@pku.edu.cn)

Confinement and manipulation of photons using microcavities have triggered intense research interest in both fundamental and applied photonics for more than two decades. Prominent examples are ultrahigh-Q whispering gallery microcavities which confine photons using continuous total internal reflection along a curved and smooth surface. The long photon lifetime, strong field confinement, and in-plane emission characteristics make them promising candidates for enhancing light-matter interactions on a chip. In this talk, I will focus on single-particle photoacoustic vibrational spectroscopy using optical microcavities.

4:25

2pPA11. Laser-based resonant ultrasound spectroscopy for monitoring secondary phases in metastable Ti-alloys: From growth kinetics to high-throughput characterization. Hanus Seiner (Inst. of Thermomechanics, Czech Acad. of Sci., Dolejskova 5, Prague 18200, Czechia, hseiner@it.cas.cz), Petr Sedlak, Michaela Janovska, and Jitka Nejezchlebova (Inst. of Thermomechanics, Czech Acad. of Sci., Prague, Czechia)

Metastable beta-Ti alloys undergo complex microstructural changes when annealed for extended times (hours, days, weeks) at temperatures below beta-transus. Contact-less RUS turns out to be a perfect tool for monitoring these processes, as well as for analyzing the properties of the alloys after the heat treatment. In the talk, we will illustrate this ability with three examples of recent advancements in this field. In particular, we will show that RUS is capable of analyzing (i) growth kinetics [1], from fast non-equilibrium phenomena to extremely slow processes; (ii) cubic symmetry conservation in single crystals with particles of various secondary phases [2]; and (iii) phase composition-elasticity relationships in series of oligocrystalline samples [3]. [1] Nejezchlebová *et al*. Elastic constants of β-Ti15Mo (2019) J Alloys Compounds, 792, pp. 960–967. [2] Olejňák *et al*. An ultrasound-based evaluation of cubic symmetry preservation and homogeneity in elastic behavior of $\beta+\omega$ and $\beta+\alpha$ Ti-alloys (2023) Materials and Design, 236, art. no. 112474. [3] Preisler *et al*., High-throughput characterization of elastic moduli of Ti-Nb-Zr-O biomedical alloys fabricated by field-assisted sintering technique (2023) J Alloys Compounds, 932, art. no. 167656.

2p TUE. PM

Contributed Papers

4:45

2pPA12. Convergence properties of eigenfrequencies in RUS. Farhad Farzbod (Mech. Eng., Univ. of MS, 1764 University Circle, Rm. 203, University, MS 38677, farzbod@olemiss.edu) and Casey Holycross (Aerosp. Systems Directorate (AFRL/RQTI), Air Force Res. Lab., Dayton, OH)

An in-depth exploration of the asymptotic behavior of resonant frequencies in Resonant Ultrasound Spectroscopy (RUS), a non-destructive method for material property evaluation, is presented in this work. This work delves into the asymptotic analysis using Legendre polynomials for cuboid samples and examines the impact of increasing elements on eigenfrequencies. In this examination, we noted various elements concerning the computational boundaries and the influence of diagonal components on the asymptotes. The presence of a boundary for the asymptotes indicates limited information beyond a certain point. Additionally, how elastic constants influence these eigenfrequencies is discussed.

5:00

2pPA13. Inverse problems in resonant ultrasound spectroscopy based on spectra perturbation. Petr Sedlak (Dept. of Ultrasonic Methods, Inst. of Thermomechanics, Czech Acad. of Sci., Prague, Czechia, psedlak@it.cas.cz), Hanus Seiner, and Michaela Janovska (Dept. of Ultrasonic Methods, Inst. of Thermomechanics of the CAS, Prague, Czechia)

In RUS, the fit between the experimental resonant spectrum and the result of the inversion procedure is usually worse than the inaccuracy of the experimental measurement of the resonant frequencies. Thus, it can be problematic to extract any valuable information from slight perturbation of experimental spectra, where the experimental frequency shift falls well below the accuracy of this fit. However, the inverse procedure can be reformulated and instead of fitting the resonant frequencies itself, it can be based on the direct fitting of the frequency perturbations. In this contribution, we show two examples of the use of this approach (i) quantitative RUS evaluation of the elasticity of micrometric and submicrometric surface layers from frequency differences measured before and after layer deposition [1] and (ii) *in situ* characterization of the thermal degradation of the bonding layer in a bimetallic system [2]. [1] Thomasová M. *et al*. Young's moduli of sputter-deposited NiTi films determined by resonant ultrasound spectroscopy: Austenite, R-phase, and martensite (2015) Scripta Materialia, 101, pp. 24–27. [2] Janovská M. et al Characterization of bonding quality of a cold-sprayed deposit by laser resonant ultrasound spectroscopy (2020) Ultrasonics, 106, art. no. 106140.

5:15

2pPA14. Measuring the internal damping of thin metal beams in flexure. Micah Shepherd (Brigham Young Univ., N249 ESC, Provo, UT 84602, mrs74@byu.edu), Peter K. Jensen (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), and Joshua T. Mills (Brigham Young Univ., Provo, UT)

Since the internal damping of most metals is a relatively small quantity, precise measurement requires that both acoustic radiation damping and boundary condition damping be either eliminated or highly minimized. In this talk, an experiment setup will be described which can measure the internal damping of metal beams in flexure. To perform a measurement, a beam specimen is placed on wire supports within a vacuum chamber and excited by an autonomous force hammer located within the chamber. The wire supports are located exactly at the nodal lines of the beam mode of interest, so that the frictional losses created by the wire supports are minimized. Once excited, the beam deflection is measured with a single point laser vibrometer. The signal decay measured by the vibrometer is used to estimate damping of the mode of interest using either the Hilbert transform or the half-power method. The wire supports are then moved, and the procedure is repeated for the next mode of interest. Results for several metal specimens will be presented and compared to Zener's thermoelastic model, which describes the losses created by thermal currents for a beam in flexure, to demonstrate the accuracy of the measurement method.

Session 2pPP

Psychological and Physiological Acoustics: The Importance of Binaural Listening: Speech Intelligibility, Localization, and Virtual Environments

Axel Ahrens, Cochair
Technical University of Denmark, Ørsteds Plads 352, Kgs. Lyngby, 2800, Denmark

Mark A. Stellmack, Cochair
Psychology, University of Minnesota, 75 East River Parkway, Minneapolis, MN 55455

Chair's Introduction—1:00

Contributed Papers

1:05

2pPP1. Using vocoded speech to study temporal weighting in spatial release from masking. G. Christopher Stecker (Ctr. for Hearing Res., Boys Town National Res. Hospital, 555 N 30th St., Omaha, NE 68131, cstecker@spatialhearing.org) and Brittany T. Williams (Boys Town National Res. Hospital, Omaha, NE)

Several aspects of binaural and spatial hearing appear to be dominated by moments of rising envelope fluctuations ("onsets"), although those phenomena are limited to events occurring less than a few hundred times per second. Thus, for example, the localization of a brief tone is dominated by the binaural information available in the overall sound onset. Whereas for a modulated sound each envelope period may contribute equally, assuming the modulation rate is not too high. For speech signals, syllabic modulations provide multiple onset-like events which may occur synchronously or asynchronously across frequency bands. Although previous studies have shown that rising envelope fluctuations play an outsized role in localization and lateralization, it is unknown whether other aspects of spatial hearing–such as spatial release from masking (SRM)–are similarly dominated by rising envelopes. Here, we used a Gabor-click-train vocoder to transform speech sounds and manipulate the spatial content of rising- versus falling-envelope segments of the speech in each frequency band. Introducing binaural differences between target and masker speech allows the assessment of localization and/or SRM across conditions in which the cues are limited to rising- or falling-envelope segments, or available throughout the signal. [Work supported by NIH R01DC016643, T32000013.]

1:20

2pPP2. A spatial digit task for assessing binaural function in individuals with hearing loss. Douglas Brungart (Walter Reed National Military Medical Ctr., Bethesda, MD, douglas.s.brungart.civ@health.mil), Alyssa Davidson, Kelli Clark (Walter Reed National Military Medical Ctr., Bethesda, MD), and Trevor T. Perry (National Ctr. for Rehabilitative Auditory Res., Bethesda, MD)

Spatial hearing is crucial in occupational tasks, yet until 2019, US Army fitness-for-duty requirements only considered hearing thresholds in the better ear. The regulation change introduced the Military Operational Hearing Test (MOHT) for those with worse-ear thresholds exceeding 40 dB HL at 0.5 or 1 kHz, or 60 dB HL at 2 kHz. This led to the development of the Spatial Digit Test (SDT) to assess binaural function in individuals with poor hearing in one ear but sufficiently good hearing in the other ear to pass a speech-in-noise test. The SDT involves digit pairs presented with Interaural Time Delays (ITDs) of $+/-800\,\mu$s. Two studies were conducted: a validation study with over 200 individuals, and a verification study with more than 130 undergoing MOHT exams. Results show that individuals with normal auditory thresholds perform very well on the SDT, with about 95% correctly identifying at least 8 digits and 75% correctly identifying all 10. In contrast, only about 50% of individuals with monaural losses who took the SDT as part of the MOHT identified 8 or more digits. Results of the SDT were correlated both with subjective hearing complaints and with the Binaural Masking Level Difference. In conclusion, the SDT proves valuable for identifying individuals struggling using ITD cues to segregate and localize simultaneously presented speech signals.

1:35

2pPP3. The effect of eye-gaze direction on speech intelligibility. Tongthai Taotong, Torsten Dau (Tech. Univ. of Denmark, Kgs. Lyngby, Denmark), and Axel Ahrens (Tech. Univ. of Denmark, Ørsteds Plads 352, Kgs. Lyngby 2800, Denmark, aahr@dtu.dk)

In many daily situations we move our eyes to capture different visual information. Previous studies have shown interactions between the eye-gaze direction and auditory perception. The eye-gaze direction has been shown to modulate activity to acoustic stimuli in the auditory cortex of monkeys. In humans, decreased thresholds of binaural lateralization cues and improved accuracy of understanding series of digits in the presence of other speech has been shown. Here, we investigated the effect of the eye-gaze direction on speech perception of single-digit and 4-digit streams while directing the gaze at different directions containing either speech or non-speech stimuli. The speech interferers were also digits but a different gender than the target speech. The non-speech interferers consisted of speech-modulated speech shaped noise. The results showed improved speech intelligibility when the gaze direction and the auditory target direction were matching, in comparison to conditions where the gaze is pointed at a non-target direction. No differences were found comparing single-digit and 4-digit streams. Presenting speech or non-speech stimuli at the gaze direction did not affect the speech intelligibility results. These findings further clarify the influence of the gaze direction on speech perception.

1:50

2pPP4. Relation between spatial release from informational masking and localization discrimination. Benjamin H. Zobel (Speech, Lang., and Hearing Sci., Univ. of Massachusetts Amherst, 358 N Pleasant St., Rm 201A, Amherst, MA 01002, bzobel@umass.edu), Patrick Zurek (Sensimetrics Corp., Woburn, MA), Emily Buss (Otolaryngology/Head and Neck Surgery, Univ. of North Carolina at Chapel Hill, Chapel Hill, NC), and Richard Freyman (Speech, Lang., and Hearing Sci., Univ. of Massachusetts Amherst, Amherst, MA)

Spatial release from masking (SRM) occurs when a signal of interest (target) and an interfering sound (masker) come from different locations in

space. SRM and sound localization are assumed to share common mechanisms, but it is still unclear whether SRM actually *depends* on hearing the target and masker in different places, even in situations dominated by informational masking. The current study used a spatial rhythmic release from masking (RMR) task [J. Middlebrooks and Z. Onsan, *JASA*, 132, 3896–3911, 2012]. The target and masker were trains of non-simultaneous noise bursts; hearing the target temporal pattern relies on segregation from the masker, which is supported by spatial separation. Using a headphone simulation of space (HRTFs), the current study replicated key aspects of Middlebrooks and Onsan's (2012) finding that RMR spatial thresholds were poorer for high-frequency than low-frequency bursts. We investigated whether this low/high frequency difference, and others created with degraded localization cues, were associated with variations in basic localization discrimination tasks using the same RMR stimuli. Data collection is ongoing, but preliminary results suggest a covariation of spatial release in the RMR task and sensitivity to change in spatial angle. [Work supported by NIDCD R01-01625.]

2:05

2pPP5. Front-back bias and the 1000-Hz interaural intensity anomaly. William Hartmann (Michigan State Univ., 749 Beech St., East Lansing, MI 48823, wmh@msu.edu)

Free field measurements were made of the level differences, as measured in an ear canal, for a noise source in front of the head compared to a source in back. The measurements were made on a Kemar manikin for 19 lateral angles over the complete 180-degree range for front-back comparisons. There were 7 octave bands, 125–8000 Hz, and two horizontal planes. Front-back level differences were generally positive except for the dramatic case of 1000 Hz, where the differences were negative over an azimuth range of 120 degrees. When the head-related elevation of the sources was increased from zero to 10 degrees, the range of the negative front-back differences grew to 140 degrees. These physical results are consistent with the strong tendency for listeners to identify a one-third octave noise band in a narrow range near 1000 Hz as a source in back. This tendency may be the origin of the often-observed Grantham effect, wherein the difference limen for the interaural level difference is larger near 1000 Hz than at higher or lower frequencies. The connection is that sources in back of the listener, whatever their frequencies, are not localized nearly as well as sources in front.

2:20

2pPP6. Localization of real and tangent-law panned phantom sound sources in the frontal horizontal plane. Mark A. Stellmack (Psych., Univ. of Minnesota, 75 East River Parkway, Minneapolis, MN 55455, stell006@umn.edu), Stanley Sheft (Dept. of Commun. Disord. and Sci., Rush Univ. Medical Ctr., Chicago, IL), and Ewan A. Macpherson (National Ctr. for Audiol., Western Univ., London, ON, Canada)

Auditory source separations of as little as 1 degree are detectable. However, presenting auditory stimuli at small separations presents technical challenges, with loudspeaker separation limited by transducer diameter. An alternative procedure is to utilize phantom sources, with the perceived position of a single source determined by the relative output levels of two spatially separated loudspeakers. Therefore, it is important to determine whether real and phantom sources can be localized with the same precision. In the present experiment, listeners localized real (individual) sources and phantom sources computed using a tangent-law model giving the same nominal azimuthal angles as the real sources. Listeners used a laser pointer to indicate perceived source location. Infrared cameras detected pointer position with responses stored in terms of azimuth. Signals were broadband or narrowband (300-700 Hz and 3800–4200 Hz) noise, 100 or 500 ms in duration. Generally, phantom sources were localized with less precision than real sources, and high-frequency signals were localized with less precision than broadband or low-frequency signals, with no effect of duration. Results show that phantom sources are localized with sufficient accuracy and precision to be useful in assessing auditory spatial acuity, but they are not localized with the same precision as real sources.

2:35–2:50 Break

2:50

2pPP7. Conversation behavior assessment to estimate communication difficulty. Stefan Klockgether (R&D, Sonova AG, Laubisrütistrasse 28, Stäfa, Zürich 8712, Switzerland, stefan.klockgether@sonova.com), Laurent S. Simon (R&D, Sonova AG, Staefa, ZH, Switzerland), and R. Peter Derleth (R&D, Sonova AG, Stäfa, Zürich, Switzerland)

The communication behavior of humans adapts to the needs of different communication situations. Humans have developed several strategies to successfully communicate in challenging acoustic environments. Some of these strategies can be consciously controlled while in a conversation, but others happen sub-consciously, or as a mixture of both. The applied strategies come with an increased effort, a deviation from common behavior, or the overcoming of personal comfort zones and are usually ceased as soon as the situation allows. This study's aim is to monitor participant's communication behavior in easy and challenging acoustic situations and use this data as a measure for the experienced communication difficulty. The Sonova Real Life Lab allows to investigate natural conversation situations in a controlled acoustic environment. Participants can move around and interact with each other across a 25m^2 stage, while their head positions and orientations are tracked with a motion capturing system. The voices of the participants are recorded using wireless headset microphones. This allows to estimate individual vocal effort and analyze backchanneling and turn taking behavior. This contribution will present and discuss data collected with participants engaged in real one-to-one conversations, while the acoustic background was systematically altered.

3:05

2pPP8. Exploring attentive listening in noise through the just-follow conversation task. William M. Whitmer (Hearing Sci. - Scottish Section, Univ. of Nottingham, Level 3, New Lister Bldg., Glasgow Royal Infirmary, Glasgow G31 2ER, United Kingdom, bill.whitmer@nottingham.ac.uk), David McShefferty (Hearing Sci. - Scottish Section, Univ. of Nottingham, Glasgow, United Kingdom), and Karolina Smeds (ORCA Europe, WS Audiol., Stockholm, Sweden)

Listening to a conversation demands comprehension, attention and prediction. To better understand the impact of these demands, as well as how hearing aids might alleviate them, a conversational listening test should be realistic, repeatable and relatable (i.e., have interpretable units). A test that potentially satisfies these needs is the just-follow conversation (JFC) task, where the listener adjusts the signal level to where they can understand, with effort, the gist of what is being said in a background of noise. Fifty-four participants sat in the centre of a circular loudspeaker array and adjusted the overall level of one monologue, one dialogue, two monologues or two dialogues presented in the front hemifield to where they could just follow the speech four times per trial. Signals were presented in surrounding fixed-level café and same-spectrum noise backgrounds. Bilateral hearing-aid users adjusted aided and unaided; non-users repeated each condition to evaluate reliability. Results showed an increase in JFC for dialogues re monologues. Individual JFC SNRs correlated with SSQ12 speech subscale scores as well as pure-tone threshold averages. JFC reliability was comparable to more objective speech understanding measures, but it may not be suitable to capture perceived conversational benefits for more subtle changes in hearing-aid processing. [Work supported by the UK Medical Research Council Grant No. MR/X003620/1 & WS Audiology.]

3:20

2pPP9. Relationship between objective measures and listener intelligibility of speech processed by source-separation algorithms. Karen L. Payton (Speech Technol. & Appl. Res., 4 Militia Dr., Lexington, MA 02421, kpayton@umassd.edu), Richard Goldhor (Speech Technol. & Appl. Res., Lexington, MA), Behdad Dousti (Commun. Sci. and Disord., Univ. of Cincinnati, Cincinnati, OH), Sarah Dugan (Rehabilitation, Exercise, and Nutrition Sci., Univ. of Cincinnati, Dayton, OH), Suzanne Boyce (Commun. Sci. and Disord., Univ. of Cincinnati, Cincinnati, OH), and Joel MacAuslan (Speech Technol. & Appl. Res., Lexington, MA)

A major determinant of success for separating a speech signal from a noisy environment is the intelligibility of the extracted speech signal. Intelligibility is best measured as the fraction of words correctly recognized by listeners, but computational measures are often preferred because they are less labor-intensive than collecting listener judgements. We compare listener intelligibility data to three acoustically derived measures: (a) SNRs estimated from the processed mixtures, (b) coherence and (c) speech-based Speech Transmission Index (sSTI). Sentences were recorded against restaurant babble, white Gaussian noise, and nonstationary noise by four microphones at different SNRs ranging from $+4$ dB to -8 dB. Processing conditions included (1) the original mixture; (2) the mixture processed by a critically determined 4-channel blind source separation (BSS) algorithm; (3) the mixture processed by a 2-channel, underdetermined, BSS algorithm; (4) the residual after subtracting noise estimates determined using a least-mean squared (LMS) algorithm; (5) an estimate of speech extracted using an LMS algorithm to remove the two noises and then 2-channel BSS to separate the sentences from the babble; and (6) pristine speech recorded with no noise. The computational measures are compared to gold-standard intelligibility results from listening tests.

TUESDAY AFTERNOON, 14 MAY 2024 ROOM 206, 1:00 P.M. TO 4:10 P.M.

Session 2pSA

Structural Acoustics and Vibration, Engineering Acoustics and Physical Acoustics: Acoustic Metamaterials II

Christina Naify, Cochair
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Abigail D. Willson, Cochair
Acoustics, Penn State University, PO Box 30, Mail Stop 3220B, State College, PA 16804

Chair's Introduction—1:00

Invited Paper

1:05

2pSA1. Higher-order Non-Hermitian skin effect in an acoustic lattice. Yun Jing (Graduate Program in Acoust., Pennsylvania, 201 Appl. Sci. Bldg., State College, PA 16802, jing.yun@psu.edu), Jiaxin Zhong (Graduate Program in Acoust., Pennsylvania, State College, PA), Wladimir Benalcazar (Phys., Emory Univ., Atlanta, GA), Kevin Kim (Graduate Program in Acoust., Pennsylvania, University Park, PA), and Mourad Oudich (Graduate Program in Acoust., Pennsylvania, State College, PA)

Non-Hermitian physical systems offer a distinct band topology that gives rise to the interesting phenomenon known as the non-Hermitian skin effect (NHSE). However, the exploration of higher-order NHSE in classical wave systems by leveraging non-reciprocity has been largely unexplored. In this study, we introduce a novel approach to experimentally realize the higher-order NHSE in a two-dimensional non-reciprocal breathing acoustic Kagome lattice. This lattice is composed of acoustic cavities with non-reciprocal coupling achieved through electrically controlled active acoustic elements. We successfully observed second-order corner modes using a 3×3 lattice, which manifest as localized pressure distributions at specific frequencies in both parallelogram and triangular topological acoustic lattices. We delve into the underlying mechanisms and their implications, thereby paving the way for a deeper understanding of higher-order NHSE and potentially innovative applications rooted in the acoustic non-reciprocity.

1:25

2pSA2. Breathers for nonlinear energy management in acoustic waveguides. Mohammad A. Bukhari (Mech. Eng., Wayne State Univ., 5050 Anthony Wayne Dr., 2140, Detroit, MI 48202, bukhari@wayne.edu), Oumar Barry (Mech. Eng., Virginia Tech, Blacksburg, VA), and Alexander F. Vakakis (Mech. Sci. and Eng., Univ. of Illinois Urbana-Champaign, Urbana, IL)

Recent investigations of nonlinear metamaterials have revealed interesting wave propagation phenomena with no counterpart in linear systems. One of the appealing phenomena is traveling discrete breathers. Breathers are traveling oscillatory wavepackets with spatially localized envelopes and energy-dependent speeds. These nonlinear waves were reported in lattices with purely (or nearly purely) nonlinear neighbor interactions (possessing zero speed of sound as defined in the classical acoustics sense, and, hence, designated as sonic vacua). However, the effect of local resonators on breather propagation is elusive to date. In this work, we aim to fill this gap by studying 1D grounded strongly nonlinear lattices with linear local resonators (SNLRMs) under impulsive force excitation. Numerical simulations have demonstrated that SNLRMs support the generation of new families of breathers. These families resulted from the opening of a new optical propagation zone (PZ) in the spectrum, which is associated with the linear dynamics of the local resonators. These families are characterized by the local resonator parameters and can support multiple fast frequencies or a single frequency that belongs to either the optical or acoustical PZs. Further aspects of the nonlinear acoustics are also demonstrated using the complexification averaging method. The newly reported breathers open the venue to employ SNLRMs in passive nonlinear energy management applications. Examples are presented through acoustics waveguides and frequency filtering.

1:40

2pSA3. Using a leaky-wave antenna as an acoustic prism for speech-separation. Abigail D. Willson (Appl. Res. Lab., Penn State Univ., PO Box 30, M.S. 3220B, State College, PA 16804, adw5@psu.edu), Andrew S. Wixom, and Amanda Hanford (Appl. Res. Lab., Penn State Univ., State College, PA)

Leaky-wave antennas (LWA) are an acoustic metamaterial comprised of an array of unit cells. For a 1D line array, the output ranges from 0° to 180° relative to the direction of the axis of the array. The direction of the output is dependent on the frequency of the input; a single frequency input will produce a highly directive output beam at a specific angle. Much like an optic prism, which can be used to split a complex optic wave into its components, a LWA can be used to split a complex input audio wave into its components. This work explores the capabilities of LWA to separate transient signals like human speech into different output directions. This is accomplished by way of an equivalent circuit model of the LWA that enables simulations in both the time and frequency domains. The frequency domain performance of the model is validated against previously published results, and then the transient response is shown to recover these as well once steady-state is reached

1:55

2pSA4. Vibroacoustic metamaterial for enhanced Sound Transmission Loss with variable frequency range designed for serial production. Klara Chojnacka (AGH Univ. of Sci. and Technol., Mickiewicza 30, Cracow 30-059, Poland, klara.chojnacka@agh.edu.pl), Aleksander Kras (Silencions Sp. z o.o., Wroclaw, Poland), and Tadeusz Kamisinski (AGH Univ. of Sci. and Technol., Cracow, Poland)

Recently, vibroacoustic metamaterials have been broadly investigated, especially for noise and vibration mitigation. By creating a band gap effect in flexural wave propagation in a base element, metamaterials improve its vibroacoustic parameters in low and mid frequency range while preserving low mass and structure dimensions. While the foundational assumptions and lab validations highlight the advantages of these metamaterials, the practical application of these structures in real-world scenarios remains a challenge. The main aspect that is widely investigated is achieving maximum effectiveness while maintaining the simplicity of geometry to optimize mass production costs. This work presents a vibroacoustic metamaterial design with geometry adapted to serial production using injection molding. The proposed geometry allows for adjusting the effective frequency range of the structure after the production process. Due to the distinctive configuration of the elements and the grouping of unit cells, the design facilitates the generation of broadband and multi-band structures as well. Numerical simulations were employed to assess the impact of the proposed metamaterial on Sound Transmission Loss and other vibroacoustic parameters. Experimental validation of prototype effectiveness was conducted through Sound Reduction Index measurements in a diffuse field.

2:10

2pSA5. General acoustic Willis media modeled as conventional materials with embedded sources. Mehmet U. Demir (Mech. Eng., Univ. of Michigan, G.G. Brown Lab., 2350 Hayward, Ann Arbor, MI 48109, udemir@umich.edu) and Bogdan-Ioan Popa (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

Willis coupling vectors provide additional degrees of freedom to manipulate acoustic waves unavailable in conventional materials, but the additional material properties make it difficult to simulate the interaction between sound and Willis media especially when inhomogeneous media with arbitrary geometries are involved. To address this challenge, we establish the equivalence between the wave equation inside a three-dimensional, inhomogeneous, and anisotropic Willis medium to the well-known acoustic wave equation in conventional materials with embedded continuous distributions of monopole and dipole sources. We validate the equivalence in numerical simulations showing that conventional materials with embedded sources replace physical Willis metamaterials anisotropic in the Willis and mass density tensors without modifying the scattered sound distribution. Moreover, the equivalence offers insights into the physics of Willis materials. For example, it shows that various combinations of Willis coupling vectors can generate identical sound scattering for any excitation. It also shows whether effective material parameters extracted from single Willis cell simulations remain valid in bulk metamaterials obtained by replicating periodically that cell. These results suggest that the equivalence model can advance the design of Willis metamaterials and provide a tool to better understand the physics of Willis media.

2:25–2:40 Break

2:40

2pSA6. Study on vibration and wave propagation characteristics of thin-walled metamaterial structure with embedded hexagonal Acoustic black hole lattice. Chinna B. Singepogu (Aerosp. Eng., Indian Inst. of Sci. Bangalore, Indian Inst. of Sci. Bangalore, Bangalore 560012, India, singepogub@iisc.ac.in), Dinesh K. Harursampath (Aerosp. Eng., Indian Inst. of Sci. Bangalore, Bangalore, India), and Ramesh G. Burela (Shiv Nadar Univ., Noida, India)

An acoustic black hole is an inhomogeneity of structure that can absorb the total incident wave energy by trapping the wave and reducing the group velocity of the propagating wave. ABH can be achieved by varying the thickness to zero, which is impossible, so the power law profile is the ideal thickness variation. The dispersion characteristics of hexagonal acoustic black hole lattices were studied using band gap analysis for the irreducible Brillouin zone. The study also used the geometric nonlinearity for ABH. Later, modal analysis of the plate with hexagonal black hole arrangement is done for harmonic excitation cases for various boundary conditions. All the numerical finite element simulations for the above analysis are done using commercially available software, COMSOL Multiphysics. Keywords: Acoustic black hole, Irreducible Brillouin zone, Modal analysis, COMSOL Multiphysics.

2:55

2pSA7. Numerical extraction of three-dimensional acoustic polarizabilities for acoustically small scatterers. A. J. Lawrence (Walker Dept. of Mech. Eng. and Appl. Res. Labs., Univ. of Texas at Austin, 204 E. Dean Keeton St., Stop C2200, Austin, TX 78712-1591, ajlawrence@utexas.edu), Samuel P. Wallen, and Michael R. Haberman (Walker Dept. of Mech. Eng. and Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX)

Polarizability is a convenient descriptor for scattering from low-*ka* inhomogeneities in both electromagnetism, where it originated, and in acoustics, where it is increasingly used in the design of metamaterial elements. In two or three dimensions, the polarizability is represented by a block matrix that couples the local pressure and particle velocity fields to the lowest-order components of the multipole expansion, usually truncated to dipole order. A fully dense acoustic polarizability matrix has components that couple spatially uniform pressure and velocity oscillations to dipole and monopole scattering, respectively. This unique coupling leads to acoustic bianisotropy, also known as Willis coupling, for a collection of scatterers. The design of Willis materials necessitates computationally efficient methods to extract all components of the polarizability matrix and, at present, only two-dimensional analytical extraction methods are found in the literature. In this work, we present an algorithm to extract all components of the three-dimensional polarizability matrix from a scatterer with arbitrary geometry and composition. The method presented here is numerically efficient and yields results that are in agreement with an analytically obtained benchmark polarizability, providing a useful tool for acoustic metamaterial design.

3:10

2pSA8. Abstract withdrawn.

3:25

2pSA9. Design and analysis of a compact sound absorber made of multiple parallel Helmholtz resonators. Zacharie Laly (Dept. of Mech. Eng., Sherbrooke Univ., 2500, boul. de l'Université, Sherbrooke, QC J1K 2R1, Canada, zacharie.laly@usherbrooke.ca), Chris Mechefske (Dept. of Mech. and Mater. Eng., Queen's Univ., Kingston, ON, Canada), Sebastian Ghinet, and Tenon Charly Kone (Aerosp., National Res. Council Canada, Ottawa, ON, Canada)

In this paper, a compact sound absorber material is proposed and studied for noise attenuation at multiple frequencies using finite element method. The global cylindrical cavity of the material is partitioned into twelve sub-cavities where four are located in the center and one extended neck is connected to each sub-cavity. Each sub-cavity with the associated neck represents a Helmholtz resonator and thus the proposed material design is made of twelve parallel Helmholtz resonators. The sound absorption coefficient and the transmission loss present twelve resonant peaks at different frequencies where the surface impedance is close to the air impedance. The resonant frequencies can be adjusted by the geometrical parameters of the necks and the sub-cavities volumes. The proposed sound absorber can be used in multiple engineering applications to attenuate noise simultaneously at twelve different frequencies.

3:40

2pSA10. Sound absorption analysis of a metamaterial based on parallel dual Helmholtz resonators. Zacharie Laly (Dept. of Mech. Eng., Sherbrooke Univ., 2500, boul. de l'Université, Sherbrooke, QC J1K 2R1, Canada, zacharie.laly@usherbrooke.ca), Chris Mechefske (Dept. of Mech. and Mater. Eng., Queen's Univ., Kingston, ON, Canada), Sebastian Ghinet, and Tenon Charly Kone (Aerosp., National Res. Council Canada, Ottawa, ON, Canada)

In this paper, a sound absorbing material consisting of four parallel dual Helmholtz resonators is proposed and its sound absorption coefficient is studied using finite element method. The dual Helmholtz resonator is made of neck-cavity-neck-cavity where each neck extends into each cavity. The sound absorption coefficient of the dual resonator presents two resonant peaks. It is demonstrated that when the radius of the first or the second neck increases, the two resonant frequencies of the sound absorption increase while they decrease when the length of the first or the second neck increases. The proposed material design, which combines four parallel dual Helmholtz resonators, presents eight sound absorption peaks and these eight resonant frequencies can be tuned to specific frequencies by adjusting the parameters of the necks. It is a compact sound absorber, which can help to attenuate the noise simultaneously at eight different frequencies in several engineering applications.

3:55

2pSA11. Acoustic properties of natural fiber reinforced composite micro-perforated panel designed using 3D printing. Umberto Berardi (Toronto Metropolitan Univ., 350 Victoria St., Toronto, ON M5B 2K3, Canada, uberardi@ryerson.ca)

The present study investigated the acoustic performance of biodegradable MPP absorbers made of natural fiber-reinforced composites (NFRC) using 3D printing. The novelty of this current research lies in the recent development of a methodology that aids industry professionals in optimizing the production of MPP (Micro Perforated Panel) at a competitive cost. This is achieved by addressing and eliminating issues commonly faced in traditional manufacturing processes, such as manual preparation and pressing. The FDM technique was used to fabricate test samples utilizing the PLA/corkwood composite. Using an impedance tube device with two microphones, the acoustic absorption coefficients of MPPs with different perforation diameters, thicknesses, and perforation rates were measured. Maa's analytical model was used to predict the acoustic absorption performance. Moreover, considering the average sound absorption and total cost of fabricating the samples, RSM-CCD was employed to optimize these samples. In the end, the parallel arrangement of MPP double layer and the combination of MPP with kenaf porous material were tested to improve the sound absorption performance. The results showed that the average sound absorption coefficient of the NFRC-MPP sound absorber is 25% more than that of conventional MPP sound absorbers.

Session 2pSCa

Speech Communication: VowelFest: Honoring the Past and Celebrating the Present II

Ewa Jacewicz, Chair
Speech and Hearing Science, The Ohio State University, 1070 Carmack Road, 110 Pressey Hall, Columbus, OH 43210

Invited Papers

1:00

2pSCa1. Acoustic vowel distances within and across dialects. Cynthia G. Clopper (Ohio State Univ., 1712 Neil Ave., Columbus, OH 43210, clopper.1@osu.edu)

Regional dialects of American English exhibit variation in the phonetic realization of vowel categories. This variation leads to different degrees of phonetic vowel similarity within and across dialects. This vowel similarity has been captured by measures of formant distance and overlap in the acoustic space, typically with static formant estimates from a single timepoint in the vowel. In contrast, dialect similarity has been assessed using measures of acoustic distance over time. In the current study, dynamic measures of acoustic distance, including dynamic time warping, root mean-square distance, and generalized additive mixed effect models, were used to assess within-dialect vowel category distances and to compare these distances across dialects. The results were generally consistent across the different distance metrics, confirming their respective ability to capture acoustic vowel distance. However, the magnitude of the distances varied across dialects and vowel variables. Whereas some dialect-specific variables showed the expected patterns of within- and across-dialect similarity based on previous descriptive work, other variables did not. These mixed findings have implications for our understanding of regional dialect variation in American English, its effect on acoustic vowel similarity, and the relationship between acoustic vowel similarity and perceptual ambiguity of vowel contrasts within and across dialects.

1:20

2pSCa2. Regional vowel patterns as shown by discrete cosine transforms. Erik R. Thomas (English, North Carolina State Univ., Dept. of English, Box 8105, Raleigh, NC 27696-8105, erthomas@ncsu.edu), Jeff Mielke (English, North Carolina State Univ., Raleigh, NC), and Kirk A. Hazen (CVS Health, Warrenton, NC)

This project presents a new approach to analyzing geographical patterning in vowel variation that combines discrete cosine transforms (DCTs) with cluster analysis. It offers a means of reducing bias from analyses of geographical patterning in vowel variation. DCT0 and DCT1 capture the overall position of vowels in the vowel envelope and DCT2 adds information about curvature. To mitigate anomalies, these metrics are based on numerous tokens and measurement points. Cluster analysis can then be applied to the DCT data to indicate which speakers are most similar without predetermined groupings. The procedure suggests how vowel realizations are correlated with geographical divisions, if at all, within the area covered by a dialect survey. Here, we apply DCTs and hierarchical clustering to a corpus of speakers born 1970 or later and covering eastern Ohio, West Virginia, and western North Carolina. The results align only partially with isophones from earlier dialect surveys. Analyses of individual vowel phonemes typically exhibit considerable intermixture of forms; clearer geographic patterns emerge primarily when multiple vowels in named chain shifts are considered together. Recent dialect leveling appears to play a role in the paucity of distinguishable regional patterns.

Contributed Papers

1:40

2pSCa3. Neurophysiological correlates of the perceptual magnet effect in speech perception. Yang Zhang (Speech-Language-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr SE, Minneapolis, MN 55455, zhang470@umn.edu)

This study investigated the Perceptual Magnet Effect (PME) in speech perception using behavioral and Event-Related Potentials (ERP) measures. There were two primary research questions: (1) whether category goodness rating influences speech discrimination with reduced sensitivity near the prototype and increased sensitivity in the vicinity of a poor exemplar, (2) whether the PME is domain-specific to speech sounds. Twenty adult native English speakers participated in identification, goodness rating, and discrimination tasks. The stimuli were synthesized vowels for /a/ and their nonspeech analogs by systematically varying the first two formants. ERP oddball conditions included four conditions presented in a counter-balanced order with the prototypical /a/ as the standard stimuli and its variants as deviants, and a non-prototypical /a/ as the standard and its variants as deviants, as well as the nonspeech matches. Results indicated that participants rated vowel category goodness based on the F2/F1 ratio for the /a/ sounds. Mismatch negativity amplitudes in the two speech conditions aligned with PME predictions, but similar patterns were also observed in the nonspeech conditions. The findings collectively demonstrated neurophysiological evidence for the perceptual organization of within-category variations in line with PME, which may transfer to auditory processing of similar acoustic patterns in nonspeech.

1:55

2pSCa4. Implicit and explicit responses to infant sounds: A cross-sectional study among parents and non-parents. M. Fernanda Alonso Arteche (School of Commun. Sci. & Disord., McGill Univ., 1295 Rue des Carrieres, apt 402, Montreal, QC H2S 0E1, Canada, maria.alonsoarteche@mail.mcgill.ca), Leatisha Ramloll (School of Commun. Sci. & Disord., McGill Univ., Montreal, QC, Canada), Lucie MENARD (Linguist, UQAM, Montréal, QC, Canada), and Linda Polka (School of Commun. Sci. & Disord., McGill Univ., Montreal, QC, Canada)

This research investigates the infant schema in auditory perception by examining how different demographics, including males, females, parents, and non-parents, respond implicitly and explicitly to baby vocalizations in comparison to adult, cat, and kitten sounds. Utilizing a single category implicit association task (SC-IAT) and a detailed questionnaire, we analyzed participants' responses to synthesized vowel sounds from infants, adults, cats, and kittens. The questionnaire focused on participants' liking and perception of cuteness for these sounds. Findings reveal a universal positive implicit preference for baby vocalizations across all groups ($p = 0.01$), without a similar effect for other sound sources. In contrast, explicit responses varied significantly. While all groups showed a preference for the sounds of cats, babies, and kittens over adults, only mothers demonstrated a statistically significant explicit preference for infant sounds over those of cats and kittens. This study highlights the discrepancy between unconscious and conscious attitudes towards infant sounds. It underscores that while an implicit affinity for baby vocalizations is widespread, explicit preferences, particularly in terms of cuteness and likeability, are markedly stronger in mothers. These insights contribute to our understanding of the auditory dimension of the infant schema, emphasizing the role of gender and parental status in shaping responses.

TUESDAY AFTERNOON, 14 MAY 2024 — ROOM 214, 2:30 P.M. TO 5:30 P.M.

Session 2pSCb

Speech Communication: VowelFest III (Poster Session)

Ewa Jacewicz, Chair
Speech and Hearing Science, The Ohio State University, 1070 Carmack Road, 110 Pressey Hall, Columbus, OH 43210

All posters will be on display from 2:30 p.m. to 5:30 p.m. Authors of odd-numbered papers will be at their posters from 2:30 p.m. to 4:00 p.m. and authors of even-numbered papers will be at their posters from 4:00 p.m. to 5:30 p.m.

Contributed Papers

2pSCb1. Losing ground?: Towards a description of palatal lateral variation in Breton. Kaitlyn Owens (Indiana Univ. Bloomington, 355 North Eagleson Ave., GA 3151, Bloomington, IN 47405, kaitowen@iu.edu)

Traditional Breton descriptions and grammars attest that [j] is an often-pronounced variant of /ʎ/ (Hemon, 1995), however, little is known about what factors influence this phoneme's realization. This study aims to elucidate what linguistic and social factors affect Breton /ʎ/ pronunciations. Given that Breton new speakers are often militant in promoting Breton revitalization (Jones, 1995) and their pronunciations, unlike those of traditional speakers, are strongly influenced by French (Hornsby, 2015), we predicted new speakers will produce [j] more than traditional speakers. We elicited 125 tokens that contain /ʎ/ from eight Breton-speaking participants in a wordlist task—the only categorical [j]-producer was the sole traditional Breton speaker to participate. Focusing on new speakers, we use mixed-effects regression and find older speakers produce [ʎ] more often than younger speakers ($p < 0.0001$), and gender is not a significant predictor. For linguistic factors, word position does not play a significant role, but [j] is more frequent in intervocalic tokens when the following vowel is palatal ($p = 0.0249$). Although /ʎ/ variation does not exhibit effects of prestige, our results suggest a potential change in progress whereby [ʎ] is being lost in favor of [j] and is particularly gaining momentum intervocalically when the following vowel is palatal.

2pSCb2. Acoustic variability underlying pathological voice quality. Jody Kreiman (Head and Neck Surgery, UCLA, 1000 Veteran Ave., 31-19 Rehab Ctr., Los Angeles, CA 90095-1794, jkreiman@ucla.edu)

Studies have shown that a few acoustic parameters (variability in spectral energy in the higher frequencies, plus formant dispersion) account for the most variability in normal human voices, regardless of sex, age, or language spoken. These parameters also characterize vocal variation in other animals and carry biologically important information about the speaker. They further allow us to conceive of voices as existing in a simple, low-dimensional "quality space" that could facilitate quick judgments of similarity and difference. How well does this model fit pathological voices, given that one hallmark of voice disorders is increased variability? In a preliminary study of sustained vowel phonation in pathologic voices [*J. Acoust. Soc. Am.* 150, A191] we found that the variability introduced by pathology did not replace or obscure universal factors. Instead, measures of variability and instability carried greater significance for the quality of pathologic voices compared to normal voices. Expanding on this finding, this paper employs principal component analysis to derive the factors characterizing vocal variability from acoustic profiles of read speech for pathological voices. Results may provide insights into treatment strategies aimed at reducing the perceived pathology in disordered voices.

2pSCb3. Revisiting the sociophonetics of sexuality in spontaneous speech. Amber Galvano (Linguist, UC Berkeley, Dwinelle Hall #2650, Berkeley, CA 94704, amber_galvano@berkeley.edu)

This study surveys sociophonetic variation in sibilant and vowel production for 44 Bay Area English speakers. I focus on (i) whether sexual orientation is an independent predictor, or interacts with gender, and (ii) what the patterning of bi + (bisexual, pansexual, queer, etc.) and non-binary speakers adds to previous claims (e.g., Gaudio 1994, Pierrehumbert 2004, Willis 2023). All participants completed a Map Task in pairs; 37 participated in a sociolinguistic interview. Recordings were auto-transcribed, forced-aligned, and hand-checked. For /s ʃ z ʒ/, tokens under 0.05 sec, in /str/ clusters, and/or adjacent to another sibilant were excluded. Before getting COG and skew, 0.02 sec were subtracted from token edges and voicing was Hann band-filtered. For all vowels and sibilants, mean and midpoint f0 were taken, and F1-F4 for vowels (using the median of values at midpoint plus six surrounding timepoints). Results show sexual orientation alone is not a robust predictor of /s/ frontedness. Both gender and sexuality interacted with duration for /s/ frontedness and vowel f0; gender and sexuality also interacted to influence /ʃ/ frontedness and /z/ devoicing, as well as vowel space shape. These results imply that, when bi + and non-binary identities are incorporated, the phonetic indexation of sexual identity is not easily generalizable within orientation labels, at least for informal speech.

2pSCb4. Speech adaptation in conversation: Effects of segment cue-weighting strategy for non-native speakers. Han Zhang (Dept. of Linguist, Simon Fraser Univ., 8888 University Dr., Burnaby, BC V5A 1S6, Canada, han_zhang7@sfu.ca), Morgan Glover, Fenqi Wang, Xizi Deng (Dept. of Linguist, Simon Fraser Univ., Burnaby, BC, Canada), Dawn Behne (Dept. of Psych., Norwegian Univ. of Sci. and Technol., Trondheim, Norway), Allard Jongman (Dept. of Linguist, Univ. of Kansas, Lawrence, KS), Joan Sereno (Dept. of Linguist, Univ. of Kansas, Kansas City, KS), and Yue Wang (Dept. of Linguist, Simon Fraser Univ., Burnaby, BC, Canada)

Research has investigated how speakers make phonetic adaptations by adjusting their speech sounds to either converge to or diverge from their interlocutor's. Less is known about the dynamic adaptive strategies non-native speakers use to improve intelligibility when interacting with native speakers, particularly how their strategies change over the course of spontaneous conversations. The current study examines phonetic adaptations in English words contrasting tense and lax vowels (e.g., sheep-ship) in unscripted conversations between non-native Japanese English and native English speakers during an engaging computer game task. Japanese speakers have previously been found to rely on temporal cues (vowel length) for tensity distinctions due to their L1 cue-weighting strategy rather than spectral cues (vowel quality) that are predominantly used in English. Acoustic analyses are conducted to examine changes in non-native vowel productions before, during and after the conversation task. We predict that, in an attempt to overcome miscommunication, non-native speakers may initially lengthen the tense vowels to distinguish them from their lax counterparts. As the conversation progresses, non-native speakers may adapt to a more native-like cue-weighting pattern by shifting to spectral distinctions. Results are discussed in terms of adaptations to cue-weighting strategies for intelligibility gains by interlocutors of different linguistic backgrounds.

2pSCb5. Examining the maintenance of dialect features in colloquial urban Armenian speech via variationist analysis of vowels in Gavaṙ, Armenia. Emma L. Portugal (Linguist, Univ. of Michigan, Lorch Hall, 611 Tappan Ave., Ann Arbor, MI 48109, esantelm@umich.edu)

This study examines the maintenance of previously described local dialect vowels in the regional city of Gavar̦, Armenia. The dialect vowel system contains additional phonemic and allophonic distinctions not found in supralocal Armenian varieties. As such, adherence to this system was conceptualized through the presence or absence of various vowel mergers. F1 and F2 were measured from picture and word list data collected from sociolinguistic interviews with 31 participants. Mergers were assessed for each participant via Pillai scores and Euclidean distances between clusters of words where different vowels were predicted to appear based on previous descriptions of the dialect. Countering previous claims about dialect leveling in regional cities, some participants were found to maintain some aspects of the dialect vowel system. Fixed effects linear models assessed the effect of demographic predictors (self-reported gender, birth year, education level) on maintenance of dialect vowels. There was an effect of gender, such that men were found to maintain some dialect vowels to a greater degree than women. Participants' commentary suggested that the most salient aspects of the dialect vowel system are the same ones in relation to which this demographic variation was uncovered, indicating a relationship between salience and maintenance of dialect vowels.

2pSCb6. Effects of formant peak flattening on vowel perception: A larger-scale web-based experiment. Filip Nenadić (Dept. of Linguist, Univ. of Potsdam, 3-28 Assiniboia Hall, Edmonton, AB T6G 2E7, Canada, nenadic@ualberta.ca), Dejan Sredojević (Dept. of Serbian Lang. and Linguist, Faculty of Philosophy, Univ. of Novi Sad, Novi Sad, Serbia), and Michael Kiefte (School of Commun. Sci. and Disord., Faculty of Health, Dalhousie Univ., Halifax, NS, Canada)

Ito *et al*. reported that suppressing the first or the second formant in Japanese synthesized vowels did not importantly change listener perception. However, quantitative analyses of data collected in the English (Nenadić *et al*., 2020) and the Serbian language (Nenadić *et al*., 2023) have shown that formant suppression increases response entropy between participants (i.e., decreases agreement in vowel identity). Previous studies never tested more than fifteen participants, but it was noted that certain listeners gave different responses to a manipulated synthesized vowel in comparison to its original version more often than others. We tested 118 native monolingual speakers of Serbian language (87% female, 13% male; age 18 to 44, M = 21.07, SD = 4.71) in an online replication of the experiment. The results again show that listeners agree less about vowel identity for stimuli with suppressed formants. Certain listeners again tended to change their response to the manipulated version of the vowel more often than others. Importantly, this tendency did not correlate with their mean response latency. We discuss possible causes for these findings, including differences in listener strategies and time requirements of a more careful processing of the incoming signal.

2pSCb7. Creaky voice across language and gender: A study of Canadian English-French bilingual speech. Jeanne Brown (Linguist, McGill Univ., 1085 Ave. du Docteur-Penfield #104, Montreal, QC H3A 1A7, Canada, jeanne.brown@mail.mcgill.ca) and Morgan Sonderegger (Linguist, McGill Univ., Montreal, QC, Canada)

This study addresses how non-contrastive creaky voice varies among languages and across speakers (as a function of gender). Spontaneous speech from 9 English-French bilingual speakers born and raised in Ontario/Québec was collected from publicly available online data sources, amounting to roughly 5 min of speech per speaker-language pair and 13 992 vowels total. This corpus will reach 40 speakers by the conference. Acoustic analysis consisted of pitch tracking in Praat, providing a proportion of unreliable f0 tracks for each vowel as well as one spectral slope measure (H1*-H2*) and two Harmonics-to-Noise Ratios (CPP and HNR05) as acoustic correlates of creaky voice. Statistical significance was tested using mixed-effects regression models, with fixed effects of language, gender, and utterance position, and maximal by-word and by-speaker random effects. The main results for gender show that men's vowels have more unreliable f0 tracks, lower H1*-H2*, lower HNR05 and somewhat lower CPP, suggesting that male speakers are creakier overall. Regarding language, English displays more unreliable pitch tracking compared to French, providing some evidence for language-dependent vocal settings. Other acoustic correlates of creak, however, do not show consistent cross-linguistic differences.

2pSCb8. Metrical segmentation across dialects. Natasha Warner (Linguist, Univ. of Arizona, Dept. of Linguist, Box 210025, Tucson, AZ 85721-0025, nwarner@email.arizona.edu), Ki Woong Moon (Linguist., Univ. of Arizona, Tucson, AZ), Seongjin Park (Linguist., Univ. of Arizona, Princeton, NJ), James M. McQueen (Donders Inst. for Brain, Cognition and Behavior, Radboud Univ., Nijmegen, Netherlands), and Mohammed K. Albusairi (Linguist., Univ. of Arizona, Tucson, AZ)

Norris, McQueen & Cutler (1995) tested the Metrical Segmentation Strategy (MSS; Cutler & Norris, 1988) as part of the spoken-word recognition model Shortlist, using British English stimuli and listeners. We replicate their study using American English listeners, who we exposed to one of two sets of stimuli. One group heard a new set of stimuli recorded in American English, while the other was exposed to the original British English recordings. Norris *et al*. used a word-spotting task: listeners had to spot words within speech (e.g. "stamp" in [stæmpɪdʒ]). Target words were CVCC (like "champ") or CVC (like "done"), and were followed by a full vowel (e.g. /tʃæmpoʊʃ/) or a reduced vowel (e.g. /tʃæmpəʃ/). The original study found different behavior for CVCC versus CVC targets, with the results suggesting that listeners hypothesize a word onset at the start of a full-vowel strong syllable (the MSS). The results for the current study with American stimuli partially replicate the original findings, showing even more consistent support for the MSS. The results with British stimuli also support the MSS, but with higher error rates. The results indicate that the MSS has a very strong effect even in the difficult setting of cross-dialectal perception.

2pSCb9. An acoustic analysis of French contrast-specific clear-speech productions. Morgan Robertson (Linguist, Univ. of Kansas, 1541 Lilac Ln., Lawrence, KS 66045, morgan.robertson@ku.edu) and Allard Jongman (Linguist, Univ. of Kansas, Lawrence, KS)

Clear speech provides insight into acoustic correlates speakers alter when attempting to increase intelligibility or enhance phonemic contrasts. The present study consists of an acoustic analysis of clearly spoken French voiced and voiceless stops, nasal and oral vowels, and front rounded and unrounded vowels. Productions were elicited using a simulated interaction between French-speaking participants and a computer program, in which participants produced casual tokens and the program 'guessed' with controlled responses. When the program incorrectly responded with "???" (indicating 'What did you say?') or a specific competitor (e.g., response *reins* [ʁɛ̃] 'kidney' to target *reine* [ʁɛn] 'queen'), participants reproduced the word more clearly. The program's responses probed at differences in speakers' productions when responding to specific competitors versus globally attempting to increase intelligibility. Preliminary results show a general increase in prevoicing for clearly produced voiced stops. Speakers also adjust coarticulatory nasality to disambiguate CVN targets from CṼ competitors. Finally, expansion is exhibited between front rounded and unrounded vowel pairs in the F2/F3 vowel space from casual to clear; however, it is largest when the program's response is "???". Overall, findings suggest that speakers modify their speech differently in response to a specific competitor versus the general "???" response.

2pSCb10. The correlation between acoustic and articulatory variation in Laurentian French high vowels. Beth MacLeod (School of Linguist & Lang. Studies, Carleton Univ., 1125 Colonel By Dr., Ottawa, ON K1S 5B6, Canada, Beth.Macleod@carleton.ca), Suzy Ahn, and Phillip Burness (Dept. of Linguist., Univ. of Ottawa, Ottawa, ON, Canada)

This study explores the relationship between acoustic and articulatory variation in Laurentian French (LF) high vowels. LF /i/, /y/, /u/ undergo laxing before a consonant other than a voiced fricative. While several studies have characterized LF vowel laxing acoustically, limited work has described it articulatorily. This study investigates the alignment between ultrasound tongue imaging data and acoustic realization in LF vowels. Using data from Burness *et al*. (2022b), seven native LF speakers participated in a study collecting ultrasound tongue imaging and audio while producing 52 French words with target vowels /i/, /y/, /u/. We chose the ultrasound frame for each token that aligns with the vowel midpoint. From these, we obtained x (tongue backness) and y (tongue height) values. Linear mixed-effect models were employed to assess the relationship between acoustic parameters (F1, F2) and articulatory measures. Findings reveal a varied but generally related pattern between acoustics and articulation. F1 relates significantly to tongue height for /u/ and /y/, while F2 aligns significantly with tongue backness across all three vowels. However, substantial unaccounted variation suggests factors like vocal tract physiology, the non-linearity between articulation and acoustics, and specific choice of articulatory measures might contribute to this variability. Future research will explore these factors to better comprehend this intricate link between acoustics and articulation.

2pSCb11. Stories and words online regional dialect corpus collection. Kevin D. Lilley, Jory P. Ross, Marie Bissell (Ohio State Univ., Columbus, OH), and Cynthia G. Clopper (Ohio State Univ., 1712 Neil Ave., Columbus, OH 43210, clopper.1@osu.edu)

Online data collection allows for access to diverse populations. In the current study, we used online recruitment and data collection methods to obtain a corpus of read short stories, real English words, and nonwords from adult talkers representing three authentic regional dialects of American English and one novel accent. The authentic dialects are New England, Northern, and Southern American English and are each represented by 8–10 talkers, ranging in age from 22 to 75 years old. The novel accent was produced by five Spanish-English bilinguals with training in linguistics, who were asked to produce Spanish /o/ in an otherwise English segmental context. The four target varieties each contain one vowel pair of interest, in which the vowels within the pair are relatively more ambiguous than in the other varieties. Each talker produced one familiar short story (e.g., Goldilocks and the Three Bears) with 40 tokens of each vowel within the target pair for their dialect, as well as a set of real words and nonwords that represent both the target vowel pair for their dialect and the other three vowel pairs for comparison across dialects. All corpus materials are available to the scholarly community.

2pSCb12. Harmony in transition: Exploring the perception-production relationship in sound change. Felix Kpogo (Dept. of Linguist, Boston Univ., 621 Commonwealth Ave., Rm. 116, Boston, MA 02215, fkpog001@bu.edu)

This study investigates the perception-production dynamics of [æ] and [e] in an ongoing merger in Asante Twi's Advanced Tongue Root (ATR) harmony system. Traditional descriptions state that [-ATR] /a/ is realized as [+ATR] [æ] before [+ATR] vowels /i, u, o/. Yet, recent acoustic evidence points to an ongoing sound change in different Twi-speaking communities: in contrast to traditional (rural/suburban) Twi speakers, urban speakers raise and merge [æ] with phonemic /e/ before /i, u/, a change that is more advanced in men and younger speakers (Kpogo, 2023). The present study aims to ascertain whether merger in production correlates with perceptual merger among the same participants in Kpogo (2023). Data from a forced-choice identification task suggests that the presence of a perceptual merger is associated with the participant's locality. Traditional participants consistently identified [æ] and [e] correctly, suggesting separate mental representations for these vowels. Conversely, urban participants often perceived [æ] as [e], suggesting less distinct representations influenced by their own production. Concerning the perception-production relationship, evidence of a link was observed across localities but, crucially, not within each locality. This study contributes to theoretical discussions on the perception-production relationship during sound change, hinting at a co-evolving link between the two modalities.

2pSCb13. Vowel space expansion following alcohol intoxication. Arian Shamei (Linguist, UBC, 2613 West Mal, Vancouver, BC V6T 1Z4, Canada, arianshamei@gmail.com), Xinglei Liu, Rima Seiilova (Tenvos Res. Labs, Sacramento, CA), and Bryan Gick (Linguist, UBC, Vancouver, BC, Canada)

Alcohol intoxication is characterized by hypermetric movements (overshoot) in manual fine motor skills [Phillips *et al*., (2009). Hum. Movement Sci,. 28(5), 619–632]. It remains unknown whether hypermetric movements manifest in speech following alcohol intoxication. Recent work on a small population ($n = 15$) revealed vowel space area (VSA) expansion following intoxication [Chang *et al*., 2023. Proc. ICPhS '23]. To validate these

findings, we made use of the publicly available Alcohol Language Corpus [Schiel *et al*., 2012. Lang. res. and eval. 46, 503–521]. VSAs were compared across 162 speakers (85 male, 77 female) while sober (n = 162) and intoxicated (blood alcohol level > 0.08%, n = 97) using an unsupervised method based on cluster center detection of normalized vowel formants [Sandoval *et al*. JASA. 134(5), EL477-EL483]. Substantial VSA expansion was observed for males (10.8%), while a smaller expansion was observed for females (3.6%). These results support recent observations of VSA expansion and suggest hypermetric tongue movement following alcohol intoxication. Further evaluations are ongoing and employ hierarchical linear modeling to compare the effects of specific blood-alcohol concentrations and speech tasks (free speech, tongue-twisters, repetition) on VSA expansion. [Research funded by Tenvos Incorporated for the development of commercial speaker state-detection algorithms.]

2pSCb14. Patterning of laxing spreading in Quebec French. Philippe Aigner-Therrien (Linguist, Carleton Univ., Ottawa, ON K2C3L5, Canada, philippeaignertherri@cmail.carleton.ca)

Quebec French laxing is a rather well-known phonological process, starting from the tense vowels [i], [y], and [u] being rendered lax in some closed syllables. This process can also trigger laxing of high vowels in preceding syllables as well, though the nature of this process remains a source of debate, whether it is a spreading process or vowel harmony. In addition to the nature of the process itself, the patterning that occurs for the laxing spreading is one that has been long discussed, particularly when it comes to words with more than two syllables. Poliquin (2006) lists three different possible patterns that the laxing may take. The goal of this thesis was to not only confirm the presence of the spreading patterns found in Poliquin(2006), but also to determine to what degree they occur within native speakers of Quebec French. While the results seem to confirm the presence of laxing spreading, the frequency at which they occur seems to be rather low and vary not only from speaker to speaker, but also within the same speaker.

2pSCb15. A descriptive study of vowels in Dialects of Pakistani English. Mumtaz Yaqub (The Dept of English Lang. and Appl. Linguist, AIOU, Islamabad-Pakistan, Dept. of Linguist, University of Arizona, Tucson, AZ 85719, mumtazyaqub@arizona.edu) and Muhammad K. Khan (The Dept of English Lang. and Appl. Linguist, AIOU, Islamabad-Pakistan, College Park, MD)

The widespread use of English as a lingua franca has created many varieties of Global English. Pakistani English is an under-documented variety. As Pakistan is a multilingual country, it has been a difficult task for the linguists to document the dialects of Pakistani English and to determine how these dialects are influenced by regional languages (e.g. Pashto, Punjabi, Urdu etc.). Previous studies focused on Pakistani English of only a few locations (influenced by only a few regional languages). A broad-based study is required to document the phonetics of Pakistani English. The current study investigates two questions about the Pakistani English vowel system: how much variety there is, and whether the dialects of Pakistani English are influenced by regional languages. Speech data was collected from 208 undergrad students from thirteen cities representing thirteen regional languages of Pakistan (e.g. Lahore, Karachi, Peshawar etc.). Participants read thirty-two words expected to contain monophthongs in initial, medial and final position. Formant measurements are measured automatically in Praat. Preliminary analyses show variation in formant values of vowels in regional dialects; The results will allow for documentation of the vowel space of Pakistani English and will provide a broad representation of Pakistani English dialects.

2pSCb16. American English diphthong transitions and the acoustic vowel space: A pilot comparison of two tasks. Christina Kuo (James Madison Univ., 235 Martin Luther King Jr. Way, MSC 4304 James Madison University, Harrisonburg, VA 22807, kuocx@jmu.edu)

The purpose of the present study is to evaluate the potential relationship(s) between diphthong transitions and the acoustic vowel space parameters in two speaking tasks. Diphthongs are associated with acoustic changes that reflect changes in vocal tract configurations, affording unique opportunities for investigating the articulatory-acoustic link. The acoustic vowel space, a representation of monophthong vowels in a two-dimensional plane defined by the first and second formant frequencies (F1 and F2), has been widely used to interpret articulatory-acoustic consequences and to index articulatory integrity. Nevertheless, limited is known as to how diphthongs and monophthong vowels account for the acoustic working space together [Lee *et al*., JASA 136(4), 1880–1894 (2014)]. To add to this inquiry, transitions of American English diphthongs /ei/, /ɑɪ/, /ɑʊ/, /oʊ/, and /ɔi/, defined by the primary F2 transition, are examined with respect to the quadrilateral vowel space anchored by /i/, /æ/, /ɑ/, and /u/. The speech sounds were obtained from four male speakers in two tasks, sentence reading and connected speech, that have been shown to elicit vowel space changes. It is hypothesized that greater diphthong transitions are associated with parameters corresponding to a larger vowel space. Findings will be discussed within the framework of the acoustic theory.

2pSCb17. Perceptual adaptation to unfamiliar dialect variation. Kevin D. Lilley (Linguist, The Ohio State Univ., 1712 Neil Ave., Columbus, OH 43210, lilley.35@osu.edu) and Cynthia G. Clopper (Ohio State Univ., Columbus, OH)

Regional varieties of a language often differ in the phonetic realization of shared phonological categories, resulting in perceptual ambiguity. In perceptual tasks, listeners with more exposure to regional dialect variation perform differently than listeners with less exposure, reflecting underlying differences in lexical competition. In this study, we examined perceptual adaptation to one authentic dialect of American English and one novel dialect to test for adaptation to unfamiliar variation. Our materials contained contrasts that are perceptually ambiguous in Southern American English and in the novel dialect we created, in which linguistically trained bilinguals substituted English /ow/ with Spanish /o/. Participants heard a familiarization passage spoken in a Southern or Novel dialect and then completed an auditory lexical decision task. Adaptation was assessed by differences in accuracy and response times for each vowel contrast, talker dialect, and familiarization dialect. Results revealed lower accuracy and slower response times for both target contrasts when pronounced in their respective dialects. These effects reflect the difficulty in processing perceptually ambiguous stimuli. Familiarization dialect had no effect on accuracy or response times; future analyses will include quantification of listener regional dialect exposure to better assess its influence on processing novel variation.

2pSCb18. Monophthongization of Diphthongs in Southern American English: A perception study. Ratree Wayland (Linguist, Univ. of Florida, 4131 Turlington Hall, P.O. Box 115454, Gainesville, FL 32611-5454, ratree@ufl.edu), Harys Dalvi (Comput. Sci., Brown Univ., Providence, RI), and Rachel Meyer (Linguist, Univ. of Florida, Gainesville, FL)

The more prevalent monophthongization of /ai/ compared to /au/ in certain dialects, particularly in Southern American English, can be attributed to several factors. Historically, /ai/ has been more susceptible to dialectal variation and change. Phonetically, the tongue movement required for /ai/ (from a low to a high front position) may be more easily simplified or reduced than the movement for /au/ (from a low to a high back position). Sociolinguistic factors might also play a role, with certain vowel changes becoming markers of regional or social identity, leading to their widespread adoption in speech. This study examines the monophthongization of the diphthongs /ai/ to /a/ and /au/ to /a/ in Southern American English. Two 11-step continua, [a-ai] and [a-au], were presented to listeners for identification. The hypothesis is that a higher incidence of /a/ will be perceived in the /a-ai/ continuum than in the /a-au/ continuum among Southern dialect speakers compared to non-Southern dialect speakers. The expected results suggest that a significant tongue position shift in the front region of the oral cavity (for the /i/ in /ai/) leads to a smaller perceptual shift than a similar shift in the back region (for the /u/ in /au/).

2pSCb19. Neural network-based measure of voice quality in Parkinson's disease. Ratree Wayland (Linguist, Univ. of Florida, 4131 Turlington Hall, P.O. Box 115454, Gainesville, FL 32611-5454, ratree@ufl.edu), Kevin Tang (English Lang. and Linguist, Heinrich-Heine-Universität Düsseldorf, Düsseldorf, Germany), Sophia Vellozzi (Comput. Sci., Univ. of Florida, Gainesville, FL), Rachel Meyer (Linguist, Univ. of Florida, Gainesville, FL), and Rahul Sengupta (Comput. Sci., Univ. of Florida, Gainesville, FL)

Parkinson's Disease (PD) significantly affects speech and voice. The Parkinsonian voice is often described as breathy, rough, hoarse, tremulous, abnormally pitched, having reduced pitch range, and unusually quiet. Changes in voice quality result from altered neurological controls of the muscles in the respiratory and phonatory systems, impacting breath support and vocal fold vibration. This study investigates the effects of PD on voice quality (e.g., breathiness) in vowel production among native Spanish speakers. The degree of breathiness is estimated from posterior probabilities calculated by recurrent neural networks trained to recognize spread glottis phonological features in Gujarati, a language contrasting breathy and modal voicing in vowels and between voiced aspirated (breathy voice) and voiceless aspirated versus plain voiced stops in consonants. It is hypothesized that vowels produced by PD patients will exhibit a higher degree of breathiness than those produced by normal controls, with degrees of breathiness potentially varying as a function of disease progression.

2pSCb20. Effects of exposure to dialect-specific allophonic variation on perceptual similarity ratings of vowels. Marie Bissell (Ohio State Univ., 4229 Avent Ferry Rd., Apt 3, Raleigh, NC 27606, marie.bissell@gmail.com) and Cynthia G. Clopper (Ohio State Univ., Columbus, OH)

Dialect exposure affects listeners' lexical processing. In this study, we examined whether listeners' exposure to dialect-specific allophonic variation in two linguistic variables in American English (nasal split /æ/ system and pre-voiceless /ɑɪ/ raising) affects their perceptual similarity ratings of matching versus mismatching allophones. Listeners from the U.S. North were assumed to have more exposure to pre-voiceless /ɑɪ/ raising and less exposure to a nasal split /æ/ system, while listeners from the U.S. Midland were assumed to have more exposure to a nasal split /æ/ system and less exposure to pre-voiceless /ɑɪ/ raising. We analyzed Northern and Midland listeners' perceptual similarity ratings on a 1–5 scale for three types of auditory vowel pairs: match (e.g., *ban-bang* or *bike-bite*), allophonic mismatch (e.g., *ban-bat* or *bike-bide*), and phonemic mismatch (e.g., *ban-bike*). We predict that listeners with more exposure to dialect-specific allophonic variation should rate allophonic mismatches as more different than allophonic matches compared to listeners with less exposure. Preliminary results suggest that listeners rate allophonic matches as more similar than allophonic mismatches, and both of these types are rated as more similar than phonemic mismatches, consistent with existing literature. There was no effect of dialect exposure to allophonic variation on perceptual similarity ratings.

2pSCb21. The effect of aging on vowel production in Western Canadian English. Brooklyn Jones (Northern Arizona Univ., 208 E Pine Knoll Dr., Flagstaff, AZ 86011, bj587@nau.edu) and Benjamin V. Tucker (Commun. Sci. and Disord., Northern Arizona Univ., Flagstaff, AZ)

Aging influences many aspects of speech production. For example, a speaker's fundamental frequency or formant space has been reported to change as they age. However, this research has largely focused on careful or prepared speech, with a few notable exceptions in the longitudinal research literature. There is limited knowledge of the relationship between aging and vowel reduction in spontaneously produced speech. It is important to investigate how spontaneous everyday speech is influenced by the aging process. In the present investigation, we work with a corpus of Western Canadian English containing spontaneous conversational speech produced by 18 younger (17–31) and 13 older speakers (64–79). The data was force-aligned and then acoustic measures were extracted from the aligned data. In this presentation, we report on the differences in formant space for stressed and unstressed vowels. The results of this study will be used to better inform our understanding of the role of vowel centralization and aging in spontaneous speech. They will further support the understanding of reduction in spontaneous speech.

2pSCb22. Binaural fusion and vowel perception in cochlear implant users. Kayla Pierre (Commun. Sci. and Disord., Penn State Univ., 308H Ford Bldg., University Park, PA 16802, kaylapierre0124@gmail.com) and Lina A. Reiss (Otolaryngol. & Biomedical Eng., Oregon Health & Sci. Univ., Portland, OR)

Excess binaural fusion can be a problem, especially in noisy environments when speech from multiple talkers are fused together. Previously, we showed that hearing aid users experience abnormally broad binaural fusion that leads to fusion and averaging of dichotic vowels of fundamental frequency (F0) differing by up to 1 octave (Reiss and Molis 2021). This study investigated whether cochlear implant (CI) listeners also experience abnormal fusion and averaging of double vowels in the free field. Six adult CI users (4 females) and four NH adults (1 female) were tested. Stimuli were synthetic vowels/æ/, /a/, /u/ and /i/, at F0 = 106.9, 151.2, and 201.8 Hz. Two different vowels were presented simultaneously from two loudspeakers at $\pm 60^{\circ}$, with same or different F0. Subjects were instructed to select which one or two vowels they perceived. When $\Delta F0 = 0$, NH and CI users often fused and identified only one vowel. As $\Delta F0$ increased, NH listeners increased identification of two vowels, while CI users continued to fuse vowels. The findings show that in realistic free field listening conditions, CI users can experience excess fusion of multiple vowels of differing F0s, presenting additional challenges in noisy listening environments. [Work supported by NIH R01DC013307 and ASA SUREIA program.]

2pSCb23. Quantifying vowel category distinctness using Bayesian modelling. Irene Smith (Linguist, McGill Univ., 1085 ave. du Docteur-Penfield, Montreal, QC H3A1A7, Canada, irene.smith@mail.mcgill.ca) and Morgan Sonderegger (Linguist, McGill Univ., Montreal, QC, Canada)

Phonetic and sociolinguistic studies of vowel merger require a measure of acoustic distinctness between vowel categories. Three desiderata for such a metric are that it be multivariate, in the sense that it account for correlations between dimensions (e.g., F1 and F2), control for other factors affecting vowel formants (e.g. surrounding consonants), and work for unbalanced data (common in naturalistic data). Previous work (Nycz and Hall-Lew, 2013; Kelley & Tucker, 2020) has considered a variety of measures, including variants of Euclidean distance, Pillai score, and Bhattacharyya affinity, but none meet all three criteria. We present a new method for quantifying vowel merger that meets all desiderata and can be applied to different metrics: we fit a Bayesian mixed-effects linear model to jointly predict F1 and F2, then compute any desired metric—here, Euclidean distance, Pillai score, and Bhattacharyya affinity—from the posterior. We evaluate each metric, and the overall method, to describe PIN-PEN across a range of English dialect corpora. We find that controlling for covariates and unbalanced data adds substantial signal, but that multivariate modelling does not perform substantially better than univariate modelling. Additionally, we argue that Bhattacharyya affinity has particularly desirable properties (e.g., allowing for heteroskedastic data: Johnson, 2015).

2pSCb24. Vowel quantity/quality redux: Revisiting vowel contrasts in Hakha Lai monophthongs. Kelly H. Berkson (Linguist, Indiana Univ., Bloomington, 1020 E. Kirkwood Ave., Ballantine Hall 540, Bloomington, IN 47405-2201, kberkson@indiana.edu), Grayson Ziegler, Amanda Bohnert (Linguist., Indiana Univ., Bloomington, IN), and Kenneth Van Bik (Dept. of English, Comparative Lit., and Linguist, California State Univ., Fullerton, Fullerton, CA)

Previous scholars have documented a phonemic length distinction in the monophthongal vowels of Hakha Lai, a Tibeto-Burman language spoken in Chin State in western Myanmar (Melnik, 1997; Peterson, 2003; Maddieson, 2004). In a previous pilot study of data from two college-aged speakers (the first instrumental acoustic analysis of Hakha Chin vowel quality), our findings revealed stark individual differences: one speaker showed robust quantity and minor quality differences, while for the other, neither duration nor quality played key roles; rather, her system had merged. Maddieson (2004) proposed that the length associated with long vowels in Hakha Lai is mainly realized through lengthening of sonorant codas; we did not find this to be the case for either speaker. However, those pilot data were limited both in terms of sample size and age range. Therefore, to expand on our previous findings, we now present analysis of data from fourteen talkers, including

both a group of nine college aged talkers (6 women) and five older talkers (2 women, ages 46 and 77; 3 men, ages 48, 60, and 62). Duration and quality measures are reported for all monophthongs in all possible syllable shapes.

2pSCb25. Vocalic contrasts in Zotung. Amanda Bohnert (Linguist, Indiana Univ., Bloomington, IN), Kelly H. Berkson (Linguist, Indiana Univ., Bloomington, 1020 E. Kirkwood Ave., Ballantine Hall 540, Bloomington, IN 47405-2201, kberkson@indiana.edu), Kenneth Van Bik (Dept. of English, Comparative Lit., and Linguist., Cal State Fullerton, Fullerton, CA), and Grayson Ziegler (Linguist., Indiana Univ., Bloomington, IN)

Though very few Maraic languages from the South Central branch of Tibeto-Burman are represented in the existing body of phonetics literature, they are of interest in that they appear to contain typologically unique dense high vowel systems. Zotung, spoken by approximately 100 000 people in Chin State in western Burma and in diaspora communities, is one such language (Eberhard *et al*., 2022). The only linguistic scholarship on Zotung (Shintani, 2015) is an extensive wordlist with brief preliminary comments on the syllable shape, tone, and vowel inventories; however, acoustic analyses are absent. To augment Shintani (2015) and further elucidate the vowel inventory of Zotung, we present data from an older male speaker deeply involved with Zotung translation, preservation, and literacy efforts, who hails originally from Tingsi on the far southern border of the Zotung speaking area. Based on analysis of 553 tokens collected via a wordlist, we substantiate some of the findings reported by Shintani (2015) and document several important differences. Shintani reported 8 monophthongal vowel qualities; we find 10: [i, y, e, ø, æ, ə, a, u, o, ɔ]. Shintani also reported an expansive diphthong inventory; our data support this claim, with 9 total vowels that can be categorized as diphthongs.

2pSCb26. An ultrasound investigation of Hnaring Lutuv (high?) vowels. Grayson Ziegler, Amanda Bohnert (Linguist, Indiana Univ., Bloomington, IN), Kelly H. Berkson (Linguist, Indiana Univ., Bloomington, 1020 E. Kirkwood Ave., Ballantine Hall 540, Bloomington, IN 47405-2201, kberkson@indiana.edu), and Sui Hnem Par (Linguist, Indiana Univ., Bloomington, IN)

Lutuv (also known as Lautu) is an under-documented Chin language from the Tibeto-Burman language family spoken by 18,000 people, both in Chin State in western Burma and in diaspora communities worldwide, including approximately 1000 people in the Indianapolis Chin refugee community. Lutuv utilizes a typologically rare six-way contrast in the higher part of the vowel space (i y ɨ ʉ ɯ u, see Bohnert *et al*. 2022), with an additional four high diphthongized vowels (ie̯ yə̯ ɯə̯ uo̯). Previous work has also identified that the high central vowels (/ɨ ʉ/) are poorly disambiguated—acoustically, they show considerable overlap with both each other and the high back vowels, and in terms of lip posture, they do not display the characteristics of a typical rounding contrast (Bohnert & Berkson 2023). The present work utilizes 3D ultrasonography to provide detailed lingual articulatory data of Lutuv vowels with special attention paid to the high central vowels, adding a new dimension to the existing acoustic and articulatory data. Real-time images of tongue position and motion provide new insights into the complex articulatory gestures involved in the production of these sounds and constitute the first ultrasound investigation into this underdocumented language.

2pSCb27. Acoustic features of running speech by Chinese college student: Effects of local dialect and second language. Yunzi Wan (Dept. of English, Chengdu Inst., Sichuan Int.. Studies Univ., School of English Studies, Chengdu Inst. Sichuan Int.. Studies University, Chegdu, Sichuan 611844, China, 463445683@qq.com), Wei Hu (Dept. of Psych., Tianjin Normal Univ., Tianjin, China), and Chang Liu (Univ. of Texas at Austin, Austin, TX)

The goal of this study was to investigate the acoustic features of running speech recorded by college students in China in three languages: standard Mandarin, Mandarin dialect (Sichuan dialect), and English (e.g., the second language). Acoustic analyses were focused on spectral features (e.g., vowel space and spectral tilt), temporal features (e.g., temporal envelope), and speaking rate. Preliminary data analysis indicated that for Chinese college students, the speaking rates of their native languages: standard Mandarin and Sichuan dialect were comparable and significantly faster than English. In addition, the vowel space (e.g., /a-u-i/) was the largest for standard Mandarin and the smallest for Sichuan dialect with English in between. Temporal properties of speech like the dominant temporal modulation frequency and temporal modulation depths will be compared and discussed across the three languages.

2pSCb28. Exploring infant talker bias: Insights from remote speech perception testing. M. Fernanda Alonso Arteche (School of Commun. Sci. and Disord., McGill Univ., 1295 Rue des Carrieres, apt 402, Montreal, QC H2S 0E1, Canada, maria.alonsoarteche@mail.mcgill.ca), Nicola Phillips, Samin Moradi, Lulan Shen, Marianne Chen-Ouellet, Leatisha Ramloll, Sumana Abraham, Lei Zeng (School of Commun. Sci. and Disord., McGill Univ., Montreal, QC, Canada), Lucie MENARD (Linguist, UQAM, Montréal, QC, Canada), and Linda Polka (School of Commun. Sci. and Disord., McGill Univ., Montreal, QC, Canada)

Lab studies show that infants (4- to 7-month-olds) prefer to listen to vowels with infant-like f0 and formant frequencies over those of an adult female (Masapollo *et al*., 2015; Polka *et al*., 2021). This *Infant Talker Bias* may facilitate infants' mapping of articulatory gestures to acoustic correlates. In this study, 4- to 12-month-olds completed a listening preference task on the *Lookit* online testing platform. Across eight trials, we presented synthesized infant and adult vowel sounds (/i/ and /a/) paired with a simple animation and recorded the infant's response via the webcam. Infant looking time and vocalization to each vowel type were coded offline. Preliminary analyses ($n = 91$) show that listening time increased with age ($p < 0.05$), and all infants listened longer to infant vowels than to adult vowels ($p < 0.01$). Preliminary analyses ($n = 62$) also show an increase in infant vocalizations with age ($p = 0.00$) and a trend towards more and longer vocalizations in response to the adult vowels. These findings replicate and extend the infant talker bias to new vowel stimuli and to older infants, and support the use of remote testing in infant speech perception studies.

Session 2pSP

Signal Processing in Acoustics, Biomedical Acoustics, Physical Acoustics, and Underwater Acoustics: Denoising and Enhancing Acoustic Signals

Kendal Leftwich, Cochair
Physics, University of New Orleans, 1021 Science Building University of New Orleans, New Orleans, LA 70148

Shaun Pies, Cochair
Physics, University of New Orleans, 2000 Lakeshore Dr., New Orleans, LA 70148

Juliette W. Ioup, Cochair
Department of Physics, 2000 Lakeshore Dr., New Orleans, LA 70148

Chair's Introduction—3:50

Contributed Papers

3:55

2pSP1. Balancing ensemble averaging and signal stationarity when processing acoustic data recorded during a rocket launch. Carson F. Cunningham (Phys., Brigham Young Univ., Brigham Young University, Provo, UT 84602, carsonfcunningham@gmail.com), Micah Shepherd (Brigham Young Univ., Provo, UT), and Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

There are many important details to consider when collecting acoustical data during a rocket launch including system noise floor, dynamic range and transducer sensitivity. In this talk, the choice of signal processing parameters will also be shown to be important. Since the pressures are created by turbulence, ensemble averaging is key to reducing random variation. As such, the use of a Hanning window with 50% overlap would be customary for performing this type of processing. However, as the rocket lifts off, the source location moves, and the data can only be considered stationary for short blocks of time. Therefore, the ability to average is limited by non-stationarity. To balance these competing phenomena, an adjustable windowing function such as the Tukey window may be more appropriate than Hanning. Noise data collected at two different launches will be used to compare of the Hanning window with 50% overlap to that using of the Tukey window with a taper of $\alpha = 0.25$, which filters 12.5% of the signal at each extremity, with 87.5% overlap. The use of both window types will be compared when estimating auto spectral densities and integrated overall levels. The effect of blocksize will also be determined.

4:10

2pSP2. A lucky covariance estimator using cumulative coherence. Daniel J. Brooker (U. S. Naval Res. Lab., Washingtron, DC) and Geoffrey F. Edelmann (U. S. Naval Res. Lab., 4555 Overlook Ave. SW, Code 7145, Washington, DC 20375, edelmann@nrl.navy.mil)

This talk demonstrates a technique to improve covariance estimation using the principles of lucky signal processing and the cumulative coherence. Lucky processing, popularized in astro-photography, is a technique that increases signal quality by selectively keeping only a small fraction from a pool of potential snapshots. Cumulative coherence, a measure of how well a set of vectors is described by its subsets, provides the measure of "data quality" that enables the lucky processing. This approach was applied to covariance estimation on an acoustic array by taking a fixed duration sample of data and creating a dense set of snapshots with higher than usual overlap. From these densely sampled snapshots, the "luckiest" ones were found using cumulative coherence, and the covariance was averaged as normal. It was found that the lucky covariance estimate was successful at adaptive matched field processing and produced a less ambiguous processor output than the conventional estimator. The lucky covariance estimate had a higher estimated signal-to-noise ratio, especially when the source was at longer ranges from the array. [This work is supported by the Office of Naval Research.]

4:25

2pSP3. Reciprocal sensitivity kernels for receiver positioning. Alexis Bottero (DGA, Av. De la tour Royale, Toulon 83000, France, alexis.bottero@gmail.com)

In their classical formulation, sensitivity kernels aim to evaluate the points where an infinitesimal variation in the properties of the medium would most affect a pressure (or displacement) measurement at a fixed location. These kernels feature typical cigar shapes, with characteristically low sensitivity on the direct path. In this presentation, the same approach is applied to the reciprocal problem, where we want to evaluate, at a constant source position, which points are the most (or least) sensitive to a given model change. These reciprocal kernels exhibit flared shapes centered on the areas where the model is perturbed. Applications to receiver positioning in acoustic are discussed.

4:40

2pSP4. Comparison of conventional and adaptive acoustic beamforming algorithms using a tetrahedral microphone array in noisy environments. Megan B. Ewers (Elec. and Comput. Eng., Portland State Univ., 1900 SW 4th Ave., Ste. 160, Portland, OR 97201, megan.ewers1@gmail.com), Adina Edwards, and Martin Siderius (Elec. and Comput. Eng., Portland State Univ., Portland, OR)

In situ acoustic measurements are often plagued by interfering sound sources that occur within the measurement environment. Both adaptive and conventional beamforming algorithms, when applied to the outputs of a microphone array arranged in a tetrahedral geometry, are able to capture sound sources in desired directions and reject sound from unwanted directions. Adaptive algorithms may be able to measure a desired sound source with greater spatial precision, but require more calculations and, therefore, computational power. A conventional frequency-domain phase-shift algorithm and a modified adaptive frequency-domain Minimum Variance Distortionless Response (MVDR) algorithm were applied to simulated and recorded signals from a tetrahedral array of omnidirectional microphones.

The algorithms are described mathematically and demonstrated on both deterministic and real-world sound data, to quantitatively validate and compare their performance and to provide listening examples of their outputs in a variety of acoustically replicated environments. [Work supported by Portland State University.]

4:55

2pSP5. Efficient methods for iterative ultrasound image reconstruction using L1 and L2 norm regularization. Marko Jakovljevic (Radiology, Massachusetts General Hospital, 101 Merrimac St., Boston, MA 02114, mjakovljevic@mgh.harvard.edu), Ettore Biondi (Div. of Geological and Planetary Sci., California Inst. of Technol., Pasadena, CA), and Anthony E. Samir (Radiology, Massachusetts General Hospital, Boston, MA)

Ultrasound image reconstruction can be posed as an inverse problem, with image pixels as parameters to a model of wave propagation that are estimated from raw ultrasound channel data. Such framework allows one to make assumptions about channel signals and image properties in form of regularization that can be used to improve reconstruction accuracy in the presence of electronic and acoustic noise. A simple model assumes spherical wave propagation from point sources in a homogeneous medium, similar to delay-and-sum (DAS) beamforming; solving such problems numerically can require many iterations and can result in blurred images when traditional L2 norm regularization is used. We use Fast Iterative Shrinkage Thersholding Algorithm (FISTA) to reduce the number of iterations to less than 10, and to apply L1-norm regularization to the data, which improves edge sharpness and image contrast. We demonstrate the concept using FIELD II simulated ultrasound signals from point, speckle, and anechoic targets with controlled levels of electronic and speckle noise. The FISTA-reconstructed point targets show reduced sidelobe levels by 15 dB compared to the traditional DAS image, and the reduction in full-width at half maximum by a factor of 2. We also implement iterative reconstruction in omega-k frequency domain, which allows factoring the forward and adjoint operators at each spatial frequency and paves the way for an intuitive and more memory efficient, multi-threaded implementation of the method.

TUESDAY AFTERNOON, 14 MAY 2024 — ROOM 215, 2:15 P.M. TO 5:15 P.M.

Session 2pUW

Underwater Acoustics: Geoacoustics of Marine Sediments

Alexandra M. Hopps, Cochair
Physics, Brigham Young University, Eyring Science Center, N-181, Provo, UT 84606

Gabriel R. Venegas, Cochair
Cntr. for Acoust. Res. and Edu. and Dept. of Civil and Environ. Eng., University of New Hampshire, 33 Academic Way, Room W137, Durham, NH 03824

Contributed Papers

2:15

2pUW1. Observations of spatial coherence from a multibeam subbottom profiler. Laura Brownstead (Acoust., Penn State, 201 Appl. Sci. Bldg., University Park, PA 16802, lgb5113@psu.edu) and Daniel C. Brown (Acoust., Penn State, State College, PA)

Sonar data from the Kongsberg SBP-29 (with a 64-channel receive array) aboard R/V Sally Ride is analyzed to judge the value of spatial coherence of signals reflected from the ocean floor. The multibeam sub-bottom profiler collected five days of sonar test data off the coast of Southern California over a variety of sediment types and marine geologies. The spatial coherence of measured signals is sensitive to different bottom types and to sediment layering deep below the ocean floor. Navigational data also permits comparison of sonar data to existing bathymetric maps of the ensonified floor. The flexibility of the transmit and receive arrays permit detailed study of potentially informative seafloor features and landmarks. The findings of primary interest are interpreted in the context of a bivariate Normal surface fit to characterize the spatial coherence of scattered signals and future work is reported based on statistical analysis of the fitted data.

2pUW2. Temporal change of seafloor scattering and its dependence on environmental parameters in shallow-water sandy sites. Jenna Hare (Ctr. for Coastal & Ocean Mapping, Univ. of New Hampshire, 33-1173 Wellington St., Halifax, NS B3H3A2, Canada, jenna.hare@unh.edu), Anthony P. Lyons, and Gabriel R. Venegas (Ctr. for Acoust. Res. and Education and Dept. of Civil and Environ. Eng., Univ. of New Hampshire, Durham, NH)

In the ocean, the performance of active sonar systems used for object detection and seafloor characterization can be affected when the acoustic properties of the seafloor change due to near-bottom hydrodynamics and biological activity. Determining the dominant environmental mechanisms and corresponding time scales that regulate seafloor scattering will increase our understanding of the performance of these remote-sensing applications. To this end, a high-frequency active acoustic system (operating at 38 kHz, 70 kHz and 200 kHz), a wave-sensing CTD, and a stereo camera were deployed on the seafloor in a series of experiments lasting from two weeks to five months. Seafloor scattering measurements were obtained in two shallow water locations in New Hampshire, USA: a wave-dominated site and a tidal current dominated site. Daily and weekly trends in mean scattered levels and the mechanisms causing their temporal variability are discussed. The temporal change in scattering as a function of angle is compared to the small-slope approximation model where seafloor roughness estimates were obtained using stereophotogrammetry.

2:45

2pUW3. Deep sediment characterization from very low-frequency features from merchant ships. Alexandra M. Hopps-McDaniel (Phys., Knobles Sci. and Anal., PO Box 2185, Laramie, WY 82073, mcdaniel.alexh@gmail.com), David P. Knobles (Phys., Knobles Sci. and Anal., Austin, TX), Tracianne B. Neilsen (Phys. and Astronomy, Brigham Young Univ., Provo, UT), Preston S. Wilson (Walker Dept. of Mech. Eng. and Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX), William Hodgkiss (Scripps Inst. of Oceanogr., San Diego, CA), and Jason D. Sagers (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

The very low-frequency noise from merchant ships provides a good wideband source to study the deep layers of the seabed. The nested striations which characterize ship spectrograms contain unique acoustic features where the waveguide invariant (β) becomes infinite. This occurs at frequencies between 20 and 80 Hz where pairs of modal group velocities are equal. The goal of this project was to identify the $\beta = \infty$ frequencies in ship noise spectrograms and use these frequencies to perform statistical inference for the deep layer sound speeds and thicknesses. The Seabed Characterization Experiment of 2022 on the New England continental shelf had three vertical line arrays strategically placed between two shipping lanes. The average water depth was 75 meters with less than one meter bathymetry change between the arrays. The results of this study are based primarily on five ships. There was a gradual shift in the $\beta = \infty$ frequencies between the three arrays, suggesting a gradual change in the deep sediment layers. [Work supported by Office of Naval Research.]

3:00

2pUW4. The effect of salinity on the rigidity and settling behavior of reconstituted water-saturated kaolinite sediments. Alicia Casacchia (Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, 204 E. Dean Keeton St., Austin, TX 78712-1591, acasacchia@utexas.edu), Megan Ballard, Kevin M. Lee (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Preston S. Wilson (Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Nicholas P. Chotiros (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

In sandy sediments, the salinity of the pore water only affects the sound speed of the mixture by way of affecting the sound speed of the pore water. With clay mineral and saltwater mixtures, the intergranular forces that depend on the salinity of the pore water affect the sediment acoustic properties beyond simply affecting the sound speed of the pore water. These dependencies were investigated via image analyses of prepared samples of two kaolinite clay types (Flat DS and RSA) via analyses of their settling dynamics, porosity, and rigidity as a function of pore water salinity. There exists an observable dependence on salinity to mixture porosity and rate of settling. In addition, three regimes of slip-stick dynamics are shown to exist at different salinity ranges: (a) an immediate transition to liquid-like behavior; (b) a viscoelastic transition to liquid-like behavior with long term creep; and (c) a delayed transition to viscoelastic behavior with long term creep. [Work supported by ONR.]

3:15–3:30 Break

3:30

2pUW5. The crucial role of granular packing structure in marine sediment acoustics. Abe Clark (Phys., Naval Postgrad. School, Monterey, CA), Derek R. Olson (Oceanogr., Naval Postgrad. School, 833 Dyer Rd., 313b Spanagel Hall, Monterey, CA 93943, olson.derek.r@gmail.com), and Andrew Swartz (Phys., Naval Postgrad. School, Monterey, CA)

Motivated by the acoustic properties of marine sediments, we study via discrete-element method simulations the frequency dependence of dispersion and attenuation in model marine sediments. By numerically solving for the motion of each grain in a packing, rather than solving a continuum wave equation, we find that the granular packing structure, which is not explicitly considered in existing sediment-acoustics models, plays a crucial role in determining how attenuation and wave speed vary with frequency. Prior work has typically postulated a wave equation that is motivated by grain-scale forces, including viscous effects from the interstitial fluid, without explicitly considering the disordered packing structure of the grains. However, the packing structure is known to produce complex, nonlinear behavior (e.g., during shear or quasistatic compression), and how it affects dispersion and attenuation is not known. We consider linearized forces between particles (i.e., springs and dashpots), and demonstrate that the disordered packing structure leads to emergent scaling laws that do not agree with PDE-based approaches. Our results demonstrate that the granular packing structure must be explicitly considered when constructing theories for the acoustic properties of marine sediments.

3:45

2pUW6. Model manifold of transmission loss from a Pekeris waveguide using VGS parameters. Michelle Wang (Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84604, msw1998@byu.edu), Tracianne B. Neilsen, and Mark K. Transtrum (Phys. and Astronomy, Brigham Young Univ., Provo, UT)

In geoacoustic inversion, selecting an appropriate seabed parametrization, especially with an unknown number of sediment layers, is a challenge that is compounded by potential bias when establishing bounds in the parameter search space. One approach to addressing these issues is rooted in the techniques of Information Geometry. Information Geometry informs model selection and parameterization by quantifying which model parameters are informed by observational data. This paper provides an information geometric analysis of the Pekeris waveguide, where the acoustic properties of the lower half-space are derived from the viscous grain-shearing (VGS) model. Specifically, we consider single frequency transmission loss (TL) across a wide range of VGS parameters. By exploring the limits and boundaries of the geometric manifolds, particularly as parameters approach both low and high extremes, this approach provides indications of parameter hierarchies and correlations. Results will include slices of the model manifold and an evaluation of the boundary structure, providing insight into the relative impact of VGS parameters and the delineation of limiting regions. In doing so, this paper seeks to inform model selection and parameterization in geoacoustic inversion studies. [Work supported by the Office of Naval Research. Grant N00014-21-S-B001.]

4:00

2pUW7. Biogeoacoustic variability in muddy ocean bottom sediment. Kevin M. Lee (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758-4423, klee@arlut.utexas.edu), Megan Ballard (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Andrew R. McNeese (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Gabriel R. Venegas (Dept. Civil and Env. Eng. and Ctr. Acoust. Res. Ed., Univ. of New Hampshire, Durham, NH), Jason Chaytor (U.S. Geological Survey, Woods Hole, MA), Kelly M. Dorgan (Dauphin Island Sea Lab, Dauphin Island, AL), and Preston S. Wilson (Walker Dept. Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Both physical and biological benthic processes can influence seabed heterogeneity and contribute to spatiotemporal variability in geoacoustic properties. In particular, how biological processes affect both sediment acoustic properties and their variability is poorly understood. To address this deficiency, recent measurements investigated spatial variability in the upper few decimeters of sediment near the water-seabed interface within a fine-grained sediment deposit on the New England shelf. At each measurement location, acoustic probes were inserted into the sediment to collect direct *in situ* measurements of sediment sound speed and attenuation at near-ambient conditions, after which cores were collected from the inter-probe propagation paths for *ex situ* analysis of sediment physical, biological, and acoustic properties. Relationships among sediment properties, such as bulk density, porosity, grain size distribution, organic matter composition, infaunal community composition, and acoustic measurements spanning several frequency decades (10–1000 kHz) will be explored in this paper. Frequency dependence of sediment acoustic properties will also be discussed in the context of sediment acoustics models for mud based on the viscous grain shearing and extended Biot theories. [Sponsored by ONR.]

4:15

2pUW8. A T-matrix approach to wave propagation and scattering in layered environments with rough interfaces. Anatoliy N. Ivakin (Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, aniv@uw.edu)

Wave propagation and interactions at the interfaces of multi-layered media can be described in terms of transition matrix coefficients, or T-matrixes, taken from solutions for a plane wave transformation, reflection, scattering, and transmission, at an interface between two homogeneous half-spaces. These solutions can be found separately for each interface of the layered system and each combination of adjacent media. Then scattering amplitudes or the T-matrix of the whole multi-layered system can be obtained using an iterative procedure that starts from a simple case of two half-spaces at the basement of the system. A quite similar iterative procedure is frequently used for calculating the reflection coefficient of compressional plane waves for a multi-layered fluid system with flat interfaces using the reflection coefficients of each interface. In this paper, we show that a similar, but a more general T-matrix approach, can be developed to include interface roughness, different types of media and waves, for instance fluid, elastic or poroelastic layers, compressional and shear waves (vertically and horizontally polarized). As an example, scattering from a rough elastic layer is considered. An explicit first-order expression for the scattering strength is obtained and it's applications to remote sensing of sea ice layer are discussed. [Work supported by ONR.]

4:30

2pUW9. Geoacoustic inversion using 3D modal rays on the New England Shelf Break. Brendan J. DeCourcy (Woods Hole Oceanographic Inst., 86 Water St., Falmouth, MA 02543, bdecourcy@whoi.edu) and Julien Bonnel (Woods Hole Oceanographic Inst., Woods Hole, MA)

During the Seabed Characterization Experiment in 2021 (SBCEX21), hydrophones deployed on the seabed of the New England Shelf Break recorded acoustic signals from SUS charge explosions. Acoustic data from one of these TOSSIT hydrophones displays evident modal structures which cannot be fully captured with 2D propagation models. In this presentation, 3D acoustic modal ray tracing is shown to provide a stronger match for recorded modal travel times, and is used to invert for geoacoustic parameters of the seabed. Performance of 2D and 3D modal propagation models are presented, and limitations of the methods are shown in the context of large parameter space inversion efforts. [This research is supported by The Office of Naval Research.]

4:45

2pUW10. The effect of microfabric heterogeneity on shear wave properties in fine-grained sediments. Gabriel R. Venegas (Cntr. for Acoust. Res. and Edu. and Dept. of Civil and Environ. Eng., Univ. of New Hampshire, 33 Academic Way, Rm. W137, Durham, NH 03824, g.venegas@unh.edu), Yu-Hsuan Chao (Dept. of Bioeng., Univ. of Pittsburgh, Pittsburgh, PA), Jane McCue (Dept. of Civil and Environ. Eng., Univ. of New Hampshire, Durham, NH), Kang Kim (Dept. of Bioeng. and Dept. of Med., Univ. of Pittsburgh, Pittsburgh, PA), and John M. Cormack (Dept. of Medicine, Univ. of Pittsburgh, Pittsburgh, PA)

The water-sediment interface is highly susceptible to physical, biological, and chemical processes, which can create significant levels of heterogeneity in sediment biogeoacoustic properties at various spatial scales that affect transmission and scattering of elastic waves. Using a suite of microscopy, spectroscopy, and medical sonoelastography techniques, relationships between heterogeneity in the sediment microfabric and shear wave speed can be investigated at spatial scales ranging from centimeters to micrometers. Here, we study fine-grained sediments extracted from two unique intertidal mudflats in the Great Bay Estuary, New Hampshire, USA. Shear waves were excited via acoustic radiation force and external vibration in the 50–200 Hz frequency band, and particle velocity was sensed in the top 2 centimeters of sediment at approximately 0.5-micrometer spatial resolution using a research-grade medical imaging array. By directly imaging the propagating shear wave, spatial heterogeneity of shear wave speed was directly measured. Next, subsamples were extracted and prepared for microscopy, where volume fraction and orientation of organic, mineral, and aqueous components were studied at various spatial scales. A comparison of between shear wave speed and microfabric parameters will be presented. [Work sponsored by ONR.]

5:00

2pUW11. Seabed Analysis on the New England shelf break using ambient sound data and trans-dimensional geoacoustic inversion. Martin Siderius (Elec. and Comput. Eng., Portland State Univ., 1900 SW 4th Ave., Ste. 160, Portland, OR 97201, siderius@pdx.edu), Stan Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada), and Jorge Quijano (JASCO Appl. Sci., Victoria, BC, Canada)

The New England Shelf Break Acoustics (NESBA) Signals and Noise experiment was conducted in April-May 2021. Ambient sound data were collected over several days in the mud-patch area as well as in deeper water closer to the shelf break. A 16-hydrophone vertical array was used to measure the natural sound of breaking waves on the sea-surface. Using beamforming in the 500–700 Hz band, these data were used to obtain an estimate of the bottom reflection coefficient as well as the seabed layering. The reflection coefficient data were subsequently used with trans-dimensional inversion techniques to produce a geoacoustic model for the seabed (e.g., sound speed, density, and layering). Results show the ambient sound data can be used to produce well resolved geo-acoustic parameters, especially in the upper part of the seabed (e.g., <10 m). These results are compared between several locations on the New England Shelf Break area and are also compared with other published results using different estimation techniques. In addition, some of the issues related to the impacts of data errors and preferred measurement geometries will also be presented. [Work supported by the Office of Naval Research.]

Session 3aAAa

Architectural Acoustics: Memoriam for Jerry Christoff

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

Invited Papers

8:00

3aAAa1. Celebrating the life and achievements of Jerry Christoff. Samantha Rawlings (Veneklasen Assoc., 1711 16th St., Santa Monica, CA 90404, srawlings@veneklasen.com), John Lo Verde, and Wayland Dong (Paul S. Veneklasen Res. Foundation, Santa Monica, California, CA)

Jerry spent 61 years of his life in the service of acoustics. He was one of the first people hired by one of the first established acoustical firms in the United States, moving from staff engineer to firm Principal. Jerry started with designing and developing microphones early in his career, followed by his work in auditorium acoustics, remnants of which we still see in today's auditorium design practices. Later in his career, Jerry worked on improving the measurement methods for exterior façade noise reduction evaluation, and ultimately on HVAC noise study and improvements. Jerry has done it all and left the acoustical community better than he found it. Come join us in celebrating his life and his achievements during this special session.

8:20

3aAAa2. Jerry Christoff: A career driven by curiosity. George C. Kourtis (Threshold Acoust., 141 W Jackson Blvd., Ste. 2080, Chicago, IL 60604, gckourtis@gmail.com)

This session aims to present accounts and examples of the curiosity that drove Jerry Christoff in his work as a consultant and mentor as seen through the eyes of George Kourtis. George joined the staff of Veneklasen Associates in 2013 and directly assisted Jerry as a consultant from approximately 2015–2017 before Jerry's retirement. George subsequently moved on from VA in 2018 and is now a Senior Consultant at Threshold Acoustics. A throughline George observed both before joining VA and while working directly with Jerry was the almost childlike fascination and curiosity Jerry expressed for all areas of acoustics, even as an 82-year-old consultant with over 60 years of experience. In the spirit of Paul Veneklasen's research-based approach and legacy, Jerry's lines of inquiry and investigations on everything from concert hall design to plumbing noise inspired others, including George, to, in good faith, think critically, ask questions, and perform their own research, with the goal of always providing the best possible acoustical outcomes.

8:40

3aAAa3. Revisiting the Seattle Opera House. Jennifer Levins (Acentech, 33 Moulton St., Cambridge, MA 02138, jlevins@acentech.com)

Jerry Christoff played an integral part of the Seattle Opera House design in the early 1960s. An overview of the acoustic design elements for this space was published in 1964 in a paper co-authored with Paul Veneklasen. Many of the concepts presented in this paper, like reverberance, clarity, and envelopment, still influence acoustical design today. We will highlight the key points from this paper and share anecdotes from Jerry's career.

9:00

3aAAa4. Jerry Christoff's contribution to the development of performing arts acoustical design practices. John Lo Verde (Paul S. Veneklasen Res. Foundation, 1711 Sixteenth St., Santa Monica, CA 90404, jloverde@veneklasen.com), Samantha Rawlings (Veneklasen Assoc., Santa Monica, CA), and Wayland Dong (Paul S. Veneklasen Res. Foundation, Santa Monica, California, CA)

In the 1960s, Jerry Christoff, alongside Paul S. Veneklasen, performed research and development into acoustics for performing arts spaces. Some of the practices developed by these researchers are precursors to modern-day design metrics, as they looked at lateral reflections and source strength, to name a few. Small-scale models of performance spaces were being constructed with an ultrasonic source to better understand and visualize how sound waves behave within a performance space. This presentation will be focused on Jerry's contribution to science with regard to the design of performing arts spaces.

9:20

3aAAa5. Advancement of the exterior façade measurement method by Jerry Christoff. Wayland Dong (Paul S. Veneklasen Res. Foundation, 1711 Sixteenth St., Santa Monica, CA 90404, wdong@veneklasen.com), Samantha Rawlings (Veneklasen Assoc., Santa Monica, CA), and John Lo Verde (Paul S. Veneklasen Res. Foundation, Santa Monica, CA)

Acoustical design of façade elements is a key element in architectural acoustics. The assessment is required for multi-family housing projects and environmental projects. In the 1980s, when labs around the US were measuring the transmission loss (TL) of exterior façade elements, it was assumed that measurement of TL with a diffuse sound field method is valid for TL of real facades where a diffuse sound field does not exist. Jerry evaluated how the TL of exterior façade elements change based on the incident angle. In this talk, we will shed some light on the history of the work done by Jerry in the advancement of the exterior façade assessment methods.

WEDNESDAY MORNING, 15 MAY 2024 ROOM 207, 8:00 A.M. TO 12:00 NOON

Session 3aAAb

Architectural Acoustics: Soundscape Simulation: Opportunities and Challenges I

Catherine Guastavino, Cochair
CIRMMT & School of Information Studies, McGill University, 3661 Peel, Montreal, H3A 1X1, Canada

Josep Llorca-Bofi, Cochair
Institute for Hearing Technology and Acoustics, RWTH Aachen University, Kopernikustrasse 5, Aachen 52074, Germany

Andre Fiebig, Cochair
Engineering Acoustics, TU Berlin, Einsteinufer 25, Berlin 10587, Germany

Chair's Introduction—8:00

Contributed Papers

8:05

3aAAb1. Simulation of acoustical parameters of churches in a virtual acoustics laboratory. Gianluca Grazioli (McGill Univ., 555 Sherbrooke St. W, Montreal, QC H3A 1E3, Canada, gianluca.grazioli@mail.mcgill.ca) and Andrea Gozzi (Université de Montreal, Montreal, QC, Canada)

Current ISO standards for acoustical assessment limit the use of traditional mono-dimensional microphones for measuring equipment. However, microphone arrays offer more accurate spatial information compared to traditional microphones. This presents an opportunity to enhance research on architectural acoustics and preserve the acoustics of cultural heritage more effectively. Furthermore, modern recording studios equipped with virtual acoustics systems allow for the integration of spatial room impulse responses. This enables real-time auralization in controlled environments and enhances the overall immersive audio experience for users and musicians. This paper analyzes the main acoustical parameters obtained from spatial impulse responses captured in churches using various ambisonic microphones and inserted into a controlled, interactive, and immersive virtual acoustics system. The captured spatial acoustic measurements are then reproduced and evaluated in a virtual acoustics laboratory to identify any discrepancies between real and virtual spaces.

8:20

3aAAb2. Sandiaoling eco-friendly tunnel soundscape and virtual reality analysis. Khaing Thinzar (Dept. of Architecture, National Taiwan Univ. of Sci. and Tech., 43 Keelung Rd. Sec. 4, Taipei 106, Taiwan, M11213801@mail.ntust.edu.tw), Phoa Angela Grace Wibowo, Ni Made Putri Indriyani, Nikita Grace Manullang, Roxana Ghadiri, Stijn Zeger van Brug, Juliana Manuela Muet, Tuan Sanh Diep, Clarissa Averina, Chau N. Truong, Gabriela Niederberge, Shiang-I Juan, and Lucky S. Tsaih (Dept. of Architecture, National Taiwan Univ. of Sci. and Tech., Taipei, Taiwan)

Soundscape of Sandiaoling Ecological Tunnel, a 1.852 km bike path tunnel, has been studied for the taxonomy of sound sources, equivalent sound pressure levels, and the perceived sonic quality. This paper focuses on examining simulated sound source propagations in the tunnel using both omni and directional sound sources with Odeon Room Acoustic software. Additionally, it compares the specific receivers' auralizations with *in situ* recorded sounds in a virtual reality model along with a questionnaire to evaluate the distinctions between the two. The preliminary results showed that the simulated sonic quality in tunnel is less reverberant than expected. Proximity of receivers to sound source, the boundary of air to source and receivers inside tunnel as well as the absorption, scattering and transparency coefficients of materials used in simulated model are adjusted. Future work will explore into audio post-production and visual effects.

8:35

3aAAb3. Ambisonic recordings of sound fields from commonly occupied built environments. Hezekiah George (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, 1110 S. 67th St., PKI 210-F, Omaha, NE 68182-0816, hgeorge@mail.bradley.edu) and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, Omaha, NE)

There is growing use of simulated soundscapes to study human perception, such as for testing hearing aids or cochlear implants. Ambisonics is one method to record and then simulate three-dimensional sound fields with more accurate directional information. In this project, our aim has been to expand the currently available database of ambisonic recordings of soundscapes in built environments. To this end, we have made recordings of sound fields in frequently occupied spaces in the built environment such as classrooms and food courts, using a commercially available ambisonic microphone capturing up to first-order spherical harmonics. While databases of such recordings are available online, many recorded samples are short in duration, typically less than 5 min. long. Furthermore, there are fewer samples taken from occupied spaces in the built environment which are of interest in architectural acoustics. We will review recordings that are already available online and describe our acquired recordings which sample additional spaces and are longer in length. We will share a link to where these can be accessed and downloaded. An ultimate goal is for researchers to utilize these recordings to generate more ecologically valid sound fields when testing human responses in more controllable, simulated test environments.

Invited Papers

8:50

3aAAb4. A virtual reality method for assessing human responses to soundscapes. Andre Fiebig (Eng. Acoust., TU Berlin, Berlin, Germany), Niklas Meier (Technische Universität Berlin, Berlin, Germany, niklas.meier@tu-berlin.de), and Christian Dreier (Inst. for Hearing Technol. and Acoust., RWTH Aachen Univ., Aachen, Germany)

Virtual realities and simulations are becoming increasingly popular due to constant technological advancement offering more and more realistic scenarios. However, cases of VR being used to assess soundscape-related aspects of perception still appear to be rare. Here, a method for conducting auditory perception experiments in a virtual urban environment is presented and discussed. Participants are exposed to virtual scenarios simulating an urban green area with nearby traffic noise sources, whose visualization draws on physically-based visual renderings. The acoustic scenario is auralized in real-time using geometrical acoustic approaches including simulated and recorded sounds. The virtual environment is presented via head-mounted display and headphones. In this setting, evaluation and behavioural experiments are conducted, with the environment's acoustic and visual characteristics being experimentally manipulable or controllable. The presented method allows for perception experiments with a high degree of ecological validity while maintaining a controlled laboratory setting. As a practical application, the method is used to determine the impact of road traffic configurations on the restoration quality in an urban green area. Proceeding from the applied method, general opportunities and limitations of experiments performed in virtual urban environments to study noise effects in urban soundscapes are discussed.

9:10

3aAAb5. Enhancement of virtual acoustics rendering using boundary mounted dipole loudspeakers. Richard King (Graduate Program in Sound Recording, McGill Univ., Music Res., Montreal, QC H3A 1E3, Canada, Richard.King@mcgill.ca), Wieslaw Woszczyk, and Michail Oikonomidis (Graduate Program in Sound Recording, McGill Univ., Montreal, QC, Canada)

The Immersive Media Lab at McGill University hosts a Virtual Acoustics Technology (VAT) system incorporating a suspended array of omnidirectional loudspeakers. Using a convolution reverb engine, acoustic simulations of real spaces can be realized via a catalog of Room Impulse Response measurements. Reflected sound in the room helps to disguise the location of the sound emitters rendering virtual acoustics. One limitation of the system, however, is the interference between the lab's natural acoustics and the virtual environment generated by the VAT system. The improvement under consideration is to enhance diffusion along the walls of the lab, in order to mask the acoustical characteristics related to the physical dimensions of the room. Dipole loudspeakers are installed on the room boundaries and used to scatter reflections and reverberation along the wall surfaces enlarging the effective radiation surface of the walls. The scattered energy may mask specular reflections and reduce localization of the virtual acoustic sources. Investigations compare the result of scattering reflected sound vertically as opposed to horizontally across the boundaries. Measurements illustrate the effect of the dipole loudspeaker system when used on its own as well as working in conjunction with the existing omnidirectional loudspeaker array.

Contributed Paper

9:30

3aAAb6. One size does not fit all: Three tools and approaches for soundscape simulations. Valérian Fraisse (Schulich School of Music, McGill Univ. / STMS Ircam-CNRS-SU / Ctr. for Interdisciplinary Res. in Music Media and Technol., Montréal, QC, Canada, valerian.fraisse@mail.mcgill.ca), Cynthia Tarlao, Richard Yanaky, and Catherine Guastavino (Information Studies, McGill Univ. / Ctr. for Interdisciplinary Res. in Music Media and Technol., Montreal, QC, Canada)

We present a reflection on three prototypes of real-time interactive soundscape simulators aimed at supporting participatory urban sound planning and interventions. These prototypes were developed as part of the Sounds in the City cross-sectoral partnership through an iterative process involving various stakeholders. Each prototype enables different ways to manipulate soundscapes through tailored interfaces and audio/visual outputs, targeting different types of users. The first version is audio-only and uses live music tools for Ambisonics spatialization, with limited environmental modeling for co-design exercises with urban and sound professionals. The second version builds on the idea of the first and adds acoustic modeling. It was used to assess the impact of sound installations in public spaces through research-creation involving sound artists and residents. The last version utilizes (desktop or head-mounted) virtual reality with binaural rendering, immersing the user in an audio-visual city to raise sound awareness and support urban soundscape design. We emphasize that there is no one-size-fits-all tool. Rather, we highlight how different tools are needed for different auralization tasks and target user groups. These tools are presented through examples of early-stage conceptualization, educational components, creative processes, laboratory-based soundscape assessments, and both individual and participatory design sessions.

9:45

3aAAb7. Urban background sound recordings for virtual acoustics under various weather conditions. Josep Llorca-Bofi (Inst. for Hearing Technol. and Acoust., RWTH Aachen Univ., Kopernikustrasse 5, Aachen 52074, Germany, josep.llorca@akustik.rwth-aachen.de), Jonas Heck, Christian Dreier, and Michael Vorlaender (Inst. for Hearing Technol. and Acoust., RWTH Aachen Univ., Aachen, Germany)

One of the major challenges in the auralization of complex urban simulations is the presence of ambience, for example background sounds. Omitting them can result in a sterile and implausible impression. In this work we present a dataset of spatially recorded ambient sounds with metadata on meteorological and acoustical parameters. The recordings are ready to be admixed with simulated soundscapes corresponding to various weather conditions. In particular, 28 soundscapes have been recorded at the IHTApark (green space next to the Institute for Hearing Technology and Acoustics) in Aachen (Germany) in winter and spring. They have been segmented into 30-s auralizable snippets. Moreover, the database contains a statistical and psychoacoustical evaluation of the recordings.

10:05–10:20 Break

10:20

3aAAb8. Validation of auralized impulse responses considering masking, loudness and background noise. Jonas Heck (Inst. for Hearing Technol. and Acoust., RWTH Aachen Univ., Kopernikusstrasse 5, Aachen 52074, Germany, jonas.heck@akustik.rwth-aachen.de), Josep Llorca-Bofí, Christian Dreier, and Michael Vorlaender (Inst. for Hearing Technol. and Acoust., RWTH Aachen Univ., Aachen, Germany)

The use of outdoor virtual scenarios promises to have a lot of potential to facilitate reproducible sensory evaluation experiments in laboratory. Auralizations allow for the integration of simulated or measured sound sources and transfer paths between the sources and receivers. Nonetheless, pure simulations can lack perfect plausibility. This contribution investigates the augmentation of auralized outdoor scenes based on simulated impulse responses (IRs) by ambient or background sounds. For this purpose, foreground events such as car pass-bys are created by simplified simulation of impulse responses. Due to their large number of events, however, ambient sounds are typically not simulated. Instead, spherical microphone array recordings can be used to capture the background sound. Using synthesized car sounds, we examine how much the augmentation by background sound improves the auditory plausibility of simulated impulse responses in comparison with the equivalent measured ones.

10:40

3aAAb9. Perceptual study on combined real-time traffic sound auralization and visualization. Christian Dreier (Inst. for Hearing Technol. and Acoust., RWTH Aachen Univ., Kopernikusstrasse 5, Aachen 52074, Germany, christian.dreier@akustik.rwth-aachen.de), Rouben Rehman (Inst. for Hearing Technol. and Acoust., RWTH Aachen Univ., Aachen, Germany), Niklas Meier, Andre Fiebig (Eng. Acoust., TU Berlin, Berlin, Germany), and Michael Vorlaender (Inst. for Hearing Technol. and Acoust., RWTH Aachen Univ., Aachen, Germany)

Auralization is a promising approach for future application in urban planning including noise aspects. From a technical point of view, auralization of the urban environment is challenging due to the high number of sound sources, time-variant and inhomogeneous atmospheric conditions and the large scale of scenarios. This work presents the modeling and computationally efficient implementation of traffic sound sources and its integration to a complex urban scenario with real-time auralization. Furthermore, it presents results from a perceptual study whose immersion is enhanced by presenting a coupled visualization of the scene to the participants by using a head-mounted display. Whereas the experiment on the one hand comprises virtual reality-related aspects considering the assessment of immersion and plausibility of the scenario, on the other hand it covers the assessment of individual noise events in the urban soundscape. The experiment focuses on acoustically prominent noise events, such as particularly loud and harsh sources, or resulting from specific driving behavior. The results are discussed in the foreground for further improvements for audio-visual simulations of the urban environment.

11:00

3aAAb10. Towards a better understanding of multimodal integration and sensorimotor adaptation to audiovisual environmental incongruence using Virtual Reality. Xinyi Zhang (Dept. of Elec. Eng., École de Technologie Supérieure, 1100 Notre-Dame St W, Montreal, QC H3C 1K3, Canada, xinyi.zhang.1@ens.etsmtl.ca), Arian Shamei (Dept. of Elec. Eng., École de Technologie Supérieure, Vancouver, BC, Canada), Florian Grond (Dept. of Design and Computation Arts, Concordia Univ., Montreal, QC, Canada), Ingrid Verduyckt (École d'orthophonie et d'audiologie, Université de Montréal, Montreal, QC, Canada), and Rachel Bouserhal (Dept. of Elec. Eng., École de technologie supérieure, Montréal, QC, Canada)

People use previous knowledge and *in situ* judgment to produce speech with a vocal effort appropriate to a given environment's acoustics. To test how people integrate auditory and visual cues in speech production, we employed a three-by-three cross-conditional audiovisual match-mismatch paradigm. Three visually distinct environments with three different room acoustics were selected: a gymnasium, a classroom, and a hemi-anechoic room. The visual environment was presented with a Virtual Reality (VR) headset and the auditory environment was a diffuse room impression, playing back the participants' speech through loudspeakers in the room with different reverberation times. Participants were prompted to speak in all nine combinations of the audiovisual conditions, with three being congruent and six incongruent. Linear mixed-effects regression modeling was used to evaluate the effect of the audiovisual manipulations and time course on mean intensity. Preliminary results indicate that participants initially spoke at a level that matched the visual expectation and then adapted to the audio condition; detailed analysis of the time course of adaptation is ongoing and will be presented.

This study furthers our understanding of multimodal integration and the sensorimotor adaptation of speech production, which finds applications in fields including communication in noise and VR soundscape design.

11:20

3aAAb11. Museum Acoustics: Classification and auralization of design approaches. Milena J. Bem (School of Architecture, Rensselaer Polytechnic Inst., 1605 Hutton St., Apt. 10, Troy, NY 12180, jonasm@rpi.edu), Mincong Huang (School of Architecture, Rensselaer Polytechnic Inst., Troy, NY), Vic Brooks, Samuel Chabot (Experimental Media and Performing Arts Ctr., Rensselaer Polytechnic Inst., Troy, NY), and Jonas Braasch (School of Architecture, Rensselaer Polytechnic Inst., Troy, NY)

Architecture defines various aspects of museum spaces. Despite often being considered secondary importance by some deciders and designers, the acoustic environment significantly shapes the interpretation of exhibitions and human-building interactions. Focusing on enhancing museum acoustics, this study builds upon previous research that developed a methodology for simulating and evaluating museum soundscapes through psychophysical experiments in immersive environments, showing that participants preferred congruent soundscapes over generic noise maskers [J. Acoust. Soc. Am. **154**, A257]. Expanding on these previous investigations, our current research involves cataloging and classifying design approaches currently used within museum contexts—architectural elements, acoustic characteristics, spatial distribution, exhibition strategies, multimodal approaches, etc. Subsequently, we present an auralization method for these museum spaces using the ray tracing method developed for collaborative virtual reality systems [J. Acoust. Soc. Am. **143**, 1824]. This method is used within the CRAIVE-Lab at Rensselaer Polytechnic Institute to facilitate tangible analysis, enriching our understanding of these solutions to support future design efforts by guiding architects and acousticians to optimize the museum experience. It offers significant insights into current practices, emerging trends, or notable innovations, thereby promoting more experiential design approaches to shaping the acoustic impacts of the museum environment.

11:40

3aAAb12. Reducing noise with augmented reality in indoor soundscapes. Nicolas Misdariis (Sound Percept. and Design group, Ircam STMS Lab, 1, Pl. Igor Stravinksy, Paris F-75004, France, nicolas.misdariis@ircam.fr), Mathieu Lagrange (LS2N Ecole Centrale Nantes, Nantes, France), and Romain Serizel (LORIA, Vandœuvre lès Nancy, France)

Noise pollution has a significant impact on quality of life. In indoor soundscapes like open offices, noise exposure creates stress that leads to reduced performance, provokes annoyance and changes in social behaviour. The ReNAR project aims at studying two augmented reality approaches, targeted towards additional sound sources which levels are below or equal to the noise sources ones The first approach tend to conceal the presence of unpleasant sources by adding some spectro-temporal cues which will seemingly convert it into a more pleasant one. Adversarial machine learning techniques will be considered to learn correspondences between noise and pleasing sounds and to train a deep audio synthesiser able to generate an effective concealing sound of moderate loudness. The second approach tend to tackle a common issue encountered in open offices, where the ability to concentrate on the task at hand is made harder when people are speaking nearby. We propose to reduce the intelligibility of nearby speech by the addition of sound sources whose spectro-temporal properties are specifically designed or synthesised with a generative model to conceal important aspects of the nearby speech. The formal position, general frame and expected outcomes of the project will be developed and discussed.

Session 3aAAc

Architectural Acoustics: AIA CEU Presenters Course Training Session

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

Kenneth W. Good, Cochair
Armstrong World Industries Inc., 2500 Columbia Avenue, Lancaster, PA 17601

Bennett M. Brooks, Cochair
Brooks Acoustics Corporation, 49 N. Federal Highway, #121, Pompano Beach, FL 33062

Chair's Introduction—9:55

Invited Papers

10:00

3aAAc1. Architectural acoustics short course presentation material. K. Anthony Hoover (McKay Conant Hoover, 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362, thoover@mchinc.com) and Kenneth W. Good (Armstrong World Industries Inc., Lancaster, PA)

The Technical Committee on Architectural Acoustics (TCAA) is a Registered Provider in the American Institute of Architects (AIA) Continuing Education System (CES). The TCAA has developed a standardized introductory short course for architects called "Architectural Acoustics," An architect can earn one continuing education unit (CEU) by attending this short course, if presented by a qualified member of TCAA. The course covers topics in sound isolation, mechanical system noise control, finish treatments, and implementation of quality acoustical spaces. This paper will cover the course material in order to prepare and qualify potential presenters. In order to qualify as an authorized presenter for this AIA/CES short course, attendance at this special session and membership in TCAA are required.

11:00

3aAAc2. Architectural acoustics continuing education course—Presenter registration and reporting. Kenneth W. Good (Armstrong World Industries Inc., 2500 Columbia Ave., Lancaster, PA 17601, kwgoodjr@armstrong.com) and Bennett M. Brooks (Brooks Acoust. Corp., Pompano Beach, FL)

The Technical Committee on Architectural Acoustics (TCAA) is a Registered Provider in the American Institute of Architects (AIA) Continuing Education System (CES). The TCAA has developed a standardized introductory short course for architects called "Architectural Acoustics," for which attendees can earn one continuing education unit (CEU). This paper will cover the administrative requirements of the ASA/CES, to prepare potential presenters. These requirements include the proper handling of paperwork so that AIA members may receive credit for the course. The manner in which the course is given is also dictated by AIA requirements. TCAA membership and attendance at this workshop are required to qualify as an authorized presenter for this AIA/CES short course. Of course, anyone is free to register with the AIA to provide their own CEU program. However, the advantages of participating in this program are that the TCAA short course is already prepared, it is pre-approved by the AIA, and the registration fees are paid by the Acoustical Society of America.

Session 3aAB

Animal Bioacoustics: Acoustic Ecology, Biological Soundscapes, and Animal Vocal Communication and Physiology I

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

Contributed Papers

8:00

3aAB1. Harmonizing nature's symphony: Computational filtering unveiling the secrets of animal acoustics. Lisa M. DiSalvo (Comput. Sci., Columbia Univ., 1612 Mount Laurel Rd., Temple, PA 19560, lisdis19@gmail.com) and Laura Kloepper (Dept. of Biological Sci., Univ. of New Hampshire, Durham, NH)

By harmonizing and unveiling the symphony of nature, we seek to reveal the mysteries embedded in the soundscape of various species with acoustic filters. Utilizing computational filtering, we can draw specific conclusions about the evolution of animal populations and understand the significant role the environment plays in interacting with animal populations. By employing these computational acoustic filtering techniques, we can analyze and decode intricate patterns, frequencies, and nuances embedded in animal vocalizations. This interdisciplinary approach not only enhances our understanding of animal communication systems but also sheds light on the ecological dynamics at play both within animal vocalization and environmental vocalization. These conclusions contribute to the broader field of bioacoustics, providing insights that may have implications for wildlife conservation, behavioral ecology, and the delicate balance of ecosystems. Beyond advancing bioacoustics, our research highlights the educational impact of teaching computational acoustic filtering in Python and R. This research seeks to emphasize the importance of equipping young ecological researchers with these skills to foster a holistic understanding of ecological dynamics. The ultimate goal of our educational research pursuit is to prepare the next generation of researchers to contribute meaningfully to wildlife conservation, and ecology as a whole with interdisciplinarity in mind.

8:15

3aAB2. A PAMGuard catalog for comparing detection and classification performance of two mode decomposition algorithms. Kerri D. Seger (Appl. Ocean Sci., 2127 1/2 Stewart St., Santa Monica, CA 90404, kerri.seger.d@gmail.com), Sujay Balebail (Dept. of Biology, Univ. of Washington, Seattle, WA), Imogen Lucciano (Oregon State Univ., Ithaca, NY), Yue Liang (Dept. of Elec. and Comput. Eng., Univ. of New Hampshire, Durham, NH), Mahdi Al-Badrawi (Elec. and Comput. Eng., Univ. of Maine, Orono, ME), and Nicholas J. Kirsch (Dept. of Elec. and Comput. Eng., Univ. of New Hampshire, Durham, NH)

The ONR project "Application of an Empirical Mode Decomposition (EMD) detection and classification process to environments for Naval monitoring and detection" created empirical and variational mode decomposition analysis workflows to detect and cluster diverse underwater signals across four distinct datasets. Twenty-five call types from the Bering and Chukchi Seas, Aeon's North Atlantic sites, CTBTO hydroacoustic stations, and the Chambal River were detected and clustered with varying levels of success. The final precision, recall, and accuracy results of these call types will be presented. Evaluating the performance of our developed detection and classification algorithms involved a comparison with the widely used open-source software, PAMGuard. However, obtaining information about the parameters used to construct detectors for different call types in PAMGuard proved challenging due to limited literature availability. To the best of our knowledge, some call types had never been processed in PAMGuard, and for those that had, not all parameters required for adapting the detectors to our datasets were consistently published. To facilitate the use of our PAMGuard detectors by other researchers for their own datasets, we will present a user-friendly guide that will outline the features that yielded optimal PAMGuard performance for each call type.

8:30

3aAB3. Spatiotemporal patterns of fish chorusing in California National Marine Sanctuaries. Ella B. Kim (Scripps Inst. of Oceanogr., Univ. of California San Diego, 8635 Kennel Way, Ritter Hall, La Jolla, CA 92037, ebkim@ucsd.edu), Annebelle Kok (Scripps Inst. of Oceanogr., Univ. of California San Diego, Groningen, Netherlands), Emily Donahue (California State Univ. Monterey Bay, Monterey Bay, CA), Emma Beretta (California Polytechnic State Univ., San Luis Obispo, San Luis Obispo, CA), Leila Hatch (Office of National Marine Sanctuaries, Scituate, MA), John E. Joseph, Tetyana Margolina (Oceanogr., Naval Postgrad. School, Monterey, CA), Lindsey Peavey Reeves (National Marine Sanctuary Foundation, Silver Spring, MD), and Simone Baumann-Pickering (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA)

In aggregations, some male fish will sing together in "chorus" for hours, to attract female mates. Through analyzing passive acoustic data, one can determine which, when, and where fish are chorusing, to identify breeding grounds, species distributions, and mating seasons. National marine sanctuaries aim to protect marine ecosystems, particularly breeding and feeding grounds for at risk species. Through the Sanctuary Soundscape Monitoring project, a cumulative 17.9 years of passive acoustic data were manually analyzed for fish chorusing in Monterey Bay, Channel Islands, and proposed Chumash Heritage National Marine Sanctuaries. We aimed to determine which fish were chorusing within the sanctuaries, where and when they occurred, and if fish were engaging in acoustic niche partitioning (minimizing overlap in time/frequency to reduce competition). We found five unique fish choruses including plainfin midshipman, bocaccio rockfish, white seabass, and two unidentified fish choruses, which showed distinct diel, seasonal, and spatial preferences. Fish were found to partition acoustic space in frequency but not in time. Understanding fish reproductive habitat and behavior within sanctuaries is crucial for safeguarding threatened fish species. Moreover, studying fish chorusing is a powerful tool to noninvasively monitor fish to understand their reproductive habits, with goals of better protecting them.

8:45

3aAB4. Particle motion polarization of offshore fish vocalizations, boat noise, and ambient soundscapes. Ian T. Jones (Ctr. for Acoust. Res. and Education, Univ. of New Hampshire, Durham, University of New Hampshire, 8 College Rd., Morse Hall, Durham, NH 03824, ian.t.jones@unh.edu) and Julien Bonnel (Woods Hole Oceanographic Inst., Woods Hole, MA)

Acoustic particle motion is an important acoustic cue for fish and aquatic invertebrate hearing. When reported for bioacoustics applications, it is often described as a magnitude only. However, particle motion is a vector quantity with polarization information (including directionality) relevant to detection and localization of sound cues by fishes and invertebrates. This study applied established metrics of particle motion polarization including ellipse amplitude, angle, orientation, and degree of polarization, to compare sounds of fish vocalizations with that of boat and ambient sounds at a 900 m-deep site (in the Atlantic Deepwater Ecosystem Observatory Network) offshore of Florida. Polarization metrics were computed for bivariate signals in the plane formed by the horizontal source-receiver and vertical axes. Boat sound polarization was quantified at multiple distances from the closest point of approach. Fish and boat sounds had narrower particle motion ellipses and larger amplitudes in the horizontal compared to ambient sounds that were more circular with larger relative vertical amplitudes. Fish and boat sounds were often strongly polarized, which may allow spatial release from masking given sufficient angular separation of sounds. This analysis framework has promising applications for monitoring *in situ* directionality of fish and other animal vocalizations and for modelling directional masking.

9:00

3aAB5. Effects from particle motion and substrate-borne vibration on fishes and invertebrates: Recommendations on research questions and methodologies. Shane Guan (Div. of Environ. Sci., Bureau of Ocean Energy Management, 45600 Woodland Rd., Ste. #455-C33, Sterling, VA 20166, guan@cua.edu) and Arthur N. Popper (Univ. of Maryland, College Park, MD)

There have been substantially increased interest in effects of particle motion and substrate-borne vibration on fishes and aquatic invertebrates over the past decade. This is because these signals, unlike sound pressure, are the basis of hearing in most fishes and in all aquatic invertebrates. However, studies to address these issues continue to face challenges due to the diversity of species, the broad range of hearing mechanism, and the complexity of the physical acoustic environment in conducting studies. Additionally, we do not yet have a good understanding of, and mechanisms for, source generation, calibration, and measurements to be able to examine effects properly. Related to this is the critical issue is how research questions are applicable to regulatory agencies. In October 2023, the US Bureau of Ocean Energy Management held a workshop to address different research methodologies and their pros and cons as the basis for study of the behavioral and physiological responses to particle motion and substrate-borne vibration. Subsequently, a review was conducted linking research priorities previously identified concerning anthropogenic noise effects on animals with appropriate research methodologies. The review also provides recommendations on experimental settings that are most appropriate to address specific research questions for regulatory concerns.

9:15

3aAB6. Using acoustic energy of vocalizations to monitor population size and phenology of anurans. Callyan L. Lacio (Dept. of Biological Sci., Univ. of New Hampshire, 105 Main St., Durham, NH 03824, callyan.lacio@unh.edu), David S. Steinberg (Dept. of Biological Sci., Univ. of New Hampshire, Durham, NH), Andrea M. Simmons (Cognit., Linguist., & Psychol. Sci., Brown Univ., Providence, RI), and Laura Kloepper (Dept. of Biological Sci., Univ. of New Hampshire, Durham, NH)

As a species that lives at the land/water interface, the American bullfrog (*Rana catesbeianus*) serve as a bioindicator in many habitats, yet also invasive in many locations. Due to challenges with traditional monitoring approaches, there is a lack of fine-scale population and phenological data for bullfrogs. Passive acoustic monitoring (PAM) can provide a low-cost alternative with high-resolution data for monitoring vocal animals. Sexually mature male bullfrogs attract mates by calling from exclusive territories. These vocalizations can be used to explore bullfrog behavior, population size, and phenology. We describe the analysis framework and initial results from an project monitoring the vocal behavior of frogs in 25 ponds in southeastern New Hampshire during the reproductive season using acoustic arrays. By using an acoustic energy index (RMS amplitude), we can estimate numbers of frogs in ponds, determine timing of reproduction, and even document anthropogenic disturbance. Our results can lead to future uses of PAM to monitor population size and phenology and develop reliable long-term management and conservation strategies.

9:30–9:45 Break

9:45

3aAB7. Using bioacoustics to determine bird community patterns in a post-industrial city. Christopher Dennison (Dept. of Biology, Carleton Univ., 1125 Colonel By Dr., Ottawa, ON K1S5B6, Canada, christopherontheriver@gmail.com), Rachel T. Buxton (Inst. of Environ. and Interdisciplinary Sci., Carleton Univ., Ottawa, ON, Canada), Katherine Brown, and Amber L. Pearson (Michigan State Univ., Flint, MI)

Urban landscapes experiencing population loss often maintain high quantities of vacant land which cause social stress but also create opportunities for conservation of wildlife, including birds. Understanding how features of the urban environment affect bird communities is needed to support planning and policy that creates more effective biodiversity outcomes. Using acoustic recorders, we explored the factors that affect bird communities in Detroit, MI, based on features in surrounding neighborhoods. We compared Shannon diversity, richness, and acoustic detection of birds at 110 recording sites from 2021 to 2023. We used a generalized linear model approach to determine the moderating effect of variables including Normalized Difference Vegetation Index (NDVI) and density of vacant lots around recording sites on bird community space use. We found increased bird diversity at recording sites surrounded by higher densities of vacant land and evidence that some habitat specialists use these areas more than others. Our results indicate habitat preference for areas with more vacant lots, and general preferential habitat selection for certain features of the urban environment. Understanding how urban bird communities use space in a post-industrial, urban landscape will help inform more effective nature-based solutions and urban plans that balance conservation, health, and social justice goals.

10:00

3aAB8. Using soundscape to monitor population size, demographics, and antipredator behavior in a dense aggregation of colonial seabirds. Valerie M. Eddington (Dept. of Biological Sci., Univ. of New Hampshire, 6 Stonecroft, Apt. 6, Portsmouth, NH 03801, valerie.eddington@unh.edu), Joseph Brosseau (Dept. of Biological Sci., Univ. of New Hampshire, Durham, NH), Liz Craig (Shoals Marine Lab., Univ. of New Hampshire, Durham, NH), Easton White, and Laura Kloepper (Dept. of Biological Sci., Univ. of New Hampshire, Durham, NH)

Migratory seabirds are vulnerable to decline due to climate change and anthropogenic disturbances. Common terns (*Sterna hirundo*) are highly vocal colonial seabirds that serve as bioindicators of their foraging grounds throughout their migratory range. Historically, monitoring colonial seabirds is invasive and time-consuming, and traditional acoustic approaches are complicated by high amounts of call overlap. Monitoring the behavioral ramifications of disturbance, as well as overall colony size and health, is crucial to implementing effective management decisions. However, methods are needed to do so efficiently and with minimal disturbance. In this study, we demonstrate that population size, demographics, and behavior can be assessed acoustically through changes in acoustic energy across varying temporal scales. To do this, we compared acoustic energy to in-person observations of nest density, chick-hatching, and investigator disturbance. We found that trends in acoustic energy align with observations of nest density, and the distribution of acoustic energy across frequency bands is indicative of colony demographics. Furthermore, we found a significant relationship between acoustic energy and investigator disturbance within 20 meters of an acoustic recorder. Overall, our findings suggest that colony-wide trends in population size, demographics, and behavior can be monitored via acoustic energy without the time-consuming analysis of individual calls.

10:15

3aAB9. Characteristics of courtship calls that could provide clues to physiological state or genetics of the emperor penguin, *Aptenodytes forsteri:* A case study of analysis using the Teager-Kaiser energy operator. Kody C. Seger (Marine Technol. Society, Columbus, OH), Kerri D. Seger (Appl. Ocean Sci., 2127 1/2 Stewart St., Santa Monica, CA 90404, kerri.seger.d@gmail.com), Justin Brackett (SeaWorld San Diego, San Diego, CA), and Ann E. Bowles (Hubbs SeaWorld Res. Inst., San Diego, CA)

Vocal behavior can be an indicator of physiological state or genetic make-up, but has not been developed as a diagnostic tool in seabirds. Aptenodytes penguins lack external sexual dimorphism, but the sexes have dimorphic courtship calls. We present a case study in which unique call structure of an emperor penguin (*Aptenodytes forsteri*) was associated with a same-sex bonded pair. Typical males produce lower frequency calls with proportionally more long bursts, while typical females produce slightly higher frequency calls with proportionally more short bursts. We recorded two male emperor penguins (E-79 and E-81) at SeaWorld San Diego that behaved as a bonded pair. E-79 produced a call that was qualitatively different from the male type to human listeners. Using the Teager-Kaiser Energy Operator to visualize bursts, we calculated and compared burst rates of E-79 and E-81 to other emperor penguins in the SeaWorld colony and a colony at Cape Crozier, Antarctica. Analysis showed E-79's calls had a unique burst rate structure that was intermediate between typical male and female patterns. Our results suggest that while emperor courtship calls are usually strongly sexually dimorphic, it is not always the case. The exceptions could provide interesting insights into call development, physiology, and/or genetics.

10:30

3aAB10. Acoustic ecology of Adélie penguins in the West Antarctic Peninsula. Danielle T. Fradet (Dept. of Biological Sci., Univ. of New Hampshire, Spaulding Hall, Durham, NH 03823, dtf1008@usnh.edu), Megan A. Cimino (Inst. of Marine Sci., Univ. of California Santa Cruz, Santa Cruz, CA), Easton White, and Laura Kloepper (Dept. of Biological Sci., Univ. of New Hampshire, Durham, NH)

Adélie penguins (*Pygoscelis adeliae*) are bioindicators for the rapidly changing Antarctic environment, making understanding their population dynamics and behavior of high research priority. However, collecting detailed population data throughout the breeding season on many colonies is difficult due to Antarctica's harsh conditions and remote location. The colonial breeding ecology of Adélie penguins has led to the evolution of a highly vocal species with individualized calls, making them well-suited for passive acoustic monitoring (PAM) with autonomous recording. PAM units can potentially provide an easily deployable and scalable way to collect fine-scale data on population estimates and breeding phenology. Here I present a framework for using acoustic indices to monitor phenology of dense penguin colonies even under high wind conditions. I evaluate the relationship between acoustic indices such as RMS amplitude and penguin colony size between distinct breeding stages (incubation, guard, crèche, and fledge) on Torgersen and Humble Islands in the West Antarctic Peninsula with an automated pipeline implemented in R. Using PAM to interpret penguin vocalizations for population size and breeding phenology estimates could lead to the development of a real-time remote monitoring system over a large spatial footprint, revealing Adélie penguin responses to climate change.

10:45

3aAB11. Ontogeny of vocalizations in Adélie penguin (*Pygoscelis adeliae*) chicks from West Antarctica. Michele L. Adams (Dept. of Biological Sci., Univ. of New Hampshire, 83 Main St., 10953 Granite Square Station, Durham, NH 03824, Michele.Adams@unh.edu), Danielle T. Fradet (Dept. of Biological Sci., Univ. of New Hampshire, Durham, NH), Megan A. Cimino (Inst. of Marine Sci., Univ. of California, Santa Cruz, CA), Easton White, and Laura Kloepper (Dept. of Biological Sci., Univ. of New Hampshire, Durham, NH)

Acoustic indices are an efficient method for monitoring dense aggregations of vocal animals but require understanding the acoustic ecology of the species under examination. The present understanding of avian behavior and vocal development is primarily derived from the research of songbirds (Passeriformes). However, given that behavior and environment can differ greatly among bird orders, passerine birdsong may be insufficient to define the vocal ontogeny of non-passerine birds. Like many colonial nesting seabirds, the Adélie penguin (*Pygoscelis adeliae*) is adapted to loud and congested environments with limited cues to identify kinship within aggregations of conspecifics. In addition to physical or geographical cues to identify offspring, adult *P. adeliae* rely on vocal modulation. Numerous studies have been conducted on mutual vocal modulations in mature *P. adeliae*, but limited research has explored the vocal repertoire of the chicks and how their vocalizations evolve over time. Using the deep learning-based system, DeepSqueak, this study characterized the vocal ontogeny of *P. adeliae* chicks in the West Antarctic Peninsula to aid in autonomously tracking their age. Understanding the phenological communication patterns of vocal-dependent seabirds can help measure the impact of climate change on this indicator species through non-invasive methods.

Session 3aAOa

Acoustical Oceanography: Topics in Acoustical Oceanography II

Ernst Uzhansky, Cochair
Marine Geosciences, University of Haifa, Izhaq Greenboim St., apt. 12, (Korean Family), Haifa 3498793, Israel

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

Contributed Papers

7:55

3aAOa1. A comparison of tidal components across multiple sites using ultra-low frequency ocean acoustic measurements. Anthony Eller (Appl. Ocean Sci., Springfield, VA) and Kevin D. Heaney (Appl. Ocean Sci., 11006 Clara Barton Dr., Fairfax Station, VA 22039, oceansound04@yahoo.com)

Recent long-term analysis of the hydrophone measurements taken in the deep sea reveals indications of multiple tidal components. Tidal harmonics corresponding to solar and lunar diurnal and semi-diurnal tide components were determined by analyzing the Cepstrum (variance of a spectral power) of the year or more long time series (0.5 Hz) of mid-water column measurements taken by the United Nations Comprehensive Test Ban Treaty Organization (CTBTO) in the Pacific, Atlantic and Indian Ocean. Postulations for the origin of this signal include rolling rocks, ocean turbulence, a change in the surface height, currents, mechanical/electrical (non-acoustic) noise and wind/wave induced noise propagating in the ocean. In this paper, this work is extended to more sites and encompasses a larger portion of the ultra-low frequency range (i < 5 Hz). it is found that different sites have quite different tidal components (as expected) but also have different frequency components (some with 0.5–1 Hz and others with 3–5 Hz). This opens the possibility of acoustic soundscape differences between sites due to regional source generation and local acoustic propagation.

8:10

3aAOa2. Marine soundscape monitoring from underwater autonomous vehicles—Passive acoustic monitoring gliders. Pierre Cauchy (Inst. des Sci. de la mer (ISMER), Université du Québec à Rimouski (UQAR), 310 allée des Ursulines, Rimouski, QC G5L 3A1, Canada, pierre_cauchy@uqar.ca)

Ocean gliders are buoyancy-driven autonomous underwater platforms, able to collect oceanographic measurements along vertical profiles during multi-months missions, covering thousands of kilometers. They glide quietly through the water column without propulsion noise and are therefore extremely suitable for Passive Acoustic Monitoring (PAM) of the marine environment. From PAM glider data in the Mediterranean Sea and the Southern Ocean, we illustrate the current and potential uses of PAM gliders for the study of physical oceanography, biology, ecology and for regulatory purposes. We evaluate limiting factors for PAM glider survey, such as platform-generated and flow noise, instrument size and power constraints, profiling ability and movement of the platform. We provide recommendations and good practices for typical PAM glider surveys and present future developments identified by the PAM glider community to further develop the readiness level and societal impact of PAM glider observation: (1) Calibration of the PAM glider to collect absolute sound levels; (2) adapted sampling methods and statistical analysis techniques to perform population density estimation; and (3) Integration of PAM glider observation to existing monitoring programs.

8:25

3aAOa3. Cooperative-hydrophone-based underwater soundscape monitoring in a bustling harbor enhanced by coral ecosystem. Ben Liu (Inst. of Deep-Sea Sci. and Eng., Chinese Acad. of Sci., Hainan, Sanya 572000, China, liub@idsse.ac.cn) and Wen Xu (Zhejiang Univ., Sanya, China)

Anthropogenic underwater noise is an emerging pollutant, urging a focus on studying the spatial and temporal dynamics of underwater soundscapes for effective environmental protection. This study centers on a bustling harbor enriched by coral ecosystems, employing 5–8 spatially distributed bottom-mounted passive acoustic monitoring (PAM) nodes strategically placed at locations of interest with diverse natural conditions and ship traffic intensities. Focused on the intricate interplay between anthropogenic and natural contributors, including shipping traffic and coral ecosystems, one-third octave band sound pressure levels SPLs were calculated and analyzed at different frequencies, revealing significant variations in levels across different times and locations. Diurnal variations in ambient noise levels highlight the complex relationship between human activities and marine ecosystems. Higher and more variable SPLs, primarily related to shipping traffic, were recorded during the daytime, while lower SPLs, accompanied by increased fish sounds, were observed at night. The collaborative deployment of multiple hydrophones enables an estimation of noise source direction, and multi-hydrophone based principal component analysis (PCA) for dimensionality reduction and subsequent clustering methods were used for noise source classification. This study significantly contributes to the field of underwater noise monitoring, offering a holistic perspective on busy harbor environments.

8:40

3aAOa4. Quantitative comparison of ocean soundscape model and measurements in the Atlantic outer continental shelf. Kevin D. Heaney (Appl. Ocean Sci., 11006 Clara Barton Dr., Fairfax Station, VA 22039, oceansound04@yahoo.com)

For 4 years, the Atlantic Deep Water Ecosystem Observing Network (ADEON), measured the acoustic soundscape using 7 bottom mounted landers off the coast of the South East United States (from Florida to Virginia). A shipping and noise model was built to generate regional sound maps, as well as to perform site-specific modeling for frequencies ranging from 20 Hz to 3150 Hz. The regional sound maps reveal that water depth is the key driver to local sound levels followed by the local geo-acoustic properties. It is predicted that the sound can be as much as 10 dB higher on the seafloor than at depths of 10 m. The site specific (where landers were deployed) model was run with a wide range of sediment uncertainty in the model, given our lack of knowledge of the sediment structure in water depths of 200 to 800 m deep, far from shore. A simple qualitative comparison of the soundscape measurements with the model provides immediate insight into the type of sediment at each lander position. Using this estimated sediment type, quantitative comparisons between the model and the data are made.

8:55–9:15 Break

9:15

3aAOa5. Comparing coastal marine habitats by combining passive acoustics with metagenomics. 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)

Past studies have combined passive acoustic monitoring with environmental genetic methods to detect target organisms, however, no studies to date have employed metagenomics concurrent with passive acoustic monitoring of soundscapes for comparison of marine habitats. The present study used both approaches simultaneously for holistic observation of marine habitats to reveal information beyond using either technique independently. Water samples for metabarcoding (four primer sets) were collected during periods of passive acoustic monitoring from three different marine habitats, each at four different geographic locations along the New Hampshire/Maine coastlines. Multivariate analyses compared discrimination among habitat types and geographic locations by analyzing acoustic metrics generated using the Soundscape Code and metagenomic taxonomic assignments. Passive acoustic monitoring provided insight into environmental features that were unobservable with metagenomics, especially anthropogenic activity and geophysical processes, whereas metagenomics provided a more complete picture of the biological composition of habitats through detection of organisms that were not actively producing sound. This enables simultaneous evaluation of biological and functional connectivity of marine habitats by detecting what organisms are present and their contributions to the soundscape. In future, genetic and acoustic indicators will be used for prediction of substrate characteristics and sound sources to model acoustic propagation environments.

9:30

3aAOa6. Doppler sonar records of fish movement through a strong tidal stream: 3-months of observations from Grand Passage Nova Scotia. Len Zedel (Phys. and Physical Oceanogr., Memorial Univ. of NF, 287 Prince Phillip Dr., St. John's, NF A1B 3X7, Canada, zedel@mun.ca), Alex E. Hay (Oceaography, Dalhousie Univ., Halifax, NS, Canada), and Shane Anderson (Phys. and Physical Oceanogr., Memorial Univ. of NF, St. John's, NF, Canada)

Coastal passages with strong tidal streams present potential for renewable energy generation with in-stream hydrokinetic turbines. However, there are concerns that these systems will lead to environmental impacts when marine life interact with moving turbine blades. We explore measurements of fish movement using Doppler current profilers as a way of quantifying the frequency of such interactions. Fish are detected in Doppler sonar data using calibrated backscatter levels in each of the four acoustic beams; by reprocessing un-averaged data it is possible to extract both fish and water velocities independently. We analyzed three months of Doppler sonar data collected in Grand Passage Nova Scotia from September until December 2014. The data show fish detections in large numbers for only a few days on three occasions at the deployment site. Most of the observations show fish moving at the same speed as the water but there are times when there is significant difference in fish and water speeds.

9:45

3aAOa7. Humpback whale song vocalization behavior and temporo-spatial distributions in the Norwegian and Barents Sea observed with a coherent hydrophone array. Saunak Samantray (Elec. and Comput. Eng., Northeastern Univ., 360 Huntington Ave., Boston, MA 02115, samantray.s@northeastern.edu), Arpita Ghosh, Sai Geetha Seri (Elec. and Comput. Eng., Northeastern Univ., Boston, MA), Hamed Mohebbi-Kalkhoran (Mech. Eng., Massachusetts Inst. of Technol., Cambridge, MA), Olav R. Godoe (Inst. of Marine Res. Norway, Bergen, Norway), Nicholas C. Makris (Mech. Eng., Massachusetts Inst. of Technol., Cambridge, MA), Heidi Ahonen (Norwegian Polar Inst., Tromsø, Norway), and Purnima R. Makris (Elec. and Comput. Eng., Northeastern Univ., Boston, MA)

The vocalization behavior of humpback whales in the Norwegian and Barents Seas is examined based on recordings of a large-aperture, densely-populated coherent hydrophone array system. The passive ocean acoustic waveguide remote sensing (POAWRS) technique is employed to provide detection, bearing-time estimation, time-frequency characterization and classification of the humpback whale vocalizations. The song vocalizations, composed of highly structured and repeatable set of phrases, were detected throughout the diel cycle between February 18 to March 8, 2014. The beamformed spectrograms of the detected humpback vocalizations are classified as song sequences based on inter-pulse intervals and time-frequency characteristics, verified by visual inspection. The song structure is compared for humpback whale vocalizations recorded at three distinct regions off the Norwegian coast, Alesund, Lofoten and Northern Finmark. Multiple bearing-time trajectories for humpback songs were simultaneously observed indicating multiple singers present at each measurement site. Humpback whale received call rates and temporo-spatial distributions are compared across the three measurement sites. Geographic mapping of humpback whale calls from their bearing-time trajectories is accomplished via the moving array triangulation technique.

10:00

3aAOa8. Active learning algorithms for autonomous platforms to characterize underwater static acoustic sources. Prajna Jandial (Ocean and Resources Eng., Univ. of Hawai'i at Manoa, 2440 Kuhio Ave., Honolulu, HI 96815, prajna@hawaii.edu), Frances Zhu (Hawai'i Inst. of Geophys. and Planetology, Univ. of Hawai'i at Manoa, Honolulu, HI), and Eva-Marie Nosal (Ocean and Resources Eng., Univ. of Hawai'i at Manoa, Honolulu, HI)

Integrating passive acoustics with autonomous platforms presents an opportunity to complement the spatial constraints of traditional fixed sensor methods by leveraging the capacity of autonomous underwater vehicles (AUVs) to make real-time decisions. We present algorithms that adaptively sample a survey region based on the sound field characteristics. These algorithms use an active learning strategy based on Gaussian Process (GP) regression to characterize a static sound field in a survey region. With each location sampled, the algorithms employ a GP to estimate the distribution and quantify the uncertainty of static acoustic sources within the region. The uncertainty metric is used to then choose the next sampling location. This dynamic approach not only maximizes the information gained by the AUV at every location that it samples but also ensures an efficient convergence toward the true distribution of underwater static sources in that region. These algorithms were developed in simulation and will be tested in controlled experiments.

10:15

3aAOa9. Underwater sound source direction finding with sparsity optimization framework using distributed fiber optic acoustic sensing system. Siyuan Cang (Southern Marine Sci. and Eng. Guangdong Lab. (Guangzhou), No.1119, Haibin Rd., Nansha District, GuangZhou, Guangdong 511458, China, cangsiyuan@gmlab.ac.cn), Huayong Yang, Chao Li, Zhongyao Wang, Xiaoming Cui, Jiantong Chen, and Dehou Yang (Southern Marine Sci. and Eng. Guangdong Lab. (Guangzhou), Guangzhou, Guangdong, China)

Distributed fiber Acoustic Sensing (DAS) technology has become increasingly popular in recent years due to its resistance to electromagnetic interference, long-distance dynamic monitoring, dense spatial sensing, maintenance-free sensing ends, and low cost. Our team conducted an underwater acoustical experiment using the DAS system at the Xin-Feng-Jiang Reservoir in Guangdong, China. This report presents the relevant experimental findings. Underwater ambient noises exhibit heteroscedastic statistics in the time/space/frequency domains, thereby posing a challenge in robustly estimating the bearings of underwater sound sources. Using the DAS system, we collected ambient noise data monthly and analyzed their statistical characteristics. To suppress the heteroscedastic noise, a robust bearing estimation method was designed for the DAS system. The proposed method relies on the generalized sparse covariance fitting and optimization framework, allowing us to capture the trajectory of a towed transducer. Furthermore, by post-processing the DAS data with high-resolution, even a fast-moving boat can be dynamically traced. Both the simulation and the experiment prove the algorithm's effectiveness, highlighting the potential of the DAS system in tracking underwater targets.

WEDNESDAY MORNING, 15 MAY 2024 ROOM 215, 10:45 A.M. TO 12:00 NOON

Session 3aAOb

Acoustical Oceanography, Underwater Acoustics and Animal Bioacoustics: Sound at Temperate and Tropical Coral Reefs

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

Chair's Introduction—10:45

Invited Paper

10:50

3aAOb1. SoundGarden: Applying healthy soundscapes to support coral reef restoration. T. Aran Mooney (Biology Dept., Woods Hole Oceanographic Inst., 266 Woods Hole Rd., Marine Res. Facilty 227 (MS# 50), Woods Hole, MA 02543-1050, amooney@whoi.edu), Nadége Aoki (Biology Dept., Woods Hole Oceanographic Inst., Woods Hole, MA), Benjamin Weiss (Woods Hole Oceanographic Inst., Woods Hole, MA), Sierra Jarriel (Biology, Woods Hole Oceanographic Inst., Woods Hole, MA), Youenn Jézéquel (Biology Dept., Woods Hole Oceanographic Inst., Trélévern, France), Alexandra Gutting, Jessica Ward (The Nature Conservancy, St. Croix, Virgin Islands (U.S.)), Weifeng G. Zhang (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA), and Amy Apprill (Marine Chemistry and Geochemistry, Woods Hole Oceanographic Inst., Woods Hole, MA)

Coral reefs are a hub of ocean biodiversity supporting a range of socioeconomic and ecosystem services. Yet reefs and these important services are imperiled due to climate change and related stressors. Healthy coral reefs can be characterized by distinctive soundscapes, and there is increasing realization that acoustic cues are vital ecosystem components. Here we present a series of experiments examining if replaying healthy soundscapes can increase the settlement of coral larvae, a fundamental ecological process. Work was conducted in the U.S. Virgin Islands, leveraging a decade of soundscape studies on those reefs, and used a novel solar-powered acoustic playback system calibrated in sound pressure and particle motion. We studied larvae from three coral species, brooding *Porites astreoides* and *Favia fragum*, and broadcast-spawning *Diploria labyrinthiformis*. Larvae on degraded reefs enriched with healthy reef sounds settled at significantly higher rates compared to control sites that were not acoustically enriched. Acoustic enrichment settlement rates were influenced by received level, ecosystem properties and species biology reflecting the need for holistically evaluating the reef restoration process. Overall, this work outlines a new, potentially scalable means of supporting healthy reef habitat, and enhancing coral settlement on imperiled reefs undergoing restoration.

Contributed Paper

11:10

3aAOb2. Unidentified fish sounds as indicators of coral reef health and comparison to other acoustic methods. Sierra Jarriel (Biology, Woods Hole Oceanographic Inst., 266 Woods Hole Rd MS#50, Woods Hole, MA 02543, sierra.jarriel@whoi.edu), Nathan Formel (Biology, Woods Hole Oceanographic Inst., Woods Hole, MA), Sophie R. Ferguson (Marine Biological Lab., Woods Hole, MA), Frants H. Jensen (Dept. of Ecoscience, Aarhus Univ., Aarhus, Denmark), Amy Apprill (Marine Chemistry and Geochemistry, Woods Hole Oceanographic Inst., Woods Hole, MA), T. Aran Mooney (Biology Dept., Woods Hole Oceanographic Inst., Woods Hole, MA), Miles J. Parsons (Australian Inst. of Marine Sci., Perth, Western Australia, Australia), and Lucia Di Iorio (Univ. of Perpignan, Perpignan, France)

As biodiversity hotspots, coral reefs are rich with sound cues. Monitoring these soundscapes for changes is essential as coral reefs decline around the world rapidly. Despite this, acoustic metrics that reliably represent reef health are still debated, and ground-truthing of methods are limited. We sought to investigate the occurrence and prevalence of fish sounds in relation to reef health, providing a foundation to compare assessment methods. We first quantified fish call rates for three U.S. Virgin Islands reefs exhibiting different community assemblages, by manually annotating fish calls across 8 days per site. Call rates were then compared with traditional visual surveys, and several acoustic methods commonly used in underwater soundscape research. Manually detected fish call rates successfully differentiated between the three reefs, capturing variation in crepuscular activity levels and predicting hard coral cover, fish abundance, and fish species richness. Meanwhile, most acoustic indices failed to parse out fine distinctions among the three sites, although sound pressure level showed the greatest correlation to call rates. Development of methods to improve the utilization of unknown fish sounds, such as automatic detection and classification tools, supports global sound library efforts and is essential to implement large-scale passive acoustic monitoring of coral reefs.

Invited Paper

11:25

3aAOb3. Unraveling biological responses to oceanographic equipment: Insights from forward-facing active acoustics and low-light imagery using small boats. Rendhy M. Sapiie (Ocean Eng., Univ. of Rhode Island, 15 Receiving Rd., Narragansett, RI 02881, renmsapiie@uri.edu), Lauren Freeman (NUWC, Naval Underwater Warfare Ctr., Newport, RI), Chris Roman, David Casagrande (Graduate School of Oceanogr., Univ. of Rhode Island, Narragansett, RI), and Brennan Phillips (Ocean Eng., Univ. of Rhode Island, Narragansett, RI)

Over six decades, extensive research has delved into the intricate interactions between marine animals and oceanographic sampling equipment, recognizing the impactful influence of light and sound emissions on mobile fauna behavior during surveys. This study, conducted through three mid-water experiments in the Solomon Islands (2019) and Bermuda (2021 and 2023), utilizing small boats, aims to unravel the effects of light on the distribution of biological scattering layers. Employing broadband active acoustics, dimmable lights, and low-light imaging on an observation-class ROV and a custom-made instrument frame, the experiments dynamically adjusted light intensity to track changes in volume backscattering strength. Amidst challenges encountered along the way, the study notably succeeded in observing avoidance behaviors from mesopelagic animals in response to the presence of our equipment. This study provides valuable insight for fisheries acoustics and echosounder applications, particularly in coral reef environments where small boats play a crucial role. This paper reviews the system setup, data processing, and presents fused backscatter data over depth, concluding with lessons learned, recommendations, and future study goals, contributing to the broader understanding of marine animal interactions with sampling equipment.

Contributed Paper

11:45

3aAOb4. Passive acoustic monitoring of tropical and temperate reefs reveals ambient noise cycles on multiple time scales. Lauren Freeman (NUWC Newport, NUWC, Naval Undersea Warfare Ctr, Newport, RI, lauren.a.freeman3.civ@us.navy.mil), Daniel Duane (NUWC Newport, Newport, RI), and Ian Rooney (NUWC Newport, Newport, RI)

Passive acoustic monitoring programs of tropical and temperate reefs in Hawaii, Bermuda, and New England have been underway for several years using omni-directional recorders placed near shallow reef sites. These multi-year time scale datasets allow us to observe not only diurnal cycles that are becoming a familiar facet of biological soundscapes, but seasonal, lunar, and inter-annual variability driven by light and sea surface temperature. Coupling acoustic data collections with non-acoustic validation such as *in situ* video and oceanographic measurements offers additional insights as to what is driving variability and increasingly predictable patterns in biological ambient noise in littoral settings. This overview will highlight work in three distinct regions with a focus on repeatable cycles associated with non-anthropogenic environmental drivers, and finally will explore the effects of human activity on reefs that may be reflected in reef soundscapes.

Session 3aBAa

Biomedical Acoustics, Physical Acoustics and Structural Acoustics and Vibration: SonoDynamic Therapy. A New Hope

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

Contributed Papers

7:55

3aBAa1. Extracellular adenosine-5′-triphosphate release kinetics following microbubble cavitation in cultured human endothelial cells. Marie Amate (Biomedical Inst., Univ. of Montréal, 900 Rue Saint-Denis, Montréal, QC H2X 0A9, Canada, marie.amate@umontreal.ca), Ju Jing Tan, Francis Boudreault, Ryszard Grygorczyk (Dept. of Medicine, Univ. of Montreal, Montreal, QC, Canada), Thomas Gervais (Dept. of Eng. Phys., Polytechnique Montréal, Montreal, QC, Canada), and François Yu (Radiology, Université de Montréal, Montréal, QC, Canada)

Ultrasound Targeted Microbubble Cavitation (UTMC) causes vasodilation, which can improve radiotherapy efficacy, but the mechanisms remain unknown. Herein, we characterize extracellular Adenosine Triphosphate (eATP) release kinetics *in vitro* following UTMC using live microscopy. eATP was measured in real-time with a bioluminescent Luciferin-Luciferase (LL) assay. Human endothelial cells were grown in microfluidic chips. A microbubble solution (Definity: 10^7 MB/ml) containing propidium iodide (25 μg/ml) and LL was added to the chips before sending a single ultrasound pulse (A303S-SU, 1MHz, 0.5 in., Olympus, pressure: 300 kPa; 10, 100, or 1000 cycles). The bioluminescence signal of eATP reaction with the LL was captured with an EMCCD camera and converted into eATP released quantity, as in the study by Tan *et al*. (2019). After the acquisition, a viability assay was done using calcein-AM (4 μg/ml). We found a significantly higher total eATP released by dead cells than live sonoporated cells. The eATP release rate was also significantly higher in dead cells than in sonoporated cells at 4s after the pulse. At that time, we could estimate the eATP released by individual cells within a cell cluster before the individual bioluminescent signals merged due to diffusion. This will able us to understand better the eATP release mechanism *in vitro* and *in vivo*.

8:10

3aBAa2. An investigation on the mechanism(s) behind sonodynamic therapy and the use of sonosentizers. Stephanie C. Walton (Eng. Sci., Univ. of Oxford, Flat 16, 1 Thames St., Oxford OX1 1SL, United Kingdom, worc6254@ox.ac.uk)

Over the last 30 years, sonodynamic therapy (SDT) has been under investigation as a treatment for malignant tumours. SDT utilizes ultrasound in combination with a sonosensitizer to cause cell death. While this treatment is in use, there remains critical questions that limit the translation of this promising therapy into the clinic. Chemical or physical changes occurring due to ultrasound may explain the efficacy observed and depending on the frequency and intensity of sound used, cavitation may be an important component of this. Some findings indicate that collapsing bubbles, or sonoluminescence, may cause the sonosensitizers to split, creating reactive oxygen species (ROS). Other research suggests damage to cell structure due to the ultrasound waves is the key. Understanding the required balance of chemical and physical effect for varied cancer cells is critical to optimization of the treatment. This paper presents a preliminary investigation of the cause of cell death observed in SDT, with focuses on ROS generation, the use of ultrasound to activate chemical compounds, and observable mechanical damage to cells following ultrasound exposure.

8:25

3aBAa3. Subharmonics are not the full story—Searching for acoustic signatures of cavitation and other fantastic beasts. Qiang Wu, Michael Gray, Cameron Smith, Robin O. Cleveland (Univ. of Oxford, Oxford, United Kingdom), Constantin Coussios (Univ. of Oxford, Headington, Oxford, Oxfordshire, United Kingdom), and Eleanor P. Stride (Univ. of Oxford, Inst. of Biomedical Eng., Oxford OX3 7DQ, United Kingdom, eleanor.stride@eng.ox.ac.uk)

Cavitation induced bio-effects are being exploiting in a wide range of applications from physiotherapy to brain surgery. The acoustic emissions generated by bubble activity are extremely useful in enabling real time treatment monitoring. The relationship between the spectral characteristics of these emissions and bubble dynamics is, however, complex. Here we report data from experiments with simultaneous ultra-high-speed optical imaging and passive acoustic mapping of individual microbubbles exposed to 50 cycles of ultrasound at 0.5 MHz with varying peak negative pressures. The spectral content of the acoustic emissions from individual microbubble was compared to the bubble dynamics observed by the imaging. As expected from prior work, both the number of discrete harmonics and broadband content in the emissions increased with increasing amplitude of bubble oscillation. There was no clear correlation, however, between the presence of ultra and sub-harmonic components and bubble behaviour. Indeed, these components were frequently absent. Moreover, phenomena such as microjetting, fragmentation and coalescence, that could produce very different effects in tissue, were indistinguishable acoustically. The results thus indicate that the definition of cavitation thresholds or doses should be very carefully considered depending on the therapeutic effect (or avoidance of unwanted bioeffects) required for a particular application.

8:40

3aBAa4. Ultrasonically induced electrical potentials in PLLA film and bone. Shouta Kitajima (Doshisha Univ., Kyotanabe, Japan, ctwj0326@mail4.doshisha.ac.jp), Keigo Maehara, and Mami Matsukawa (Doshisha Univ., Kyotanabe, Kyoto, Japan)

Osteosynthesis materials are used for the fixed treatment of serious bone fractures. Titanium and bioabsorbable poly-L-lactic acid (PLLA) are often used. PLLA is known to have piezoelectricity, and its contribution to bone fracture healing has been discussed. On the other hand, bone fracture healing using low-intensity-pulsed-ultra-sound (LIPUS) is also popular although the initial mechanism (how bones sense ultrasound) is still unclear. One key factor is the weak piezoelectricity of bone. If ultrasound was able to induce higher electrical potentials in PLLA than in bone, the combination treatment of PLLA and LIPUS would be effective for bone fracture healing. Then, we experimentally investigated the piezoelectricity of PLLA and bone in the MHz range. First, we fabricated an ultrasonic receiver using a cortical bone plate (thickness 1 mm) covered by a stretched PLLA film (thickness 50 mm)

3a WED. AM

as a piezoelectric material. Second, we irradiated ultrasound in the MHz range to the receiver and measured electrical potentials as the output of the receiver. As a result, the average electrical potentials were about 1.4 times higher than those of a receiver made of bone without the PLLA film. This result indicates that ultrasonically induced potentials around bone may increase by the PLLA film.

8:55

3aBAa5. Investigation of the ultrasound-mediated toxicity mechanisms of IR780 iodide. Kritika Singh (Univ. of Oxford, Old Rd., Headington, Oxford OX3 7LD, United Kingdom, kritika.singh@magd.ox.ac.uk), Alexandra Vasilyeva, LuNa Hu, Jia Ling Ruan, Michael Gray (Univ. of Oxford, Oxford, United Kingdom), John Schiller (National Institutes for Health, Bethesda, VA), and Eleanor P. Stride (Univ. of Oxford, Oxford, United Kingdom)

IR780 iodide is a lipophilic cation heptamethine dye that has emerged as a potential fluorescent probe for *in vivo* tumor imaging. Previous work has shown it to be a sono- and photo-sensitizer (sound- and light-activated molecule) suitable for use in sonodynamic therapy (SDT) due to its proposed ability to produce ROS when 'activated' by ultrasound. This study evaluated IR780 iodide as an SDT agent *in vitro* using A549s, HeLas, and HeLa S3 cell lines, a broad range of ultrasound and light parameters, multiple ultrasound and light systems, and with and without cavitation nuclei. Through temporal uncoupling of ultrasound application and compound administration *in vitro* and *in vivo*, evaluation of cavitation-only related cell death, assessment of the dark toxicity of IR780 iodide, and development of positive controls for cell permeabilization (sonoporation), this study shows that dark toxicity and cavitation play an important role in IR780 SDT-induced cell death. This potentially explains the high levels of cell death observed for comparatively low concentrations of ROS. Additionally, this study proposes a standard set of controls for SDT mechanistic studies, which were used to further help study the mechanisms of other SDT drugs including Rose Bengal, 5-aminolevulinic acid, and indocyanine green.

9:10

3aBAa6. Ultrasound Targeted Microbubble Cavitation (UTMC) for the treatment of Myocardial Microvascular Obstruction (MVO). Muhammad Wahab Amjad (Medicine, Univ. of Pittsburgh, 3550 Terrace St., 959 Scaife Hall, Pittsburgh, PA 15213, MUA56@pitt.edu), Soheb Anwar Mohammed, Xucai Chen (Medicine, Univ. of Pittsburgh, Pittsburgh, PA), Flordeliza S. Villanueva (Medicine/Cardiology, Univ. of Pittsburgh, Pittsburgh, PA), and John J. Pacella (Medicine, Univ. of Pittsburgh, Pittsburgh, PA)

Congestive heart failure following acute myocardial infarction is increasing due to microvascular obstruction (MVO), for which there is no effective therapy. We have been developing ultrasound-targeted microbubble cavitation (UTMC) as a potential treatment. Rapacz familial hypercholesterolemic (RFH) pigs were used in this study. MVO was created in the left anterior descending (LAD) microcirculation. UTMC therapy was applied during infusion of Definity®. Left ventricular (LV) segmental wall motion and microvascular perfusion were assessed with ultrasound. Cardiac MRI was obtained to measure infarct size and area of MVO; ultrasound imaging and coronary angiography were performed at 48h. LAD angiographic flow was improved at 48 h post-treatment in comparison to control. UTMC treatment significantly improved echo-based LV systolic performance. UTMC was also found to significantly enhance LAD blood volume at 48h versus control. MRI-derived LV segmental wall motion and ejection fraction also improved post-treatment. Infarct size was reduced as assessed by both Evans Blue/TTC staining and MRI. In conclusion, we demonstrated that UTMC significantly reduced infarct size, enhanced LAD microvascular perfusion and improved LV systolic performance.

Invited Paper

9:25

3aBAa7. Curiouser and curiouser—Sonoluminescence, sonoporation, and sonodynamic therapy. Luca Bau (Univ. of Oxford, Oxford, United Kingdom), Niclas Westerberg, Richard Lane (Glasgow Univ., Glasgow, United Kingdom), LuNa Hu, Kritika Singh (Univ. of Oxford, Oxford, United Kingdom), John Callan, Anthony McHale (Ulster Univ., Coleraine, United Kingdom), Daniele Faccio (Glasgow Univ., Glasgow, United Kingdom), and Eleanor P. Stride (Univ. of Oxford, Inst. of Biomedical Eng., Oxford OX3 7DQ, United Kingdom, eleanor.stride@eng.ox.ac.uk)

The potential of sonodynamic therapy (SDT) for delivering highly localised therapy with minimal side-effects is extremely attractive for a range of applications, including multiple forms of cancer and antibiotic resistant infections. There is also increasing evidence of beneficial immunostimulatory effects for treating metastatic disease. Yet, despite the growing evidence for both the pre-clinical and now clinical efficacy of SDT, the mechanisms underpinning ultrasound mediated drug activation remain unclear. This has inhibited optimisation of ultrasound exposure conditions and dosing protocols. This talk will review the range of mechanisms proposed in the literature and the corresponding supporting and contradictory evidence. These will include recent investigations by the authors into the role of sonoporation, and theoretical and experimental quantification of sonoluminescence. The importance of selecting appropriate treatment monitoring protocols to detect cavitation will also be discussed.

Session 3aBAb

Biomedical Acoustics: Biomedical Acoustics Best Student Paper Award Poster Session

Kenneth B. Bader, Cochair
Department of Radiology, University of Chicago, 5835 South Cottage Grove Ave., MC 2026, Q301B, Chicago, IL 60637

Kevin J. Haworth, Cochair
Internal Medicine, University of Cincinnati, 231 Albert Sabin Way, CVC 3939, Cincinnati, OH 45267-0586

The ASA Technical Committee on Biomedical Acoustics offers a Best Student Paper Award to eligible students who are presenting at the meeting. Each student must defend a poster of her or his work during the student poster session. This defense will be evaluated by a group of judges from the Technical Committee on Biomedical Acoustics. Additionally, each student will give an oral presentation in a regular/special session. Up to three awards will be presented to the students with USD $500 for first prize, USD $300 for second prize, and USD $200 for third prize. The award winners will be announced at the meeting of the Biomedical Acoustics Technical Committee.

Below is a list of students competing, with abstract numbers titles. Full abstracts can be found in the oral sessions associated with the abstract numbers. All entries will be on display, and all authors will be at their posters from 10:00 a.m. to 12:00 noon.

1aBAa8. Evaluation of sonobiopsy feasibility and safety in a mouse model of diffuse intrinsic pontine glioma
Student author: Dingyue Zhang

1aBAb5. Towards reduced ultrasound localization microscopy acquisition time by means of monodisperse microbubbles uncoupling
Student author: Giulia Tuccio

1aBAb6. Acoustic emissions-based estimation of the temporal changes in microbubble radius during ultrasonic excitation
Student author: Hohyun Lee

1aBAb7. Investigating the resonance response of a system of two ultrasound-driven lipid encapsulated microbubbles confined within a viscoelastic vessel
Student author: Hossein Yusefi

1pBAa1. Native bubble nuclei for acoustic cavitation in 3D cell cultures
Student author: Ferdousi Sabera Rawnaque

1pBAa12. Analysis of gas evolution in the heart, liver and kidney of turtles presenting with gas embolic pathology based on ultrasonography
Student author: Katherine Mary Eltz

1pBAa2. Real-time assessment of focused ultrasound-induced bioeffects in elastic tissues
Student author: Jacob C Elliott

1pBAa3. The role of fluid flow patterns in microbubble-mediated endothelial cell membrane permeabilization
Student author: Elahe Memari

1pBAa4. Focused ultrasound and microbubble induced changes in the phenotype of breast cancer cell lines
Student author: Dure S. Khan

1pBAa9. Focused shear wave beam propagation through a 3D printed human rib cage
Student author: Yu-Hsuan Chao

1pBAb5. Improving photoacoustic imaging through the skull using deep learning: a numerical study
Student author: Matthew James Olmstead

1pBAb6. Plane wave approaches with dual-frequency arrays for superharmonic contrast imaging
Student author: Jing Yang

2pBAa1. In vitro comparison of subharmonic-aided pressure estimation sensitivity among microfluidic monodisperse microbubbles, Sonazoid, and Definity
Student author: Ga Won Kim

2pBAa3. Microstreaming profile of a phospholipid-coated wall-attached microbubble undergoing shape oscillation
Student author: Hongchen Li

2pBAa5. An in vitro investigation into Lumason's utility for subharmonic-aided pressure estimation with direct comparison to Sonazoid and Definity
Student author: Hailee Mayer

2pBAa8. Extracellular matrix stiffness affects microbubble-assisted endothelial permeabilization under flow
Student author: Zoe Daniela Katz

2pBAb10. Experimental and numerical comparison of multiple passive beamformers for separating intra- and extra-canal cavitation activity during transvertebral spinal cord therapy
Student author: Andrew Paul Frizado

2pBAb2. Volumetric beamforming in real-time using commodity hardware
Student author: Sebastian Kazmarek Praesius

2pBAb3. Tunable liquid-based lenses for ultrasonic beamforming
Student author: Sina Rostami

3aBAa1. Extracellular Adenosine-5. '-Triphosphate Release Kinetics following microbubble cavitation in cultured human endothelial cells
Student author: Marie Amate

3aBAa2. An investigation on the mechanism(s) behind sonodynamic therapy and the use of sonosentizers
Student author: Stephanie Carmen Walton

3aBAa4. Ultrasonically induced electrical potentials in PLLA film and bone
Student author: Shouta Kitajima

3aBAa5. Investigation of the ultrasound-mediated toxicity mechanisms of IR7. 8. 0 iodide
Student author: Kritika Singh

4aBAa11. Validation of mSOUND using a fully heterogeneous skull model
Student author: Jeff James Bell

4aBAa12. Simulation-corrected focusing to the vertebral canal
Student author: David Martin

4aBAa2. Poroelastic model of the lungs at low frequencies predicted by Biot's theory
Student author: Arife Uzundurukan

4aBAa5. Surfacic characterization of soft tissues biomechanical properties using impact-based methods: A comparative study
Student author: Arthur Bouffandeau

4pBAa3. Ultrasound responsive multi-layered emulsions for drug delivery
Student author: Aaqib Haroon Khan

4pBAa8. Optimization of ultrasound contrast agent and treatment duration for drug delivery to methicillin-resistant Staphylococcus aureus diabetic wound biofilms in mice
Student author: Kelly VanTreeck

4pBAb1. Heterogeneous ultrasonic wave properties in leg cortical bones of thoroughbreds
Student author: Shuta Kodama

4pBAb13. Shear wave propagation in a fiber-laden viscoelastic waveguide under prestress: inverse modeling challenges
Student author: Lara Nammari

4pBAb2. Assessment of cortical bone phantom properties using ultrasonic guided waves transduced with a multi-element transducer
Student author: Aubin Antoine Chaboty

5aBAa4. Simulation of high frame rate spread-spectrum color Doppler imaging of pulsatile flow
Student author: Kian Esmailian

5aBAb10. Correlation of Escherichia coli inactivation with histotripsy bubble cloud size
Student author: Pratik A Ambekar

5aBAb12. The roles of pulse length and duty cycle in the fractionation of tendinopathic tendons
Student author: Grace M Wood

5aBAb14. Active targeting of nanotherapeutics using power cavitation imaging with a linear array transducer
Student author: Kamso Onyemeh

5aBAb3. Targeted delivery of miR-1 to the heart using clinical contrast ultrasound
Student author: Davindra Singh

5aBAb4. Monodisperse microbubble-mediated drug delivery: influence of microbubbles size on drug delivery outcome
Student author: Yuchen Wang

5aBAb5. Focused ultrasound-guided delivery of gene editing protein in human induced pluripotent stem cells
Student author: Kyle Hazel

5aBAb8. Towards real-time decompression sickness mitigation using wearable capacitive micromachined ultrasonic transducer arrays
Student author: Joshua B Currens

5aBAb9. Designing a benign prostatic hyperplasia dual-mode cavitation cloud and boiling histotripsy therapy transducer
Student author: Yashwanth Nanda Kumar

WEDNESDAY MORNING, 15 MAY 2024 ROOM 102, 8:30 A.M. TO 11:55 A.M.

Session 3aCA

Computational Acoustics, Physical Acoustics and Structural Acoustics and Vibration: The Phononic Dispersion Relations: Calculation, Interpretation, and Applications in Phononics and Metamaterials

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

Matthew D. Guild, Cochair
Naval Research Lab, 4555 Overlook Ave SW, Acoustics Division, Code 7160, Washington, D.C. 20375

Chair's Introduction—8:30

Invited Papers

8:35

3aCA1. Synthetically non-Hermitian nonlinear wave-like behavior in a topological mechanical metamaterial. Haning Xiu, Ian Frankel (Univ. of California San Diego, San Diego, CA), Harry Liu (Phys., Univ. of Michigan, Ann Arbor, MI), Kai Qian (Univ. of California San Diego, San Diego, CA), Siddhartha Sarkar (Phys., Univ. of Michigan, Ann Arbor, MI), Brianna MacNider (Univ. of California San Diego, San Diego, CA), Zi Chen (Harvard Univ., Boston, MA), Nicholas Boechler (Univ. of California San Diego, San Diego, CA), and Xiaoming Mao (Phys., Univ. of Michigan, 450 Church St., Ann Arbor, MI 48108, maox@umich.edu)

Topological mechanical metamaterials have enabled new ways to control stress and deformation propagation. Exemplified by Maxwell lattices, they have been studied extensively using a linearized formalism. Herein, we study a two-dimensional topological Maxwell lattice by exploring its large deformation quasi-static response using geometric numerical simulations and experiments. We observe spatial nonlinear wave-like phenomena such as harmonic generation, localized domain switching, amplification-enhanced frequency conversion, and solitary waves. We further map our linearized, homogenized system to a non-Hermitian, nonreciprocal, one-dimensional wave equation, revealing an equivalence between the deformation fields of two-dimensional topological Maxwell lattices and nonlinear dynamical phenomena in one-dimensional active systems. Our study opens a regime for topological mechanical metamaterials and

expands their application potential in areas including adaptive and smart materials and mechanical logic, wherein concepts from nonlinear dynamics may be used to create intricate, tailored spatial deformation and stress fields greatly transcending conventional elasticity.

9:00

3aCA2. Pseudo-spin polarized one-way elastic wave eigenstates in one-dimensional phononic superlattices. Pierre A. Deymier (New Frontiers of Sound (NewFoS), Univ. of Arizona, 1235 E. James E. Rogers Way, Mater. Sci. and Eng., University of Arizona, Tucson, AZ 85721, deymier@arizona.edu), Keith Runge (New Frontiers of Sound (NewFoS), Univ. of Arizona, Tucson, AZ), Alexander Khanikaev, and Andrea Alu (Phys. Program/NewFoS, City Univ. of New York, New York, NY)

We use a one-dimensional discrete binary elastic superlattice bridging continuous models of superlattices that showcase one-way propagation character and the discrete elastic Su-Schrieffer-Heeger model that does not. By considering Bloch wave solutions of the superlattice wave equation, we demonstrate conditions supporting elastic eigenmodes that do not satisfy translational invariance of Bloch waves over the entire Brillouin zone, unless their amplitude vanishes for some wave number. These modes are characterized by a pseudo-spin, and occur only on one side of the Brillouin zone for given spin, leading to spin-selective one-way wave propagation. We demonstrate how these features result from the interplay of translational invariance of Bloch waves, pseudo-spin, and a Fabry-Pérot resonance condition in the superlattice unit cell.

9:25

3aCA3. Multiple scattering analysis of phononic crystals. Daniel Torrent (Universitat Jaume I, Av Vicente Sos Bayntat, Castellon de la Plana, Castellón 12071, Spain, dtorrent@uji.es), Marc Martí Sabaté, and Sebastien Guenneau (Imperial College London, London, United Kingdom)

Multiple scattering theory allows for the computation of states and scattering fields on a diverse variety of wave systems. Through rigorous analysis, this theory provides insights into wave phenomena such as diffraction, reflection, and transmission, contributing to a comprehensive understanding of the intricate dynamics of classical systems. Our study focuses on flexural waves in thin elastic plates, with point-like mass spring resonators serving as scatterers attached the plate's top surface. Finite clusters of scatterers will be analysed without imposing symmetries or periodicity in the system. Despite the apparent simplicity of this model, it proves capable of unveiling and predicting many behaviours related to the latest topics on metamaterials, including phononic crystals, quasicrystals, topological insulators, rainbow trapping effects or spatio-temporal modulation. In essence, this technique offers a foundation for advancements in materials science, structural engineering and all the domains where metamaterials and phononic crystals find application.

9:50–10:05 Break

10:05

3aCA4. Considering the role of the phononic dispersion relations in understanding the performance of phononic pseudocrystals and similar structures. S. Hales Swift (Sandia National Labs., P.O. Box 5800, MS 1082, Albuquerque, NM 87123-1082, shswift@sandia.gov)

Dispersion calculations have been an essential tool for investigating and predicting the performance of conventional phononic crystals because they offer the opportunity to accurately characterize the behavior of a structure based on the smallest characteristic piece of that structure. For phononic pseudocrystals—in which translational symmetry is replaced by cyclic symmetry and radial self similarity—the scaled dispersion relations of the unit pseudocell provide useful information; however, the behavior of a completed article comprised of multiple pseudocells is less completely articulated than for conventional phononic crystals. This paper will explore the uses of dispersion-type calculations applied to characteristic wedges of phononic pseudocrystals in an effort to evaluate what additional information they can potentially provide. [*SNL is managed and operated by NTESS under DOE NNSA contract DE-NA0003525*.]

10:30

3aCA5. Inverse design of arbitrary non-reciprocal acoustic dispersion in non-local phononic crystals and metamaterials. Muhammad Bilal Khan (Mech. Eng., Stevens Inst. of Technol., Hoboken, NJ) and Christopher Sugino (Mech. Eng., Stevens Inst. of Technol., 1 Castle Point, Hoboken, NJ 07030, csugino@stevens.edu)

We present inverse methods to obtain arbitrary non-reciprocal dispersion relations in periodic, multi-degree-of-freedom lattices with non-local interactions. Just as phase-delayed interactions such as local resonance govern the frequency-dependence of a lattice's dispersion relation, spatially non-local interactions govern the wavenumber dependence of the coefficients of the lattice's characteristic equation. Thus, the use of fully general non-local interactions can, in principle, specify every branch of the dispersion diagram at every wavenumber. However, calculating a viable set of non-local interactions for a given set of desired dispersion curves is challenging, as there are multiple possible sets of coefficients that yield the same polynomial roots. To solve this problem, we first present a general method to calculate the dispersion relation of multi-degree-of-freedom non-local lattices, showing that non-local interactions create a Fourier series expansion that governs the wavenumber dependence of each matrix element. Next, we present numerical techniques to solve the inverse problem—i.e., given the desired dispersion relation, calculate the required non-local interactions—and discuss the associated computational challenges. Finally, we discuss practical methods to realize non-local interactions leveraging piezoelectric sensors and actuators, and we highlight how non-local interactions can be introduced in unit-cell based finite-element dispersion calculations.

10:55–11:55
Panel Discussion

Session 3aED

Education in Acoustics and Musical Acoustics: Artistic and Technical Approaches to Acoustics

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

Andrea Calilhanna, Cochair
Faculty of Arts, Elder Conservatorium of Music, The University of Adelaide, New South Wales, Sydney, 2126, Australia

Olivier Robin, Cochair
Université de Sherbrooke, 2500, bd de l'université, Faculté de Génie - Dpt Génie Mécanique, Sherbrooke, J1H1L2, Canada

Chair's Introduction—8:30

Invited Papers

8:35

3aED1. Beatboxing: Teaching technical acoustical concepts through pop culture. Alex C. Brown (Phys. and Astronomy, Brigham Young Univ., N 283 ESC, Provo, UT 84602, alexbrownbass@gmail.com), Micah Shepherd, and Tracianne B. Neilsen (Phys. and Astronomy, Brigham Young Univ., Provo, UT)

The motivation for this presentation is to demonstrate that vocal percussion, or "beatboxing," can be a rich method of teaching and demonstrating various introductory acoustical concepts. Beatboxing can be an entertaining way to demonstrate the different acoustical sources of energy—namely burst, noise, and voice—and how they can be combined to create familiar and unfamiliar phonemes, such as the "PF" snare. It also provides an engaging way to introduce the anatomy of the vocal tract as the students can see and hear how vibrating different parts of the mouth and throat (e.g., vocal cords, epiglottis, vestibular folds, lips, etc.) alter the sound. Finally, beatboxing can be used to demonstrate the source-filter model of speech production through altering the sound source in the throat and displaying how to change your mouth shape to produce different formant-altering effects, such as the rapid formant shifting that gives a record scratch its characteristic sound. Beatboxing provides an entertaining and effective way to demonstrate technical concepts to students in an artistic format which captures the attention, especially of younger audiences. [Undergraduate research supported by the College of Physical and Mathematical Sciences, at Brigham Young University.]

8:55

3aED2. An artistic approach is taken that uses principles in quantum mechanics to understand the signature sound created by granular synthesis in an undergraduate course on music synthesis techniques. Jill A. Linz (Phys., Skidmore College, 815 N Broadway, Saratoga Springs, NY 12866, jlinz@skidmore.edu)

Music Synthesis Techniques is an advanced undergraduate course that explores how various methods are used to create and manipulate synthesized sounds used in sound design and audio production. Students explore their understanding through the creation of a series of short, one-minute compositions that utilize each technique. This paper focuses on that of granular synthesis. Granular synthesis is a music synthesis technique that is used to create unique and alluring sounds that has captivated modern music listeners. For audio engineers and sound designers, the term granular synthesis has always held an aura of mystery surrounding it. Recent developments in plugins for digital audio workstations allow them to easily create its signature sound, which have created a way to use granular synthesis as an art form. Unlike other methods, granular synthesis is rooted in the laws of quantum mechanics. The Heisenberg Uncertainty Principle is explored to understand how granular synthesis creates its signature sound, which in turn is then used to create artistic compositions through the manipulation of the variables. Several examples from student work as well as collaborative work will be included.

9:15

3aED3. Emotion equalization app: The effectiveness of dynamic music therapy approaches. 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)

A recent study explored how static music playlists of relaxing or uplifting music can improve our mood. Unlike static playlists that rely exclusively on relaxing, uplifting, or consoling music, this study explores eight distinct methods to curate playlists. These approaches include transitions from one dimension to another (e.g., Angry-to-Relaxing, Angry-Uplifting, Relaxing-Uplifting, Relaxing-

Natural, Sad-Relaxing, Sad-Uplifting, Uplifting-Relaxing, and Uplifting-Natural). To evaluate the effectiveness of these playlists, online experiments were conducted, wherein participants were surveyed before and after music listening to monitor changes in valence and arousal. The results reveal that Sad-Relaxing, Relaxing-Uplifting, Relaxing-Natural, and Uplifting-Natural playlists give positive changes in participants' valence levels, indicating mood improvement. As for the arousal level, the Angry-Relaxing playlist made people feel calmer. Correspondingly, the Relaxing-Uplifting playlist made people feel more energetic. Moreover, this study has identified specific pathways on a two-dimensional plane where both arousal and valence values could be profitably to good effect, including transitioning from Angry to Relaxing to Uplifting especially for arousal, and Sad to Relaxing or Uplifting especially for valence. These findings highlight how dynamic music can give rise to distinct emotional responses and influence our energy levels, and give guidelines on how to use music therapy playlists for mood enhancement.

9:35–9:50 Break

Contributed Paper

9:50

3aED4. Extended playing techniques as a source of unusual acoustic phenomena to explore with music students. Montserrat Pàmies-Vilà (Dept. of Music Acoust. - Wiener Klangstil (IWK), Univ. of Music and Performing Arts Vienna, Anton-von-Webern-Platz 1, mdw - Inst. 22, Vienna 1030, Austria, pamies-vila@mdw.ac.at)

When teaching acoustics, instructors typically consider musical instruments to introduce students to the science of sound. Common examples, such as the flute as an open pipe or the behavior of a string fixed at both ends, are often used to illustrate the acoustics of musical instruments. In addition, when teaching acoustics to advanced musicians, one could take advantage of the precise prior knowledge that students have regarding both conventional and extended playing techniques. When using extended techniques, musicians may incorporate unusual ways of producing sound with their instruments, as is often found in contemporary classical compositions. This unusual sound production might even result in a counterexample to the classical textbook case. For example, the so-called slap-tongue technique on the saxophone or clarinet produces a free (and highly damped) oscillation of the reed, which contrasts with the self-sustained oscillations of the reed if the instrument is blown as usual. Based on insights gathered during an introductory course in musical acoustics at the mdw—University of Music and Performing Arts Vienna, this paper highlights how extended techniques not only challenge our understanding of musical acoustics but also serve as interesting phenomena to explore interactively with students in the lecture room.

Invited Paper

10:05

3aED5. Fantastic acoustics: A magazine describing the importance and the scope of acoustics research in Québec. Olivier Robin (Université de Sherbrooke, 2500, bd de l'université, Faculté de Génie - Dpt Génie Mécanique, Sherbrooke, QC J1H1L2, Canada, olivier.robin@usherbrooke.ca), Tamara Krpic, Alexis Carrion, François Proulx (Université de Sherbrooke, Sherbrooke, QC, Canada), Michel Demuynck, Lucie Gallerand (Mech. Eng., ETS (Ecole de technologie superieure), Montréal, QC, Canada), Valérian Fraisse, Christopher Trudeau, Cynthia Tarlao (McGill Univ., Montreal, QC, Canada), Cécile Perrier de la Bathie, Coralie Bernier-Breton (Institut des Sci. de la mer (ISMER), Université du Québec à Rimouski (UQAR), Rimouski, QC, Canada), Thomas Dupont, Olivier Doutres, Jeremie Voix (Mech. Eng., ETS (Ecole de technologie superieure), Montréal, QC, Canada), Pierre Cauchy, Guillaume St-Onge (Institut des Sci. de la mer (ISMER), Université du Québec à Rimouski (UQAR), Rimouski, QC, Canada), and Catherine Guastavino (McGill Univ., Montreal, QC, Canada)

It's often difficult for the general public and young people to grasp the importance of acoustics in everyday life and its many branches and fields of application. However, it's equally complex for people working or studying acoustics to communicate to this audience while avoiding the same focus on detail as scientific publications. Science communication skills are seldom taught, and multidisciplinary approaches are little used. This paper describes the work carried out as part of the 'Fantastique acoustique—Fantastics acoustics' project, which aims to raise awareness of the acoustics-related research at four Québec universities and train students in science communication. Eight student teams were partnered with cartoonists to produce 3-page comic strips and an associated corpus (including references and videos). The steps involved in producing the magazine are described, as are some of the final works. In particular, a survey including the students and the artists identifies the main positive points and the most significant challenges of such a partnership.

Contributed Paper

10:25

3aED6. What's that sound? Real world observations as teaching opportunities. John A. Case (Graduate Program in Acoust., Penn State, 201 Appl. Sci. Bldg., University Park, PA 16802, jac7175@psu.edu)

In our day-to-day lives, we encounter numerous interesting acoustic events that can offer a starting point for a rich learning experience. In the past, making recording and performing signal analysis took expensive equipment and expert knowledge. Now, smart phones offer a readily available measurement tool to capture interesting phenomena in the world around us. A simple phone video file can provide audio for introducing signal processing techniques as well as a motivating visual context, and readily available signal processing tools enable for the investigation of these events. Several examples of such events will be presented and discussed. These examples are used as a motivation for discussion and signal processing analysis, trying to uncover the underlying physics behind each example.

10:40–11:15
Panel Discussion

Session 3aMU

Musical Acoustics, Physical Acoustics and Computational Acoustics: Applications of Physical Modeling of Musical Instruments

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

Mark Rau, Cochair
Music, Stanford University, 660 Lomita Court, Stanford, CA 94305

Chair's Introduction—8:30

Invited Papers

8:35

3aMU1. An open-source project for wind instrument modeling using digital waveguides. Gary Scavone (Music Res., McGill Univ., 555 Sherbrooke St. West, Montreal, QC H3A 1E3, Canada, gary.scavone@mcgill.ca)

Digital waveguide methods for time-domain modeling of acoustic structures comprised of interconnected piece-wise one-dimensional cylinders, cones and toneholes have been well reported in the past. In this work, an open-source Matlab class is presented that can be used to model arbitrary wind instrument air column structures, defined by length, radii and hole geometric parameters, using digital waveguide methods. Fractional-delay filtering can be enabled or disabled, with support for both Thiran and Lagrange filter types of specified order. Thermoviscous losses can be modeling using either a discrete-time least squares or a shelf-filter fit, both of arbitrary order. Three different tonehole model options are provided (two-port, three-port or wave digital filter). End boundary conditions include closed, anechoic, ideally open, open flanged and open unflanged. The class object can be used either in single-sample iterative contexts (for example, with an attached excitation model) or with arbitrarily-sized input signals (for example, to compute a reflection function or impedance). The modeled geometry can be constructed manually by iteratively adding segments or holes, or a complete geometry can be specified in a file. This project is bundled together with a parallel and previously reported open-source project for frequency-domain transfer matrix modeling of air columns (https://github.com/garyscavone/acmt).

8:55

3aMU2. Time-domain simulation of a recorder using the model of a jet allowed to deflect overall. Seiji Adachi (School of Marine Sci. and Technol., Tianjin Univ., 92 Weijin Rd., Nankai District, Tianjin 300072, China, seiji_adachi@yahoo.co.jp)

A sound production model of the flute-like instrument which allows the jet to deflect overall has been proposed recently. The overall deflection is introduced to satisfy flow consistency at the instrument's window (embouchure hole or mouth), i.e., the condition that the volume velocity swept by the oscillating jet should be equal to that through the window. The full oscillation of the jet is the sum of the deflection and the oscillation caused by fluid dynamical sinuous instability. This model can correctly reproduce the reflection function of a recorder's head as seen from the resonator. The next question is whether the model can predict the sound level of the flute-like instrument correctly. To answer the question, the model was first formulated in the time domain. In this formulation, measures were taken to prevent the oscillation from drifting. Using the time-domain model, physical modeling simulation of a recorder was performed to obtain sound levels produced for various blowing pressures. Artificial blowing experiments with a real instrument are in progress to compare the simulation results with the actual sound levels.

9:15

3aMU3. Spectral analysis of the air flow pattern near the labium of a recorder. Nicholas Giordano (Phys., Auburn Univ., College of Sci. and Mathematics, Auburn, AL 36849, njg0003@auburn.edu)

Navier-Stokes-based simulations of the air flow through a recorder allow detailed tests of analytic theories of how the pattern of flow—specifically the creation and time evolution of vorticity near the labium—gives rise to the sound of the instrument. A new spectral analysis algorithm has been developed to extract the flow pattern at the fundamental frequency for a soprano recorder playing a steady tone. This analysis allows the flow pattern at the fundamental frequency to be separated from the flow patterns at other frequencies and from the uniform flow pattern, i.e., the time independent flow associated with the air jet emerging from the flue. This gives an especially clear elucidation of the time dependent process in which vortices are created and evolve during the course of a musical tone. This analysis technique can also be used to extract the flow patterns at the frequency of the second partial and at the frequency of the recently discovered half harmonic [1], which show a geometrical structure that is distinctly different from that found at the fundamental frequency. Our spectral analysis technique should allow more detailed tests of the theory of vortex produced sound for a flue instrument than have been possible to date. N. Giordano and K. L. Saenger, J. Acoust. Soc. Amer. **154**, 2917 (2023). [Work supported by NSF under Grant No. PHY2306035.]

9:35

3aMU4. Physical modeling and time-domain simulation of a piano. Eiji Tominaga (Res. and Development Div., Yamaha Corp., 10-1 Nakazawa-cho, Chuo-ku, Hamamatsu, Shizuoka-prefecture 430-8650, Japan, eiji.tominaga@music.yamaha.com) and Masanao Sato (Res. and Development Div., Yamaha Corp., Hamamatsu, Shizuoka-prefecture, Japan)

The elaborate method to simulate the sound produced by a piano is presented, using the physical model that considers almost all parts of the piano with wood complex elastic orthotropy and manufacturing stress, but excluding action parts. The purpose of this work is to improve the efficiency of piano development by clarifying the causal relationship between the design and the produced sound. The completed piano is modeled as a 3D coupled system consisting of "nonlinear systems including hammers and strings" and "a linear coupled body-air system." The hammer felt property are represented by the double layer nonlinear generalized Maxwell model. For the strings with geometrical nonlinearity and their support end anisotropy, the Galerkin method with the component mode synthesis (CMS) method is applied. And also, for the large-scale linear coupled body-air system, the complex CMS method via nonlinear eigenvalue analysis is used. The simulated sounds, including the ringing and body sounds, are so realistic that they help piano designers predict the actual sounds before making heavy prototypes. The method presented here can be applied to other musical instruments that consist of strings and a body, such as guitars and violins.

9:55–10:10 Break

10:10

3aMU5. Towards better copies of guitars: Compensate material variability with geometry. Pierfrancesco Cillo (Inst. of Eng. and Computational Mech., Univ. of Stuttgart, Stuttgart, Germany), Pascal Ziegler (Inst. of Eng. and Computational Mech., Univ. of Stuttgart, Pfaffenwaldring 9, Stuttgart 70569, Germany, pascal.ziegler@itm.uni-stuttgart.de), and Peter Eberhard (Inst. of Eng. and Computational Mech., Univ. of Stuttgart, Stuttgart, Germany)

Luthiers have been trying to copy the sound of iconic instruments like Torres guitars for many years. Though precise geometric copies were manufactured, audible differences were found which can be attributed largely to the natural variability of wood. We were able to present a methodology allowing non-destructive material identification based on which a shape optimization was performed to compensate material variablility with specific geometric variations, allowing a much more exact copy of a guitar soundboard in terms of eigenfrequencies. We will present a generalization of this methodology to full guitars including almost arbitrary geometric adaptions and consideration of mode shapes in the optimization. A mesh morphing strategy allows to simultaneously define very general geometric adaptions and comparing mode shapes without re-meshing. Further, a parameterized reduced order numerical model will be presented in which all possible global geometric variations are mapped to the individual elements of the underlying finite-element model in symbolical form. This symbolical representation is then carried over to the assembled finite-element matrices resulting in an extremely efficient reduced model keeping the symbolic relationships. This is the absolute core requirement of an efficient shape optimization. The approach will be presented for hexahedral elements.

Contributed Papers

10:30

3aMU6. Influences of electric guitar pickup magnetic force on string vibrations. Takuto Yudasaka (Res. & Development, Yamaha Corp., 10-1, Nakazawa-cho, Naka-ku, Hamamatsu, Shizuoka 4300912, Japan, takuto.yudasaka@music.yamaha.com), Gary Scavone (Music Res., McGill Univ., Montreal, QC, Canada), and Kenta Ishizaka (Res. & Development, Yamaha Corp., Hamamatsu, Default Choice, Japan)

The design of a magnetic pickup has a significant impact on the tone of an electric guitar. In particular, the magnetic force from the pickup can influence the string vibrations and cause a beating effect, depending on the strength of the magnetic force and the amplitude of the string vibrations. To understand this behavior, we performed experiments and simulations. We measured and modeled the transversal string restoring force at various displacements from the magnetic pickup in directions both parallel and perpendicular to the guitar body. We observed that the magnetic force distribution is asymmetric in the perpendicular direction but symmetric in the parallel direction, which distorts the frequency of vibrations in opposite ways for the two directions and causes a resultant beating. This effect increases with larger string amplitudes or greater magnetic force and also varies over the duration of a plucked tone.

10:45

3aMU7. Aeroacoustic modeling of blown-closed free reeds. Ninad V. Puranik (Music Res., McGill Univ., 550, Rue Sherbrooke Ouest, Area (Ste. 500), Montreal, QC H3A1E3, Canada, ninad.puranik@mail.mcgill.ca) and Gary Scavone (Music Res., McGill Univ., Montreal, QC, Canada)

We present ongoing work on the development of an aeroacoustic model of blown-closed free reeds used in a hand harmonium. Physics-based models of free reed instruments involve modeling of the oscillating free reed, the air-flow in the upstream region and around the reed, and the interaction between these two systems. The minimal model of free reeds by Millot and Baumann (2007) assumed the reed as a damped spring-mass system and approximated the air flow field by two discrete zones representing the flow near the reed and the upstream region respectively. Our recent work included the adaptation of the minimal model to match the physical setup and control parameters of the hand harmonium and the development of a distributed clamped-bar model of the free reed. These updates resulted in a closer agreement of the parameters for simulation and the observed physical variables. To address limitations of this model, we present a revised modeling of the air flow as a continuous potential flow and discuss implications for numerical stability, accuracy and real-time sound synthesis.

11:00

3aMU8. Results from a time domain clarinet model: Effects of non-harmonic air column resonances. Stephen C. Thompson (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, sct12@psu.edu)

A time domain model of a clarinet-like system was previously reported [S. C. Thompson, *J. Acoust. Soc. Am.*, **150** A98 (2021)] . That model has been used to investigate the improvement in playing behavior when the instrument air column has accurately harmonic input impedance peaks. Two tube models were designed with an effective length to produce a 400 Hz first resonance frequency. Both tubes are terminated at the "closed" end by a SDOF reed with a reasonably high resonance frequency. The first tube is purely cylindrical terminated at the "open" end by a tone hole lattice with a cutoff frequency of 1500 Hz. This tube has two resonances below cutoff whose peaks differ from true harmonicity by approximately 7%. The second tube has a small region of slightly increased bore radius positioned to provide resonances whose inharmonicity is reduced to less than 1%. These tubes have very similar playing behavior in the model at low playing amplitude, but increasingly differ as the playing level is increased. The model has also been upgraded to include a mechanical hard stop for the reed position as it closes against the mouthpiece. Preliminary results with the beating reed will be presented.

11:15

3aMU9. Sound synthesis applications using a physical model of a single-reed woodwind instrument. 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), Alex Hofmann, and Montserrat Pàmies-Vilà (Dept. of Music Acoust., Univ. of Music and Performing Arts Vienna, Vienna, Austria)

Physical modeling can be used to analyze and predict the vibrations of a musical instrument. This also enables to numerically synthesize sounds as if they were produced by a real instrument. Focusing on single-reed woodwind instruments, a physical model should incorporate the actions of the player in order to synthesize realistic sounds. This interaction mostly takes place at the instrument mouthpiece—toneholes opened by the player's fingers may be approximated by changing the instrument geometry. A physical model is presented that is able to take embouchure effects into account, in order to reliably simulate note transitions. Regarding physical modeling synthesis, the numerical efficiency of the underlying algorithms should also be considered. Capturing as many physical phenomena as possible may lead to models that require longer running times. Omitting less significant phenomena may lead to models suitable for real-time performance, while retaining most of the sound characteristics of the real instrument. In either case, it is of utmost importance to prove that the simulation algorithms are numerically stable. Sound examples are presented in an attempt to imitate real recordings, as well as in a live performance setting (examples available at https://iwk.mdw.ac.at/sound-synthesis/)

WEDNESDAY MORNING, 15 MAY 2024 ROOM 205, 8:00 A.M. TO 11:45 A.M.

Session 3aNSa

Noise, Architectural Acoustics, Engineering Acoustics, and Physical Acoustics: Assorted Topics on Noise I

Aaron B. Vaughn, Chair
Structural Acoustics Branch, NASA Langley Research Center, 1 NASA Drive, Hampton, VA 23666

Contributed Papers

8:00

3aNSa1. Solving noise issues: Understanding feasible and effective temporary and permanent noise control options. Matt Cott (Behrens & Assoc., Environ. Noise Control, 9536 E I-25 Frontage Rd., Longmont, CO 80504, mcott@baenc.com)

Development projects, especially in densely populated areas, face a critical challenge in managing noise impact levels resulting in an increase in specification of noise regulations in bid documents. Uncontrolled noise issues carry the potential for offsite receiver complaints and regulatory actions, necessitating a strategic approach to compliance with project noise level specifications and municipal noise ordinances while understanding what feasible mitigation solutions are readily available are key to compliance and minimizing community disruption while working with project budgets. The presentation will focus on ways to work with the project owner and acoustical consultant to find solutions that are effective, feasible, readily available and take costs into consideration.

8:15

3aNSa2. Adaptive zone based active noise control for a moving target. Wintta Ghebreiyesus (Aerosp. Eng., Toronto Metropolitan Univ., 350 Victoria St., Toronto, ON M5B 2K3, Canada, wghebrei@torontomu.ca), Sifat Hasan (Aerosp. Eng., Toronto Metropolitan Univ., Toronto, ON, Canada), and Fengfeng (Jeff) Xi (Aerosp. Eng., Toronto Metropolitan Univ., Toronto, ON, Canada)

This study focuses on enhancing active noise control (ANC) in room enclosures, specifically targeting zones of quiet (ZoQ) around aircraft passenger seats. By integrating virtual sensing and motion tracking techniques, we aim to dynamically adapt the ZoQ to moving targets. Key to our approach is the strategic placement of actuators and sensors, forming the core of the ANC system. Our methodology includes virtual sensing for ANC analysis, ZoQ optimization for varied applications, and in-depth case studies. We introduce an innovative combination of a speaker gimbal system, a vision system, and custom software for precise motion tracking, significantly improving ZoQ localization. The findings offer insights into maintaining effective ZoQs for multiple input multiple output (MIMO) local ANC configurations, laying the groundwork for adaptive ZoQ control about sound sources and desired cancellation locations. This research marks a significant step towards more effective and adaptable noise cancellation in enclosed spaces.

8:30

3aNSa3. Bowling alleys in residential buildings—Noise control and measurements. Guangsheng (Sam) Du (Valcoustics Canada Ltd., 25-30 Wertheim Court, Richmond Hill, ON L4B1B9, Canada, sam@valcoustics.com), Mark Levkoe, Jessica Tsang, and Qiang (Richard) Li (Valcoustics Canada Ltd., Richmond Hill, ON, Canada)

In the competitive world of residential development, developers are striving to construct residential buildings that will be more desirable to occupants by providing a diverse range of amenity spaces, such as a bowling alley. Understandably, noise concerns from the addition of bowling alleys in residential buildings were raised during the design stage of various projects. To address the issues, a jack-up floating floor system was proposed for the bowling alley to mitigate the noise impact from the pin setting machine, initial ball impact on the bowling alley lanes, rolling on the lanes, and ball impact with the pins. Apparent Impact insulation class (AIIC) and heel drop testing were completed at various stages during the floating floor construction to confirm the impact sound isolation performance of the floating floor system. Testing results concluded that the implemented floating floor system reduced impact noise significantly and satisfactory sound levels resulted in the closest occupied spaces.

8:45

3aNSa4. Challenges in urban data center design. Scott Hamilton (Stantec, 233 S Wacker Dr #5300, Chicago, IL 60606, scott.hamilton@stantec.com)

The data center market is experiencing significant growth, with each new facility requiring increased power density and cooling needs. This growth brings new challenges, including noise pollution and its impact on nearby communities, which becomes excessively difficult in urban environments. This presentation will explore the issues faced by data centers in urban environments and discuss potential solutions to create quieter facilities. Key points include:—Physical challenges: Every site design faces numerous physical constraints, including limited rooftop and equipment room space, tight and complex adjacencies of neighboring sites, and design and performance standards for equipment.—Political issues: Every site and community have a different set of laws and regulations that a site needs to meet, and often go beyond to ensure that the community will accept the site.—People issues: Data centers must navigate contentious relationships with neighbors and address noise concerns to maintain a positive reputation. By addressing these challenges, data centers can minimize noise pollution and maintain positive relationships with surrounding communities, ensuring a sustainable future for the industry in urban environments.

9:00

3aNSa5. Survey of background noise levels inside classrooms on a college campus. Laura Ruhala (Mech. Eng., Kennesaw State Univ., 840 Polytechnic State University, Rm. Q319, MD 9075, Marietta, GA 30060, lruhala@kennesaw.edu), Richard J. Ruhala, Antonio Patino, and Charles Packer (Mech. Eng., Kennesaw State Univ., Marietta, GA)

Sound level measurements were made for 18 classrooms and learning spaces on the Marietta Campus of Kennesaw State University, Georgia, USA. Excessive background noise in learning spaces diminishes the speech intelligibility from the speaker to the listeners. The noise measured was in empty or nearly empty classrooms with no one talking. The primary background noise source was observed to be due to heating, ventilation, and air-conditioning equipment. The results are compared with a student survey to see if the noise complaints correlate with the noisiest classrooms in terms of their average A-weighted decibel levels. In addition, conformance with ANSI-ASA S12.60-2010/Part 1 (Acoustical Performance Criteria, Design Requirements, and Guidelines for Schools) is evaluated.

9:15

3aNSa6. Background noise and reverberation time in a college classroom. Richard J. Ruhala (Mech. Eng., Kennesaw State Univ., 840 Polytechnic Ln., KSU - Mech. Eng. Dept. Rm. Q131, Marietta, GA 30060, rruhala@kennesaw.edu), Laura Ruhala, Charles Packer, and Antonio Patino (Mech. Eng., Kennesaw State Univ., Marietta, GA)

A sound survey of noise levels in unoccupied classrooms on the Marietta Campus of Kennesaw State University identified one classroom as the most problematic in terms of background noise levels. This level is mostly associated with heating, ventilation, and air-conditioning equipment. Excessive background noise in learning spaces diminishes the speech intelligibility from the speaker to the listeners and can cause stress and fatigue. The overall A-weighted average sound levels are compared with ANSI-ASA S12.60-2010/Part 1 (Acoustical Performance Criteria, Design Requirements, and Guidelines for Schools). In addition, frequency analysis of the background sound and time variations are studies. Binaural headset microphone measurements are compared to the single microphone measurements. Reverberation time is measured using the engineering survey method and compared to the ANSI guidelines.

9:30

3aNSa7. Development and verification of a method to simulate non-stationary vehicle interior wind noise. Jinghe Yu (Ray W. Herrick Labs., Purdue Univ., 3879 Amber Ln., West Lafayette, IN 47906, yu1140@purdue.edu) and Patricia Davies (Ray W. Herrick Labs., Purdue Univ., West Lafayette, IN)

As speeds and directions of the vehicle and wind change, the time-varying flow creates variations in wind noise, which can be referred to as non-stationary wind noise. To investigate people’s perceptions of non-stationary wind noise inside the vehicle, a method to simulate the non-stationary wind noise is needed. Previously, a method was developed that used stationary wind noise recordings taken at several constant wind speeds and directions to form functions that relate the 1/3 octave sound pressure level with wind speed and direction. These functions are used to create time-varying filters based on provided time histories of wind speed and direction. To reduce the cost of taking many stationary measurements, an improved method was investigated. At each yaw angle, one speed sweep wind tunnel measurement was used to estimate the relationship between sound pressure level and wind speed. Two partially correlated white noise signals were then filtered to simulate binaural sounds that had a similar coherence structure between the left and right ear sounds to that observed in binaural measurements in the vehicle. The accuracy of the simulations was validated by comparing wind noise simulations to vehicle interior noise measured in the wind tunnel and on the road.

9:45–10:00 Break

10:00

3aNSa8. Identifying noise control strategies for variable air volume (VAV) boxes in acute care hospital design. Jessica Carolina (Acoust., bkl Consultants Ltd., 3999 Henning Dr., Burnaby, BC V5C 6P9, Canada, carolina@bkl.ca)

Variable air volume boxes are frequently used within new acute care hospital design of heating, ventilation and air conditioning systems in Canada. Spatial and room-use noise limits as defined within the project requirements [PM1] are often necessarily onerous to provide acoustical conditions that promote well-being and patient recovery, with appropriate noise control design crucial to the success of meeting the project requirements. Additionally, the desire for fiber-free linings to ductwork exacerbates the noise control limitations. This paper will review the available noise control strategies, the acoustic performance of fiber-free variable air volume box types with and without an attenuator and identify cost-benefits to the Design-Builder. This study will demonstrate how the implementation of a variety of variable air volume box models, sizes, operating conditions, pressure drops are affecting the noise performance. This study will summarize the appropriate variable air volume box types and design conditions that meet the project noise limits used in Canadian healthcare standards such as CSA Z8000, LEED and other provincial technical guidelines.

10:15

3aNSa9. Simulink modeling of carbon nanotube thin film thermophones for applications in active noise control. Kourtney Libenow (Acoust., Graduate Program in Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, kra5346@psu.edu) and Andrew Barnard (Acoust., Penn State, University Park, PA)

Carbon Nanotube (CNT) Thin Film Thermophones are transducers that operate by varying the input voltage to rapidly heat and cool the thin film, resulting in a local fluid pressure variation which results in radiation of sound. These devices are promising for active noise control in HVAC due to being able to withstand higher ambient heat, requiring no rare earth materials, and being able to stretch into many different configurations. An important step to future HVAC implementation is to model an ANC system using CNT thermophones as the canceling speaker. CNT thermophones have a nonlinear response due to the proportionality between input electrical power and output pressure. In order to use them in active noise control applications, preprocessing techniques such as DC offset have been used. A traditional FXLMS system with preprocessing is simulated using Simulink, with an aim to evaluate different nonlinear control structures in the future.

10:30

3aNSa10. Quantifying crowd engagement with machine learning. Jason D. Bickmore (Phys., Brigham Young Univ., N283 ESC, Provo, UT 84602, jdb387@byu.edu), Mitchell C. Cutler, Mark K. Transtrum (Phys. and Astronomy, Brigham Young Univ., Provo, UT), and Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Inferring a crowd's engagement at a sporting event using only acoustic signals is difficult for humans to do precisely, but it may be a great job for machine learning. This presentation will summarize the results of using acoustic data from crowds at four types of college sporting events to train a random forest machine-learning model to identify behaviors such as cheering, chanting, and distracting the opposing team. The model uses spectral data and other features derived from acoustic measurements such as spectral slope, spectral kurtosis, and zero cross rate to quantify the level of crowd engagement. This presentation will then discuss an ongoing investigation of how the model performs at predicting crowd engagement from acoustic data collected with a new data-collection system. It will discuss specific challenges and opportunities associated with incorporating new data into an existing machine-learning pipeline.

10:45

3aNSa11. Construction noise modelling—A comparison of equipment noise emissions data sources. Sam Zokay (1004 Middlegate Rd #1100, Mississauga, ON L4Y 1M4, Canada, samz@aercoustics.com)

The accuracy of any environmental noise impact model depends on the quality and application of its input data, typically consisting of at-source emission levels, propagation factors, and receiver characteristics. In the realm of construction noise modelling, this starts with the determination of the noise emission level for each construction activity or item of equipment. This data can be procured from a variety of sources including manufacturer data, standards and guideline documents, or proprietary measurements. This study explores how emission levels from different data sources can be utilized to assess noise impacts against project criteria, and aims to validate the reliability through real-world case studies.

11:00

3aNSa12. Case study of construction noise and vibration monitoring at an elementary school. Galen Wong (Stantec Consulting Ltd., 300 - 1331 Clyde Ave., Ottawa, ON K2C3G4, Canada, galen.wong@stantec.com)

Construction noise and vibration monitoring was performed for long-term construction activities occurring directly adjacent to an elementary school and playground area. Noise and vibration limits were determined based on measurements taken indoors and outdoors during preliminary construction activities, and in discussions with the school. Noise and vibration alerts during monitoring were provided in real-time to the construction managers during the activities, and notices of exceedances provided as methods of administrative controls. Various temporary noise mitigation barriers erected on-site were also assessed.

11:15

3aNSa13. Active cancellation of oblique noise entering a window. Robert Nelson (Graduate Program in Acoust., Penn State, 201 Appl. Sci. Bldg., State College, PA 16802, rwn5136@psu.edu) and Stephen C. Thompson (Graduate Program in Acoust., Penn State, University Park, PA)

Environmental noise propagating through an open window may be cancelled by a sparse array of transducers placed in the window frame. This approach towards global noise cancellation mitigates noise pollution at its source while retaining the light and ventilation received from an open window. Heretofore the array's performance has been studied with a normally incident noise source; however, oblique noise sources must be considered to fully illustrate the array's viability as a commercial product. Analytical and finite element models were used to perdict the array's global noise cancellation capabilities with regards to oblique noise sources, and these results were verified via experimental implementation. Flaws inherent to the current array, including cross bar vibrations and nonoptimal transducer placement, will be discussed along with how they will be circumvented in a future array iteration.

11:30

3aNSa14. Mapping past and future shipping noise in European seas. Roberto Racca (JASCO Appl. Sci., 2305 - 4464 Markham St., Victoria, BC V8Z 7X8, Canada, roberto.racca@jasco.com), Michael A. Ainslie (JASCO Appl. Sci., Den Haag, Netherlands), Johan Bosschers, Marjolein Hermans, Thomas Lloyd (MARIN, Wageningen, the Netherlands), Alexander MacGillivray (JASCO Appl. Sci., Victoria, BC, Canada), Federica Pace (JASCO Appl. Sci., Rotterdam, Netherlands), Max Schuster (DW-ShipConsult, Schwentinental, Germany), Özkan Sertlek (JASCO Appl. Sci., Den Haag, the Netherlands), and Michael Wood (JASCO Appl. Sci., Droxford, United Kingdom)

Against the backdrop of a steadily increasing demand for sea transport of goods and people, the development of a reliable marine shipping soundscape model is an essential planning requirement to assess the effect on ocean noise of operational and technological changes aimed at mitigating the environmental impact of the shipping sector. The NAVISON (Navis Sonus) project, conducted with the support of the European Maritime Safety Agency, employs a specially developed parametric vessel source model with the objective of producing shipping sound maps in European seas for past, present, and potential future conditions over a time span from 2016 to 2050. The source model is combined with historical ship tracking data from the automated identification system (AIS), or projected shipping densities and mitigation scenarios, to calculate spatial ship noise emissions data for input to a sound mapping tool. The mapping tool computes underwater sound propagation using the parabolic-equation method, drawing upon ocean-scale databases of bathymetric, oceanographic, and sediment properties. Project outputs are provided as map layers of sound pressure level and sound energy according to vessel type, season, region, year, and operational conditions; from these layers, maps can be generated for user-specified combinations of mitigation measures. Maps are presented in two frequency bands (centred at 63 Hz and 125 Hz) selected for assessing Good Environmental Status in the context of the European Union's Marine Strategy Framework Directive.

Session 3aPAa

Physical Acoustics, Underwater Acoustics and Acoustical Oceanography: Meteorological Acoustics

Roger M. Waxler, Cochair
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Jelle D. Assink, Cochair
R&D Seismology and Acoustics, KNMI, Utrechtseweg 297, Utrecht, 3731GA, Netherlands

Natalia Sidorovskaia, Cochair
Physics, Univ. of Louisiana at Lafayette, P.O. Box 44210, Lafayette, LA 70504-4210

Contributed Papers

8:00

3aPAa1. Influence of refracting atmospheric profiles on trace velocity and arrival time. Michael J. White (U.S. Army Engineer Res. and Development Ctr., Champaign, IL, michael.j.white@usace.army.mil) and Matthew G. Blevins (U.S. Army Engineer Res. and Development Ctr., Champaign, IL)

The influence of the atmospheric profile on the trace velocity and arrival time has received little attention, although these properties of sound propagation are important for direction finding and localization. The refracting temperature profile causes the received sound to be delayed or advanced in time compared to that for the temperature at ground height, and the situation is made anisotropic in the presence of wind. We consider refraction by linear temperature and logarithmic wind profiles on the trace velocity and arrival time for distant impulses received at the ground, and use this information in small arrays for direction finding and large arrays for localization.

8:15

3aPAa2. Verifying turbulence models in the atmospheric boundary layer by acoustical means. D. Keith Wilson (Cold Regions Res. and Eng. Lab., U.S. Army Engineer Res. and Development Ctr., U.S Army ERDC-CRREL, 72 Lyme Rd., Hanover, NH 03755-1290, D.Keith.Wilson@usace.army.mil), Vladimir E. Ostashev, Carl R. Hart, and Matthew J. Kamrath (Cold Regions Res. and Eng. Lab., U.S. Army Engineer Res. and Development Ctr., Hanover, NH)

The impacts of turbulence on sound propagation in the atmospheric boundary layer (ABL) are important in many practical applications such as acoustic source localization, sonic boom propagation, and auralization of flying aircraft. However, turbulence impacts can vary dramatically in response to changing meteorological conditions. To address this issue, a turbulence spectral model has been introduced that captures the complexities of turbulence generation by wind shear, buoyancy instabilities, and ground blocking in the ABL. An experiment on vertical and slanted sound propagation through the ABL was conducted to verify the model. Various statistical characteristics of sound signals, including variances of the log-amplitude and phase fluctuations, coherences, and signal probability distributions were measured and compared to theoretical predictions. The dependence of the statistics on meteorological conditions was found to be accurately predicted in conditions of well-developed turbulence, as occurs in windy conditions and the daytime. Predictions in conditions of weak and intermittent turbulence, as often occur at night and around sunrise and sunset, remain challenging.

Invited Paper

8:30

3aPAa3. The statistical characteristics of the atmospheric anisotropic inhomogeneities and their effect on the intensity fluctuations of infrasound field. Igor P. Chunchuzov (Atmospheric Dynam., Obukhov Inst. of Atmospheric Phys., 3 Pyzhevsky Per, Moscow 119017, Russian Federation, igor.chunchuzov@gmail.com), Sergey Kulichkov, Oleg Popov, and Vitaly Perepelkin (Atmospheric Dynam., Obukhov Inst. of Atmospheric Phys., Moscow, Russian Federation)

The influence of anisotropic wind velocity and temperature inhomogeneities on the attenuation of infrasound field intensity with increasing distance from a point source and on its altitude distribution is studied. The field is calculated as a function of receiver height and horizontal distance from the source using method of the pseudo-differential parabolic equation for the atmosphere with model realizations of anisotropic effective sound speed fluctuations. These realizations are obtained from the nonlinear shaping model for the gravity wave perturbations which produces the fluctuations with both the vertical and horizontal spectra consistent with the observed spectra. When propagating in the stratospheric and thermospheric wave guides the multiple scattering of infrasound field from the anisotropic fluctuations results in certain vertical wave number spectra and probability density functions of infrasound intensity fluctuations in the stratospheric (altitudes 30–40 km) and mesospheric layers (50–70 km). The statistical characteristics of the intensity fluctuations as a function of distance from the source (up to 2200 km) were studied. The same characteristics were obtained for the infrasound field scattered from the inhomogeneities whose vertical profile was retrieved from the infrasound signals from surface explosions detected in the shadow zone.

Contributed Paper

8:50

3aPAa4. On the possibility of acoustic solitons in open air. Michael S. McBeth (Res. and Appl. Sci., Naval Information Warfare Ctr. Atlantic, NASA Langley Res. Ctr., 22 West Taylor St., M.S. 060, Hampton, VA 23681, michael.s.mcbeth@navy.mil)

Sugimoto *et al*. reported on the generation and propagation of acoustic solitary waves in an air-filled tube with an axially connected periodic array of Helmholtz resonators [Phys. Rev. Lett. **83**, No. 20, 4053–4056 (1999)]. The generation and propagation of acoustic solitons in open air has been discounted due to the nondispersive nature of the speed of sound in air. However, the sound speed is known to be dispersive at certain levels of humidity. Over the ocean there often exists a water vapor density gradient known as an evaporation duct. Here we show that humidity driven frequency dispersion and nonlinearity can combine to allow acoustic solitons in open air. For certain water vapor density gradients found over the ocean there are elevations at which acoustic soliton generationa and propagation are possible. If these open air acoustic solitons can be experimentally verified it could lead to applications including evaporation duct sensing and acoustic communications.

Invited Paper

9:05

3aPAa5. Probing atmospheric waves and the middle atmosphere dynamics using infrasound. Patrick Hupe (Federal Inst. for Geosciences and Natural Resources (BGR), Stilleweg 2, Hannover 30655, Germany, patrick.hupe@bgr.de), Christoph Pilger (Federal Inst. for Geosciences and Natural Resources (BGR), Hannover, Germany), Alexis Le Pichon (CEA, DAM, DIF, Arpajon, France), and Lars Ceranna (Federal Inst. for Geosciences and Natural Resources (BGR), Hannover, Germany)

Infrasound is defined as pressure fluctuations with frequencies between acoustic cut-off (5 min) and human-hearing frequency threshold of sound (16 Hz). Low-frequent infrasonic waves can travel long distances, ranging from hundreds to thousands of kilometres. The middle atmosphere dynamics mainly control the presence of atmospheric waveguides where energy transmission loss is low. These properties are utilized to record atmospheric explosions at highly sensitive pressure sensors (micro-barometers). A global network of 60 infrasound stations was designed as part of the International Monitoring System (IMS) for the Comprehensive Nuclear-Test-Ban Treaty. IMS infrasound stations can record small pressure fluctuations of a few millipascals, which can originate from numerous atmospheric infrasound sources, including meteorological phenomena. In this study, the capability of IMS infrasound arrays for capturing a broad spectrum of atmospheric wave phenomena is highlighted; for instance, mountain-associated infrasonic waves show features that seem to be correlated with orographic gravity waves. Moreover, the interaction of ocean waves produces quasi-continuous infrasound, so-called microbaroms. Infrasonic signatures from microbaroms can be used for probing the middle atmosphere dynamics and assessing atmospheric circulation models. For opening the unique global infrasound network of the IMS for meteorological applications, we also present open-access BGR infrasound data products of two decades and highlight selected case studies.

Contributed Paper

9:25

3aPAa6. Calculating the acoustic and internal gravity wave dispersion relations in Venus's supercritical lower atmosphere. Gil Averbuch (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., 266 Woods Hole Rd., MS #11, Woods Hole, MA 02543-1050, gil.averbuch@whoi.edu) and Andi Petculescu (Dept. of Phys., Univ. of Louisiana at Lafayette, Lafayette, LA)

There is a growing interest in quantifying Venusian seismic events through their infrasonic signatures detected by balloon-borne sensors at $\sim$ 55 km altitude. The extreme pressure and temperature at Venus's surface correspond to supercritical conditions in the planet's deep atmosphere. Therefore, an appropriate real-gas equation of state (EoS) must be used to study the acoustic properties and thermodynamics in the Venusian lower atmosphere. In previous work, the Peng-Robinson (P-R) EoS was used to obtain the acoustic sound speed and attenuation coefficient in the lower atmosphere of Venus. Here, the P-R EoS is coupled with the fluid dynamics equations in order to derive the acoustic and internal gravity wave (IGW) dispersion equations. Results show that in Venus's deep atmosphere, the acoustic cut-off frequency corresponds to a period of $\sim$480 s (0.0020 Hz), and the buoyancy frequency corresponds to a period of $\sim$600 s (0.0016 Hz). By comparison, the values in Earth's lower atmosphere are $\sim$310 and $\sim$340 s, respectively. These differences in acoustic and IGW propagation characteristics will be useful in later efforts to discriminate between the various waves detected by high-altitude sensors.

9:40–10:00 Break

Invited Papers

10:00

3aPAa7. How can acoustic measurements help us understand severe weather phenomena such as lightning? Thomas Farges (DAM, DIF, CEA, Arpajon 91290, France, thomas.farges@cea.fr), François Coulouvrat, and Damien Bestard (Institut Jean Le Rond d'Alembert, Sorbonne Université, Paris, France)

Lightning is an indicator for monitoring severe weather events like heavy precipitation, flash floods, hail… It is also a climate variable for monitoring global warming. Lightning emits light, sound and radioelectromagnetic fields, allowing remote detection and analysis. Acoustic measurements can be used to, for instance, track global warming over more than 70 years via the keraunic level (the

number of thunderstorm days per year in a specific location), or detect and monitor severe weather events such as cyclones, e.g. the Medicanes, or characterize optical phenomena such as sprites associated with violent thunderstorms, or highlight temporal changes in the upper layers of the atmosphere, such as the semi-annual oscillation of stratospheric winds in tropical zones. At closer ranges, less than 30 km, acoustic network measurements complement electromagnetic observations to reconstruct the 3D structure of cloud-to-ground and intra-cloud discharges. Recently, it has been shown possible to provide the 3D structure of acoustic power within the lightning source. From flash to flash, this power shows four orders of magnitude variations, similarly to electromagnetism or optics. Moreover, it outlines that large variations of power also exist even within a lightning flash, reflecting heterogeneities in conductivity within the discharge.

10:20

3aPAa8. Low frequency sounds from tornadoes. Aaron Alexander, Douglas Fox, Real J. KC (Mech. and Aerosp. Eng., Oklahoma State Univ., Stillwater, OK), and Brian R. Elbing (Mech. and Aerosp. Eng., Oklahoma State Univ., OSU-MAE, 201 General Academic Bldg., Stillwater, OK 74078, elbing@okstate.edu)

It has been well established that tornadic storms can generate infrasound (i.e., sound at frequencies below human hearing) signals, but the mechanism of sound generation is still a mystery. There is speculation that these infrasound signals have the potential to serve as an alternative method for alerting and/or tracking of life-threatening tornadoes. Yet, until the mechanism for infrasound generation is understood, it remains a possibility that the infrasound signals are present due to other storm effects and would be improper to use for tornado warning or tracking. One possible explanation for the infrasound signals is an amplification of turbulent fluctuations due to latent heat production. This presentation will detail laboratory efforts to isolate noise production from turbulent structures with and without latent heat production due to condensation of water in saturated warm air. [This work was supported by the Gordon and Betty Moore Foundation, grant DOI 10.37807/gbmf11559.]

Contributed Paper

10:40

3aPAa9. A theory for the emission of infrasound from Tornadoes. Bin Liang (Univ. of MS, Oxford, MS), Roger M. Waxler (Univ. of MS, P.O. Box 1848, University, MS 38677, rwax@olemiss.edu), and Paul Markowski (The Penn State Univ., University Park, PA)

Tornadoes have been shown to radiate infrasound to great distances, however convincing fundamental sound mechanisms are still absent. After using vortex sound theory to study sound generated by two numerical tornadoes, we found that there is a significant low-frequency signal between 0.1 Hz and 1 Hz. The sound is closely related to rotation of the non-axisymmetric vorticity field and its frequency depends on the rotational frequency. The non-axisymmetric vorticity field is represented by a Kirchhoff vortex-like flow in baseline tornado model and by multiple-vortex flow in eddy injection tornado model. Interestingly, there also exist high-frequency components in the later model which are hypothesized to originate from vortex merging process. Field detection data of tornado infrasound provides some support for the low-frequency sound.

Invited Paper

10:55

3aPAa10. A study of acoustic array signal processing applied to tornado tracking. Garth Frazier (NCPA, Univ. of MS, NCPA, University of MS, P.O. Box 1848, Oxford, MS 38677, frazier@olemiss.edu) and Bin Liang (National Ctr. for Physical Acoust., Oxford, MS)

In low signal-to-noise ratio (SNR) applications it is often necessary to integrate array measurement data over an extended temporal period to achieve satisfactory estimates of source bearings-of-arrival (BOA), especially in the case of multiple sources. For example, in popular frequency-domain methods such as MUSIC and its variants, the number of so-called snapshots (temporal windows) can be increased to improve performance. However, this improvement is achieved under the assumption that the bearing-of-arrival is not changing during the integration period. In this presentation, a study of the estimation performance of two algorithms (a MUSIC variant and a maximum-likelihood based method) is performed as a function of the number of snapshots, and the result of application of these algorithms to infrasound data recorded during the presence of tornadic storms (moving sources) is presented. Additionally, a method for integration over time using a moving source model is introduced.

Contributed Paper

11:15

3aPAa11. Discussion of infrasonic detection of tornadoes in the Southeastern United States. Roger M. Waxler (NCPA, Univ. of MS, P.O. Box 1848, University, MS 38677, rwax@olemiss.edu), Garth Frazier, Carrick Talmadge (NCPA, Univ. of MS, Oxford, MS), Claus Hetzer (NCPA, Univ. of MS, Tempe, AZ), Hank Buchanan, Bin Liang (NCPA, Univ. of MS, Oxford, MS), Naveen Thirunilath (NCPA, Univ. of MS, University, MS), Chip Audette (Benchtop Eng. LLC, Springfield, VT), and Islam Hamama (NCPA, Univ. of MS, University, MS)

It has been established that tornadoes emit an infrasonic signal that is regularly detected in the 1 to 10 Hz band, although the actual band is expected to be wider. The physical mechanism through which this signal is generated is not yet fully understood and is the subject of current research. Here we discuss some of our experimental efforts, past and present. During the tornado seasons of 2017, 2018, and 2019, we deployed a network of infrasound sensor arrays in northern Alabama, southern Tennessee, and northwestern Georgia. We performed detailed analysis of one particular storm front that spawned at least eight identified tornadoes in northwestern Alabama during its passage. For that storm we find that when propagation modeling and wind noise analyses suggest that the signal should have been detected at a given array it always was. A long term monitoring effort is now underway in Mississippi. The Mississippi network deployment strategy relied on statistical analyses of tornado touchdown probability, signal transmission loss, and local wind noise studies. The network and array design for our Mississippi network was heavily informed by our experiences in Alabama and, we believe, represents the current state-of-the-art. In this presentation we will give an overview of what was learned from our previous deployments in Alabama, what strategies were used for the current deployment in Mississippi, and the current state of our data collection and analysis.

WEDNESDAY MORNING, 15 MAY 2024 ROOM 202, 9:45 A.M. TO 12:00 NOON

Session 3aPAb

Physical Acoustics, Engineering Acoustics, Structural Acoustics and Vibration, and Signal Processing in Acoustics: Characterization of Electronic Materials, Components, Devices, and Batteries

Michael R. Haberman, Cochair
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Bogdan-Ioan Popa, Cochair
Univ. of Michigan, 2350 Hayward St, Ann Arbor, MI 48109

Haley N. Jones, Cochair
Materials Science and Engineering, Penn State University, N-225 Millenium Science Complex, State College, PA 16802

Lauren Katch, Cochair
The Pennsylvania State University, 212 Earth and Engineering Sciences Building, State College, PA 16802

Chair's Introduction—9:45

Invited Papers

9:50

3aPAb1. Air-coupled ultrasound for testing of batteries and components. Tomas Gomez Alvarez-Arenas (ITEFI, CSIC, Serrano 144, Madrid 28006, Spain, t.gomez@csic.es)

This work describes the use of an air-coupled and through transmission ultrasonic (0.2–1.5 MHz) technique, both at normal and oblique incidence, for the characterization and test of pouch-cell Li-ion (and similar types) batteries and components. Time domain measurements show that it is possible to measure ultrasonic velocity in the battery. Frequency domain measurements reveal the generation of thickness resonances. From them, solution of the inverse problem permit to extract: ultrasound velocity and attenuation (and variation with frequency), thickness and density. At normal incidence, these resonances provide information about the compressional wave.

At oblique incidence, it is first observed that there is a range of incidence angles where propagation is no permitted. This response can be explained in terms of the battery layered structure, once these angles are exceeded, the shear wave is generated, propagated and observed, giving rise to the generation of thickness resonances. Shear wave propagation is strongly anisotropic: lower velocity in the direction of the battery plane, compared with a higher velocity in the direction normal to the battery plane. No limit angle for the shear wave is observed. Possibilities of this technique for the testing of new batteries, continuous state monitoring and battery sorting for re-use or recycling are discussed.

10:10

3aPAb2. Cryogenic acoustic microscopy and evaulation of electronic components. Steven Doran (Phys. and Astronomy, Iowa State Univ., 2323 Osborn Dr., Ames, IA 50011-1026, doran@iastate.edu), Leonard J. Bond (Aerosp. Eng., Iowa State Univ., Ames, IA), Daniel Barnard (Ctr. of Nondestructive Evaulation, Iowa State Univ., Ames, IA), Navaneeth Poonthottathil (Phys., IIT Kanpur, Kanpur, India), Amanda Weinstein, Yue Feng, Sijith Edayath, Hariom Sogarwal, and Frank Krennrich (Phys. and Astronomy, Iowa State Univ., Ames, IA)

Electronics operating at cryogenic temperatures are essential in fields like space exploration and particle physics. The Deep Underground Neutrino Experiment (DUNE), a next generation particle physics experiment, will rely on tens of thousands of custom designed application specific integrated circuits (ASICs) operating directly in liquid argon (87 K) for decades without repair or replacement. Ensuring circuit functional reliability throughout the duration of the experiment is mission critical. Both functional and nondestructive testing are employed to safeguard circuit quality and reliability. Part of this work involves the design, testing, and data analysis of a cryogenic acoustic microscope (CryoSAM) operating at 15 MHz. The CryoSAM is capable of interrogating ASICs at both room temperature (300 K) and in liquid nitrogen (77 K), identifying acoustic anomalies likely arising from thermal stress and manufacturing-related defects, using a powerful correlation analysis technique. These anomalies can lead to functional degradation and suboptimal electronic performance of the circuit sensors. Image analysis and correlations are used to compare differences seen before and after cryogenic temperature cycling. Data are reported that were collected at both room temperature and when cooled using liquid nitrogen. Designs and challenges of operating the instrument at cryogenic temperatures are also discussed.

Contributed Papers

10:30

3aPAb3. Measurements of nonlinear electric-acoustic interactions in lead zirconium titanate. Robert Lirette (Communications Technol. Lab., National Inst. of Standards and Technol., 325 Broadway, MS67201, Boulder, CO 80305, robert.lirette@nist.gov), Tomasz Karpisz (Communications Technol. Lab., National Inst. of Standards and Technol., Boulder, CO), Małgorzata Musiał (Mater. Measurement Lab., National Inst. of Standards and Technol., Boulder, CO), Gabriela Petculescu (Univ. of Louisiana at Lafayette, Lafayette, LA), Aaron Hagerstrom, Nathan Orloff, and Angela C. Stelson (Communications Technol. Lab., National Inst. of Standards and Technol., Boulder, CO)

When electric and acoustic waves simultaneously pass through a piezoelectric material, nonlinearities can arise. The electric and acoustic waves can mix, generating a signal consisting of both sum and difference spectral components. Here we present a method of directly measuring the nonlinearly mixed electric-acoustic signal using a vector network analyzer (VNA). In normal VNA operation, sources and receivers operate at the same frequency. In our measurements, we employ a frequency offset mode, where receivers in the VNA can be tuned to a frequency different from the source frequency. Here, we measure at the expected mixing product frequency of the electrical and acoustic signals we introduce into the measurement to capture nonlinear electric-acoustic mixing. We present measurements of electric-acoustic mixing in a block of lead zirconium titanate (PZT) mounted on top of an electric co-planar waveguide connected to the VNA. The results show the presence of many coupled electric-acoustic modes. To interpret these modes, we characterized the PZT block with modal analysis and finite element simulations. Some of the observed nonlinear electric-acoustic modes directly correlated with the mechanical modes of the block, while others did not. Overall, this method allows for direct probing of non-linear electric-acoustic mixing in materials and devices.

10:45

3aPAb4. Micro-electromechanical systems sensor characterization for acoustic prognostic health management. Ruth Willet (Penn State, 201 Old Main, University Park, PA 16802, raw5930@psu.edu) and Karl Reichard (Penn State, State College, PA)

Prognostic Health Management (PHM) is the process used to optimize machinery use by detecting faults and predicting failures and a machine's remaining useful life. PHM is especially important to electro-mechanical systems in order to maximize a system's availability and effective operation. The use of Micro-Electromechanical Systems (MEMS) accelerometers and microphones for PHM analysis is beneficial because it enables nondestructive acoustic testing with reduced cost, reduced size, and more efficient vehicle integration with other electronics than current methods. This research quantitatively studies a variety of MEMS sensors to understand how they measure acoustic data in a lab setting and on a truck with a diesel engine compared to piezoelectric accelerometers and condenser microphones. Diesel engines are common in many vehicles and hybrid-electric power systems, so the results will be applicable to a variety of situations. By applying signal processing techniques and PHM condition indicators to the data, standardized metrics can be developed to determine the best sensor for different PHM analyses.

11:00

3aPAb5. Degradation monitoring of multilayer ceramic capacitors during high accelerated lifetime testing using ultrasound. Haley N. Jones (Mater. Sci. and Eng., Penn State Univ., N-225 Millenium Sci. Complex, State College, PA 16802, hnj5051@psu.edu), Andrea P. Arguelles (Eng. Sci. and Mech., Penn State Univ., State College, PA), and Susan Trolier-McKinstry (Mater. Sci. and Eng., Penn State Univ., University Park, PA)

Multilayer ceramic capacitors (MLCCs) are vital circuitry components where long-term stability at high temperature and fields is critical to device operation. High accelerated lifetime testing (HALT) statistically investigates reliability and lifetime of MLCCs through exposure to temperatures and voltages exceeding normal operating conditions. The long-term reliability of the dielectric layers in MLCCs strongly depends on potential variability in the dielectric microstructure, such as cracking, which can be challenging to detect. Nondestructive ultrasonic evaluation is sensitive to microstructural changes through measurements of wave scattering. This work uses high frequency (100 MHz) focused ultrasonic scattering methods to nondestructively monitor structural and microstructural changes in MLCCs from the pristine to the failed state during HALT to understand structural origins of electrical failure. Printed circuit board mounted MLCCs were electrically and ultrasonically characterized in a pristine state and during various degraded states before and after electrical failure under HALT. Evidence of damage pre-HALT was ultrasonically indicated by high attenuation regions on the ultrasonic maps suggesting that the mounting process

may induce damage. The high attenuation regions in the pristine samples evolved throughout HALT indicating that the damage in the pristine state might be the spatial origin of part failure, which underscores the impact and importance of detection of structural defects on the reliability of MLCCs.

11:15

3aPAb6. Ultrasonic C-scan for micro-crack inspection on semi-flexible solar modules. Dicky Silitonga (George W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., Metz, France), Pooja Dubey (George W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., 2 Rue Marconi, Georgia Tech-Europe, Metz 57070, France, pooja.dubey@gatech.edu), and Nico Declercq (George W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., Metz, France)

Solar photovoltaic modules are versatile power sources in diverse materials and configurations, including compact and flexible variants for portable electronic devices. Ensuring the reliability of these modules is crucial for sustaining the functionality of the devices they power. Manufacturing or handling-induced defects, such as cracks or scratches, pose a threat to the performance of solar modules. Hence, non-destructive inspection becomes essential in the quality control process. Ultrasonic C-scan has been an established inspection technique within various industries; however, its application on solar modules remains uncommon. On the other hand, Scanning Acoustic Microscopy (SAM) has been implemented for observing defects in solar cells, yet employing SAM for comprehensive module scanning is inefficient. This study aims to assess the capability of ultrasonic c-scan in detecting micro-cracks within semi-flexible solar panels and to evaluate the effects of frequency selection on the results. This work investigates the specimen with an Ultrasonic C-Scanner at different frequencies. Subsequently, the outcomes are validated by comparing them with the results from the SAM. The potential of using a widely known ultrasonic technique for this purpose, while understanding its limitations, such as the c-scan, will enable a more straightforward integration of the technique into the solar module quality control process.

11:30

3aPAb7. Probing layer interface behavior in Lithium-ion batteries via concurrent ultrasonic and modal measurements. Alexandra Litvinov (Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, 204 E. Dean Keeton St., Texas, TX, alitvinov@utexas.edu), Tyler McGee, Ofodike A. Ezekoye, and Michael R. Haberman (Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX)

Lithium-ion batteries are pivotal in various technological applications, from powering electric vehicles to supporting renewable energy storage systems. Understanding and monitoring the intricate chemo-mechanics within lithium-ion cells is imperative for ensuring their reliability and performance over time. Previous research has shown that both ultrasonic and vibrational measurements provide a measure of a cell's state of charge (SOC) and state of health (SOH) and provide indications of existing or previous thermal or electrical abuse. Recent work suggests that ultrasonic and modal analysis may provide complementary insights into the evolution of layer interfaces during early-life aging due to charge-discharge cycling [J. Acoust. Soc. Am., **154**, A284 (2023)]. This work presents the results of concurrent ultrasonic and modal measurements over multiple charge-discharge cycles. Namely, we monitor concurrent changes in the ultrasonic time-of-flight and signal amplitude and the resonance frequency of a clamped-clamped 10 Ah Nickel–Manganese–Cobalt pouch cell as a function of electrical cycling. The evolution of these metrics paired with analytical and numerical models of the cell will be used to understand changes in the material properties at layer interfaces and potential structural alterations within the battery which may be important indicators of SOC and early-life aging mechanisms.

11:45

3aPAb8. Assessing spatial non-uniformities in lithium-ion battery state of charge using ultrasound immersion testing. Mac Geoffrey Ajaereh (Mech. Eng., Univ. of Bath, Claverton Down, Bath, Bath BA2 7AY, United Kingdom, mgoa20@bath.ac.uk), Olivia J. Cook (Penn State, University Park, PA), Haley N. Jones (Mater. Sci. and Eng., Penn State Univ., State College, PA), Nathan Kizer (Penn State Univ., University Park, PA), Lauren Katch (The Penn State Univ., State College, PA), Christopher Wheatley (Penn State, State College, PA), Chris Vagg, Charles Courtney (Mech. Eng., Univ. of Bath, Bath, United Kingdom), Christopher M. Kube (Eng. Sci. and Mech., The Penn State Univ., University Park, PA), and Andrea P. Arguelles (Eng. Sci. and Mech., Penn State Univ., State College, PA)

Enhancing the performance, safety and reliability of battery management systems is crucial for advancing the state of the art in battery electric vehicles. Current research explores the potential of ultrasound to monitor state of charge (SoC) changes in individual cells. Understanding spatial variations in SoC is essential, as non-uniformities could lead to sub-optimal performance, premature ageing, and possible safety risks. This study uses ultrasound immersion C-scans to map wave speed and attenuation at different SoC levels during battery cycling. Results indicate non-uniform wave speed and attenuation suggestive of SoC spatial variations within single cells, emphasising the importance of addressing this issue. Acoustic measurements under various C-rates and relaxation periods are discussed, providing insights into lithium-ion rearrangement in graphite particles. Potential causes of structure and manufacturing variations of the cell are discussed, highlighting the need to address these issues to prevent overcharging or overdischarging in specific battery areas.

Session 3aPP

Psychological and Physiological Acoustics: Temporal Processing, Aging, and Hearing Loss: Research Inspired by Christian Fullgrabe

G. Christopher Stecker, Chair

Center for Hearing Research, Boys Town National Research Hospital, 555 N 30th St., Omaha, NE 68131

Chair's Introduction—8:00

Contributed Papers

8:05

3aPP1. Modeling the relationship between listener factors and signal modification: A pooled analysis spanning a decade. Varsha H. Rallapalli (Commun. Sci. & Disord., Northwestern Univ., 2240 Campus Dr., Evanston, IL 60208, varsha.rallapalli@northwestern.edu), Jeff Crukley (Univ. of Toronto, Toronto, ON, Canada), Emily Lundberg, James M. Kates, Kathryn Arehart (Speech Lang. and Hearing Sci., Univ. of Colorado Boulder, Boulder, CO), and Pamela E. Souza (Northwestern Univ., Evanston, IL)

The overarching goal of our work is to understand how listeners respond to hearing aid processing and consequently inform individualized hearing aid treatment. Our previous work demonstrated that individual cognitive abilities, age, and hearing loss contribute to variability in response to cumulative signal modifications introduced by hearing aid processing and background noise. Specifically, older individuals with poorer working memory and more hearing loss were more susceptible to signal modifications introduced by hearing aid processing. These relationships were established in independent studies involving systematic manipulations of compression, digital noise reduction, frequency lowering, or microphone directionality. In this study, we present a hierarchical pooled analysis of data collected from six previous studies to develop a unified statistical model of the relationships between response to signal modification and individual listener variables. Across studies, signal modification is quantified using a cepstral correlation metric that accounts for cumulative envelope distortions arising from hearing aid processing and background noise. The statistical model will determine how working memory, age, and degree of hearing loss mediate the relationship between signal modification and speech intelligibility in noise across a large dataset. Both inferential and predictive applications of the combined data and model will be discussed. [Work supported by NIDCD.]

8:20

3aPP2. Physiological characterization of a macaque model of presbycusis. Swarat S. Kulkarni (Hearing and Speech Sci., Vanderbilt Univ., 111 21st Ave. S, Nashville, TN 37240, swarat.s.kulkarni@vanderbilt.edu), Amy N. Stahl, Troy Hackett, and Ramnarayan Ramachandran (Hearing and Speech Sci., Vanderbilt Univ., Nashville, TN)

Age-related hearing loss (presbycusis) is the leading cause of hearing loss worldwide, but current human studies do not separate this pathology from lifetime noise exposure. Laboratory macaques live in environments with minimal noise-induced hearing trauma and have long lifespans, making them optimal presbycusis models. We describe age-related auditory changes using two clinical, non-invasive physiological measures in aging macaques: distortion product otoacoustic emissions (DPOAEs) and auditory brainstem responses (ABRs). ABRs and DPOAEs were measured in anesthetized young (6–9 years old, $n = 36$ ears) and aging (26–35 years old, $n = 20$ ears) *Macaca mulatta*. ABR and DPOAE thresholds were elevated in aging subjects compared to young macaques at all tested frequencies. ABR thresholds were significantly correlated with age for frequencies >8 kHz, but DPOAE thresholds were not, a pattern likely attributable to a high-frequency ceiling effect for DPOAE measures. For frequencies <8 kHz, DPOAE thresholds showed significant correlation with age. However, this was not observed with ABR thresholds, suggesting differential sensitivity to age-related changes for auditory nerve and brainstem function compared to cochlear outer hair cell function. Ongoing analysis on temporal processing/adaptation metrics will further elucidate age-related changes. Future work will correlate these findings with histological analysis. [Work Supported by NIH R01-DC-015988.]

8:35

3aPP3. Perceptual benefits of implicit auditory learning of rule-based nonspeech sound patterns in younger and older adults. Brian Gygi (Nottingham Hearing Biomedical Res. Unit, Martinez, CA), Christian Füllgrabe (MRC Inst. of Hearing Res., Chicago, IL), and Valeriy Shafiro (Commun. Disord. and Sci., Rush Univ. Medical Ctr., 600 South Paulina St., 1015 AAC, Chiago, IL 60612, Valeriy_Shafiro@rush.edu)

Sounds in everyday environments typically precede and follow one another in systematic ways determined by the properties and interactions of corresponding sound sources. Familiar and predictable contextual relationships between consecutive sounds are known to provide perceptual benefit under adverse listening conditions. Previous work has demonstrated implicit learning of arbitrary rule-based sound patterns by young normal hearing adults. However, the role of such rule-based learning on identification of individual sounds within a sound pattern and the effect of listener age has not been investigated. Here younger and older adults with age-appropriate hearing were first familiarized with a set of rule-based nonspeech sound sequences produced by a finite-state machine. Subsequently, they identified individual target sounds within sequences under a range of signal-to-noise ratios (0 to -15 dB). The test sequences were (a) familiar, rule-based; (b) novel, rule-based; or (c) novel, non rule-based. The results for both younger and older listeners revealed superior sound identification in sequences versus in isolation, and a greater overall accuracy of rule-based versus non rule-based sequences. Younger listeners significantly outperformed older listeners. Implicit learning benefit did not correlate with measures of working memory processing. Current findings have implications for audiologic rehabilitation and auditory display design.

8:50

3aPP4. Age-related changes in gap detection thresholds in adult cochlear-implant users: Effects of stimulus level and electrode-to-neural interface. Miranda Cleary (Hearing and Speech Sci., Univ. of Maryland-College Park, 7251 Preinkert Dr., Ste. 0100 Lefrak Hall, College Park, MD 20742, micleary@umd.edu) and Matthew J. Goupell (Hearing and Speech Sci., Univ. of Maryland-College Park, College Park, MD)

Aging is associated with reduced sensitivity to brief temporal gaps in auditory stimuli. Although gap detection is a relative strength among late-deafened cochlear-implant users, aging and electrode-to-neural interface (ENI)

factors may contribute to temporal processing difficulties. Because aging may negatively impact perception particularly at very low and very high input levels, we hypothesized larger gap detection thresholds (GDTs) at poor-ENI electrodes and an interaction between age and level such that GDT improvement with increased level would be less evident in older listeners. GDTs were measured on three electrodes selected to span a range of good-to-poor ENI based on electrically-evoked compound action potential amplitude growth or threshold sensitivity to increased pulse rate. GDTs were assessed using constant-amplitude single-electrode 1000-pps pulse trains presented at 40/60/80/100% dynamic range. In preliminary data (N = 4), GDTs averaged 3 ms at 100%DR and 40 ms at 40%DR. Listeners with better average ENI metrics displayed better GDTs. Within-subject, an advantage for best-ENI electrode over poorer-ENI electrodes was evident only at low levels. Finding an interaction between age and level such that temporal processing is impaired at both low and high (but not mid) levels for older listeners may help elucidate the contributions of peripheral versus central factors.

9:05

3aPP5. Effects of auditory processing ability on the phonological and acoustic processing of Mandarin tone by Thai speakers: An ERP study. Yu Zou (English Dept. & Lang. and Cognit. Neurosci. Lab, School of Foreign Studies, Xi'an Jiao Univ., No. 28 Xianning Rd. (W), Xi'an, Shaanxi 710049, China, zouyu_02@stu.xjtu.edu.cn), Bing Cheng (English Dept. & Lang. and Cognit. Neurosci. Lab, School of Foreign Studies, Xi'an Jiao Univ., X'an, China), and Yang Zhang (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN)

This study investigated how Mandarin-naïve Thai speakers' auditory processing ability affects their phonological and acoustic processing of Mandarin tone. The participants were 20 native Mandarin speakers and 19 Mandarin-naïve Thai speakers. A 10-step computer synthesized Tone 1 (T1)–Tone 2 (T2) continuum was manipulated for identification task. The stimulus 1, 3, 5, 7, 9 from the T1–T2 continuum were used in a multi-feature passive oddball ERP paradigm, encompassing within-category and across-category stimuli with equivalent large-scale and small-scale acoustic intervals. We used pitch discrimination, melody reproduction, and rhythm reproduction to assess participants' auditory processing ability. The identification results showed that Mandarin-naïve Thai speakers' exhibit a weaker categorical perception (CP) of Mandarin tones compared to the native Mandarin speakers, a pattern consistent with ERP results observed at both sensor and source levels. Further analysis revealed higher auditory processing ability leads to enhanced CP of Mandarin tone for learners from Thai, and the same improvement is observed in the melody memory for acoustic processing of Mandarin tone. These findings add to our understanding on how individual differences influence phonological and acoustic processing of Mandarin tone during the initial stage of second speech learning.

9:20

3aPP6. Estimating the high-frequency extent of binaural sensitivity to temporal fine structure across listeners. Matthew J. Goupell (Hearing and Speech Sci., Univ. of Maryland-College Park, College Park, MD), Anhelina Bilokon (Dept. of Hearing and Speech Sci., Univ. of Maryland, College Park, MD), Brittany T. Williams (Boys Town National Res. Hospital, Omaha, NE), Daniel J. Tollin (Physiol., Univ. of Colorado School of Medicine, Aurora, CO), and G. Christopher Stecker (Ctr. for Hearing Res., Boys Town National Res. Hospital, 555 N 30th St., Omaha, NE 68131, cstecker@spatialhearing.org)

Füllgrabe *et al*. [2017, Int. J. Audiol. 56:926-35] developed a novel procedure to estimate the frequency dependence of listeners' sensitivity to temporal fine structure in the form of interaural phase differences (IPD). Whereas traditional approaches fixed tone frequency and varied IPD, the new "TFS-AF" test fixed the IPD and varied frequencies adaptively so as to estimate the upper frequency limit (UFL) of IPD sensitivity, even in listeners who struggle with traditional tests. UFL estimates for young normal hearing listeners were consistently 1200–1400 Hz, with notable exceptions of lower UFL (e.g. 800–1000 Hz) in a small subset. The consistency of UFL and abrupt loss of IPD sensitivity above it remain among the most puzzling aspects of binaural hearing, unexplained by the (rather shallow) frequency dependence of audibility, neuronal phase locking, or internal noise in this region. Recently, our group adapted a version of the TFS-AF test to measure UFL as a function of sound level, comparing the resulting slope to predictions of candidate mechanisms. The limitation closely matches the upper edge of an auditory filter centered at ~700 Hz, suggesting that IPD sensitivity is mediated exclusively by neurons tuned to that frequency or below. [Work supported by NIH R01DC014948, R01DC016643, and R01DC017924.]

9:35–9:50 Break

9:50

3aPP7. Behavioral evidence of the modulation filterbank in an avian animal model of complex-sound perception. Kenneth S. Henry (Otolaryngol., Univ. of Rochester, 601 Elmwood Ave., Box 629, Rochester, NY 14642, kenneth_henry@urmc.rochester.edu), Kristina Abrams (Neurosci., Univ. of Rochester, Rochester, NY), and Margaret Youngman (Otolaryngol., Univ. of Rochester, Rochester, NY)

Behavioral detection of amplitude modulation (AM) can be adversely affected by competing AM "noise" present in the acoustic environment. Called AM masking, human studies suggest the existence of a modulation filterbank that separates sounds (signal from noise) based on differences in AM frequency. The modulation filterbank is an important theoretical advancement in hearing science because in addition to explaining AM masking, the model successfully predicts differences in speech perception across noisy listening environments with different AM statistics. Currently, there are no behavioral animal models of the modulation filterbank, presenting a serious impediment to understanding neural underpinnings. We trained budgerigars, a parakeet species, to detect AM frequencies from 64 to 400 Hz in the presence of narrowband AM maskers applied to a 2.8-kHz carrier signal. AM masker center frequencies spanned a >2 octave range centered on the target AM frequency. Behavioral AM sensitivity was evaluated with operant conditioning, a single-interval two-alternative discrimination task, and two-down one-up adaptive threshold tracking procedures. Budgerigar AM masking functions had a band-pass characteristic, consistent with the modulation filterbank and with human AM masking results for the same stimuli. To our knowledge, these are the first behaviorally estimated modulation filter shapes in a nonhuman species. [Funding: NIDCD R01-DC017519.]

10:05

3aPP8. Low gain hearing aids use and benefit: A viable management option for those with hearing sensitivity within the normal range. Alyssa Davidson (Walter Reed National Military Medical Ctr., 4954 North Palmer Rd., Bldg 19, Rm 5506, Bethesda, MD 20889-5630, alyssa.j.davidson.civ@health.mil), Gregory M. Ellis (Audiol. and Speech Pathol., Walter Reed National Medical Military Ctr., Bethesda, MD), and Douglas Brungart (Walter Reed National Military Medical Ctr., Bethesda, MD)

Low (or mild) gain hearing aids (LGHAs) are increasingly considered for individuals with normal peripheral hearing but self-reported auditory complaints. The Tinnitus and Hearing Survey-Hearing Subscale (THS-H) offers a normative cutoff, aiding identification of significant self-reported hearing difficulties (SHD). This research assesses the benefits of LGHAs as a management option for individuals with normal hearing sensitivity and significant SHD, comparing LGHA use and benefit to individuals without SHD and those with peripheral hearing loss. 186 participants across four groups, including those with or without SHD and peripheral hearing loss were recruited. Participants completed questionnaires that addressed hearing aid usage, benefit, SHD and tinnitus. Individuals with significant SHD and hearing sensitivity within the normal range (NHT) reported higher LGHA usage and benefit than individuals with normal hearing difficulties (NHD) and NHT. Comparable use and benefit were noted between groups with significant SHD regardless of peripheral hearing loss status. The findings support LGHAs as a suitable management option for individuals with NHT and SHD, as indicated by hearing aid use and benefit. Quantifying the level of perceived auditory processing deficits (i.e., SHD), notably with the THS-H, enhances sensitivity in identifying those who may benefit from this management option.

Session 3aSA

Structural Acoustics and Vibration, Education in Acoustics and Physical Acoustics: My Favorite Homework Problems in Structural Acoustics in Vibrations

Samuel P. Wallen, Cochair
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Andrew Barnard, Cochair
Acoustics, Penn State, 201C Applied Sciences Building, University Park, PA 16802

Kathryn Matlack, Cochair
University of Illinois at Urbana-Champaign, 1206 W Green St., Urbana, IL 61801

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

Contributed Papers

10:15

3aSA1. Fundamental acoustics concepts taught through the development of an impedance tube. Nathan P. Geib (Appl. Res. Labs., The Univ. of Texas at Austin, 1587 Beal Ave Apt 13, Ann Arbor, MI 48105, geib@umich.edu), Allan Pham, and Christina Naify (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

As part of its mission to educate students, Applied Research Laboratories, The University of Texas at Austin (ARL:UT) employs student technicians to assist on a variety of ongoing research projects. Recently, an undergraduate student technician was tasked with the design and fabrication of a low-cost impedance tube for use in characterizing acoustic materials. To assist the student in developing a set of design requirements, and to determine whether preliminary data were being accurately collected and processed, a set of homework problems were developed to teach the student the fundamental concepts in acoustics necessary for impedance tube development. Homework problems required utilizing both travelling- and standing-wave solutions to the one-dimensional (1D) Helmholtz equation, as well as a basic understanding of the relationships between pressure, particle velocity, and acoustic impedance. Homework solutions were compared with 3D finite element simulations and with experimental data collected using the impedance tube. We felt that these homework problems were effective at conveying basic ideas in acoustics to a student with no prior knowledge in the subject, and how those concepts could be used to solve real-world problems.

10:30

3aSA2. Don't say "modal analysis:" A backdoor introduction to vibrations via programming and numerical methods. Samuel P. Wallen (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, sam.wallen@utexas.edu)

As an undergraduate degree requirement, The Walker Department of Mechanical Engineering at the University of Texas at Austin teaches a lower-division, introductory course on computer programming and numerical methods. The enrollment consists primarily of second-year mechanical engineering (ME) students, approximately half of whom enter the course with no prior programming experience. Due to its relatively early position in the ME curriculum, the course presents the unique challenge of teaching students to apply numerical methods to ME-relevant problems, with little background about the physics involved or the contexts in which they are likely to occur in practice. This talk presents an end-of-term project in which students use two numerical methods discussed earlier in the course, singular value decomposition and Runge-Kutta methods, to construct, simulate, and benchmark a reduced-order, dynamic model of a vibrating guitar string. In addition to reinforcing course outcomes associated with programming, mathematics, and data visualization, this project provides an introduction to acoustics, vibration, and modal analysis at a level that belies its position in the overall curriculum.

10:45

3aSA3. Evaluating years in hours. Robert W. Smith (PSU/ARL, P.O. Box 30, State College, PA 16804, rws100@arl.psu.edu)

The Graduate Program in Acoustics (GPA) at Penn State requires prospective Ph.D. candidates to pass a qualifying exam consisting of written and oral components. Objectives are to evaluate mastery and integration of some key concepts, and to determine if students have developed a fluency with acoustic concepts. The over-riding goal is to evaluate if the students have emergent qualities and understanding to potentiate success as researcher. Most of the exam is focused on the content of two required graduate courses—one primarily on vibrations in solids, and the second in fluids; these provide a common foundation to supplement the diverse undergraduate backgrounds of students, characteristic of students attracted to attend the GPA. We strive to avoid esoterica, but problems are often intentionally unfamiliar and often inspired by applications encountered in research. The author has participated as one of four on the committee that administers the exam for the past 7 years; each member is charged with developing two problems for the exam, typically offered twice per year. This talk will describe some illustrative problems developed by the author and some of the acoustic concepts they probe we think are important. Some discussion of observations of students' difficulties will be discussed.

11:00

3aSA4. Lord Rayleigh versus Chladni and KFC Sanders: Who is correct about tuning forks? Daniel A. Russell (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg, University Park, PA 16802, dar119@psu.edu)

The tuning fork is one of my favorite pieces of acoustical apparatus (as evidenced by three published papers in the *American Journal of Physics* and an article in *Acoustics Today)*. A favorite problem (formerly used on final exams, and currently used as a hands-on activity) for the first-year graduate

3a WED. AM

structural vibrations course I teach, requires students to read excerpts from four textbooks which make a statement about the boundary conditions for the tines of a tuning fork. Lord Rayleigh and Barton describe a tuning fork as two fixed-free bars joined at the base, while Chladni and Kinsler, Frey, Coppens, and Sanders describe a tuning fork as free-free bar bent into a U-shape. Who is correct? Students are required to measure the frequency spectrum of a tuning fork and match the resulting frequency ratios to what they would obtain by solving the boundary condition problem by treating the fork tines as a free-free or fixed-free bar undergoing flexural vibrations. This talk will work through the problem and will illustrate some of the pitfalls that often trip up students. And yes, the question of who is correct will be answered!

11:15

3aSA5. Applying intuition to Fourier analysis by rendering images with vibration. Tre DiPassio (Elec. and Comput. Eng., Univ. of Rochester, 402 Comput. Studies Bldg., 160 Trustee Rd., Rochester, NY 14627, tredipassio@rochester.edu)

It has been said that "Everything in life is vibration." While this may seem dramatic, it is true that we can describe any shape as a superposition of a structure's bending modes using Fourier decomposition. In this session, I present an assignment that involves the rendering of images and artwork using only a subset of a rectangular membrane's mode shapes. This assignment allows students to engage with fundamental concepts such as Fourier analysis and the modal nature of structural vibrations in a way that allows them to see and interact with the results of their work. A natural extension is to simulate the response of a plate to an external stimulus using the Fourier series, and to verify the results of their simulation experimentally using tools such as laser vibrometry. While exciting complex patterns in the vibrational response of a real plate is difficult in practice, this assignment provides a memorable and engaging baseline for understanding the modal composition of induced structural vibrations.

WEDNESDAY MORNING, 15 MAY 2024 ROOM 203, 8:00 A.M. TO 11:15 A.M.

Session 3aSC

Speech Communication, Architectural Acoustics, Psychological and Physiological Acoustics, and Education in Acoustics: Classroom Acoustics and Speech Communication

Pasquale Bottalico, Cochair
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Reanne Pernitsky, Cochair
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David S. Woolworth, Cochair
Roland, Woolworth & Associates, 356 CR 102, Oxford, MS 38655

Contributed Papers

8:00

3aSC1. Understanding speech in everyday conditions—Effects of room acoustics and noise. Heui Young Park (Graduate Program in Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, hkp5188@psu.edu), Navin Viswanathan (Dept. of Commun. Sci. and Disord., The Penn State Univ., University Park, PA), and Michelle C. Vigeant (Graduate Program in Acoust., The Penn State Univ., University Park, PA)

Understanding speech in noisy, indoor settings is a considerable challenge for listeners. Room acoustics, such as reverberation and heating, ventilation, and air conditioning (HVAC) noise, all contribute to such challenges. In the current study, the effects of different backgrounds were examined in anechoic versus reverberant conditions at two different signal-to-noise ratios (SNRs) were examined. 60 monolingual American English listeners transcribed spoken English sentences under different noisy conditions. Overall, we tested two acoustic conditions: anechoic and reverberant, reverberation time = 1.5 s, with four speech conditions (target speech only, or combined with HVAC noise, two-talker babble, or both) and two different SNRs (0 dB and -3 dB). Both the target and babble speech were female voices speaking in English. Complex backgrounds resulted in decreased performance, and the detrimental effects of such masking were accentuated by reverberation and lower SNR. This study serves as a basis for future research exploring the effects of room acoustics on speech intelligibility to gain a more wholistic understanding of speech perception under realistic conditions.

8:15

3aSC2. Impact of background noise and dysphonia on elementary students' listening comprehension and listening effort. Silvia Murgia (Speech and Hearing Sci., Univ. of Illinois - Urbana Champaign, 901 South Sixth St., Champaign, IL 61820, smurgia2@illinois.edu), Mary M. Flaherty (Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, Champaign, IL), Kara Federmeier (Dept. of Psych., Univ. of Illinois Urbana Champaign, Champaign, IL), and Pasquale Bottalico (Dept. of Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, Champaign, IL)

This project investigated how background noise and dysphonic voice impact elementary students' listening comprehension and effort. Listening comprehension is essential for academic success, and this study explored the extent to which adverse conditions hinder sentence comprehension and the mental resources utilized. The experiment used speech material recorded by an actress with normal and dysphonic voice quality. Nineteen 8–12 year-old children participated, using a computer in a laboratory with a talkbox for stimuli and loudspeakers for classroom noise at two signal-to-noise ratios (SNRs) of + 6 dB and 0 dB. Listening effort was measured subjectively on a 5-point scale and objectively through response time, the seconds taken to select a figure onscreen. Executive functions like working memory, attention, and inhibitory control were also assessed. Results showed that lower SNR and dysphonia significantly affected comprehension accuracy and increased perceived listening effort. Inhibitory control correlated with increased perceived effort. Similarly, both factors significantly lengthened response times. However, no significant correlations with executive functions were observed for comprehension or response time. In conclusion, background noise and dysphonia significantly affect listening comprehension and effort among elementary students. These findings emphasize the need for educators and policymakers to mitigate these factors for optimal classroom learning conditions.

8:30

3aSC3. Building a real-time convolution system for assessing vocal health of teachers. Bethany Wu (Dept. of Phys. & Astronomy, Brigham Young Univ., N284 ESC, Provo, UT 84602, bwu98@student.byu.edu) and Brian E. Anderson (Phys. & Astronomy, Brigham Young Univ., Provo, UT)

Due to prevalent vocal health issues in teachers, the acoustics of K-12 classrooms has become a common topic of study in acoustics. One way to understand the impact of a classroom's acoustics on speech is through real-time convolution of speech with a binaural room impulse response (BRIR). This is done by having a talker seated in an anechoic chamber and their vocal effort can be assessed while using the real-time convolution system to simulate the acoustics of a variety of classroom conditions. Keeping the talker in one physical space provides more control over the testing environment. A system that can successfully execute convolution in real-time requires parameters to be fine-tuned, an optimized algorithm, and appropriate hardware. Current efforts and lessons learned during the development of this system are shared. Goals for a finished real-time convolution system include specific testing to determine the effects of background noise and reverberation on a teacher's vocal effort.

Invited Papers

8:45

3aSC4. Perceptual consequences of reverberant environments on spatial unmasking. Gabriel S. Weeldreyer (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, 1110 S 67th St., Omaha, NE 68182, gweeldreyer2@unl.edu), Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, Omaha, NE), and Z. Ellen Peng (Boys Town National Res. Hospital, Omaha, NE)

Spatial hearing provides access to auditory spatial cues that promote speech perception in noisy listening situations. However, reverberation degrades auditory spatial cues and limits listeners' ability to utilize these cues for segregating target speech from competing babble. Hence, spatial unmasking—an intelligibility benefit from a spatial separation between a target and masker—is reduced in reverberant environments as compared to free field. To understand the perceptual consequences of poorer spatial unmasking in reverberation, we assessed three aspects of functional spatial hearing in virtual reverberant environments: perceived auditory source width, auditory spatial acuity, and spatial unmasking. Three auditory environments were simulated and auralized using ODEON to vary interaural coherence (IC): (1) a control anechoic environment, (2) a classroom designed to meet classroom acoustics standards (IC = 0.58), and (3) a classroom of the same size with more severe reverberation (IC = 0.37). Individually measured head-related transfer functions were used to binaurally reproduce the auralized signals over headphones. We hypothesize that interaural decorrelation, the result of increasing reverberation, will broaden the perceived auditory source width with a cascading effect of reduced auditory spatial acuity and subsequently poorer spatial unmasking. Preliminary data from normal-hearing adults will be presented. [Work supported by NIH R21DC020532.]

9:05

3aSC5. Effect of reverberation and informational masking on speech perception by adolescents and young adults with cochlear implants. Z. Ellen Peng (Boys Town National Res. Hospital, 555 North 30th St., Omaha, NE 68131, ellen.peng@boystown.org) and Abbie A. Mollison (Boystown National Res. Hospital, Omaha, NE)

Children with severe to profound hearing loss fitted with cochlear implants (CIs) can develop speech with degraded auditory inputs. Little is known about how speech perception by CI users who developed speech through electrical hearing is affected by realistic reverberation—a source of additional degradation in acoustic inputs to the CI speech processor. Moreover, how these young CI users handle competing maskers with high informational masking, such as two-talker babble, in speech-in-noise perception is not well characterized. Both reverberation and competing maskers are relevant factors in everyday classroom listening scenarios. In this work, we studied a cohort of adolescents and young adults with early fitting of CIs. Speech reception thresholds (SRTs) were measured in five types of maskers: speech-shaped noise (SSN), SSN modulated by a speech envelope, two-talker babble in English and in French, and time-reversed two-talker English babble. SRTs from each masker type were repeated in an anechoic condition and two reverberation conditions mimicking a classroom meeting classroom acoustics standard versus a lecture hall. When compared with a group of age-matched normal-hearing listeners, young CI users showed steeper increase of masked SRTs by increasing reverberation but comparable release from informational masking. [Work supported by the Hearing Health Foundation.]

9:25–9:40 Break

9:40

3aSC6. Achieving "Good" acoustical conditions for speech communication in active university classrooms. Young-Ji Choi (Kangwon National Univ., 1 Ganwondaehak-gil, Gangwon-do 24341, Korea (the Democratic People's Republic of), youngjichoi@kangwon.ac.kr)

This paper discusses acoustical parameter values for achieving 'good' acoustical conditions for speech communication in active university classrooms. Both room acoustics and background noise influence desirable conditions in classrooms for speech communication. The useful-to-detrimental sound ratio ($U_{;50}$) and speech transmission index (STI) values were calculated from both measured clarity ($C_{;50}$) and signal-to-noise ratio (SNR) values in the active classrooms. To investigate the combined effects of room acoustics ($C_{;50}$) and background noise (SNR) on speech intelligibility, multiple regression analyses were performed regressing both predictors of speech intelligibility, $U_{;50}$(125-4k) and STI, on the combinations of $C_{;50}$(125-4k) and SNR(125-4k) values. If the room has a $C_{;50}$(125-4k) value of 6.3 dB or greater, a SNR(125-4k) value of 10 dBA or greater is required for achieving 'Good' or 'Excellent' acoustical conditions for both measures, $U_{;50}$ and STI, in active university classrooms. However, the required $C_{;50}$(125-4k) value is slightly different for both measures if the room has higher or lower SNR(125-4k) values of 10 dBA.

Contributed Paper

10:00

3aSC7. Enhancing speech comprehension in large classrooms: Addressing alignment delay in assistive listening. Thomas Kaufmann (Dept. of Speech and Hearing Sci., Arizona State Univ., 976 S Forest Mall, Tempe, AZ 85281, tkaufmann@asu.edu), Mehdi Foroogozar, Julie Liss, and Visar Berisha (Dept. of Speech and Hearing Sci., Arizona State Univ., Tempe, AZ)

This study investigated the impact of alignment delay between assistive listening audio signals and acoustic signals from loudspeakers in large classrooms. Focusing on speech-in-noise comprehension, subjective comprehension confidence, and listening effort, the research aimed to illuminate auditory processing challenges and suggests improvements for assistive listening technology. Using a modified QuickSIN speech-in-noise hearing test in a simulated large auditorium setting, 53 participants with normative hearing experienced varying signal-to-noise ratios and alignment delays. The findings reveal a significant and progressive impact of alignment delay on speech comprehension. The study emphasizes the importance of compensating for alignment delay in assistive listening system design in large classrooms to enhance speech comprehension and comprehension confidence while minimizing listening effort. These insights are crucial for individuals with hearing disabilities, who likely face greater challenges than listeners with normative hearing. The study advocates for improved standards in assistive listening system design to ensure auditory accessibility and comfort in educational settings.

Invited Papers

10:15

3aSC8. "I feel like a pack a day smoker": Teacher and student voices from noisy classrooms. Pam Millett (Deaf and Hard of Hearing Program, York Univ., 104 Winters College, 4700 Keele St., Toronto, ON M3J 1P3, Canada, pmillett@edu.yorku.ca)

Concerns about the effects of poor classroom acoustics on student learning (particularly on students who are deaf or hard of hearing) have been expressed since the 1950s, yet research continues to indicate that poor classroom acoustics are both common, and related to poorer student outcomes in academic achievement, attention, behavior, and most recently, mental health. Teachers are not immune to the detrimental effects of noise; for them, research indicates a higher incidence of voice problems, absenteeism and job stress. However, little is reported about what teachers and students themselves have to say about their experiences attempting to learn under adverse listening conditions, and what changes for them when acoustical conditions improve. Their voices are largely missing from the research literature but they illuminate the problem in ways that quantitative data does not always capture. This presentation by an educational audiologist with over 35 years of experience in classrooms will provide an overview of the research on learning under adverse listening conditions, with a focus on representing teacher and student voices through qualitative research and anecdotal reports. The title is a quote from a teacher describing her voice problems and fatigue after a day of teaching.

10:35

3aSC9. Long- and short-term implicit talker familiarity: Effects on children's word and sentence recognition. Mary M. Flaherty (Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, 901 S Sixth St., Champaign, IL 61820, maryflah@illinois.edu)

While previous research has highlighted the positive effects of talker familiarity on children's word recognition in quiet, its role in challenging listening conditions remains poorly understood for this age group. Additionally, the amount of familiarity required to yield benefits remains unclear. The current study explored implicitly acquired short and long-term talker familiarity effects in school-age children (8–12 years), utilizing speech or noise maskers. Experiment 1 assessed open-set sentence recognition in a two-talker-female masker, with the child's mother's voice serving as the target or masker speech. Experiment 2 measured closed-set word recognition in a noise masker, both before and after a 5-day exposure to a female talker. Results from Experiment 1 indicate that children benefit from familiarity when their mother is the target talker, but not when she is the masking speech. In Experiment 2, children exhibited familiarity effects after only a 5-day exposure period to a previously unfamiliar voice, demonstrating a significant impact of implicit short-term familiarity. Working memory and selective attention did not predict performance or benefit. Together, these findings underscore the importance of talker familiarity in children's speech recognition in challenging listening situations. The implications of these findings for understanding the mechanisms underlying familiarity effects will be discussed.

10:55

3aSC10. Summary of National Standards on classroom acoustics: The data collection. Virginia Tardini (Dept. of Industrial Eng., Univ. of Bologna, Viale Risorgimento 2, Bologna 40136, Italy, virginia.tardini2@unibo.it), Giulia Fratoni, and Dario D'Orazio (Dept. of Industrial Eng., Univ. of Bologna, Bologna, Italy)

Improving acoustic comfort in classrooms is paramount for enhancing intelligibility, fostering student concentration, and alleviating the vocal strain on teachers. The global discourse on acoustic quality in educational buildings has intensified in recent decades, with each country establishing its unique set of building codes, standards, guidelines, and voluntary point protocols to delineate acoustic quality parameters. Geometrical and architectural classroom variations, influenced by cultural and historical contexts, further contribute to the diversity of approaches. The present work provides a summary that investigates and compares local standards across various countries worldwide, aiming to offer a holistic understanding of the classroom acoustic landscape. The motivation for this summary lies in finding regulations that are not easily searchable on Scholar or Scopus. It involves human interpretation and consideration of building codes, often in local languages. More than 70 experts, distributed across continents, were queried about his local standards, their enforceability, room criteria, design rules; and non-acoustic factors like classroom geometry, average occupancy, and flexibility of learning spaces. By synthesizing insights from diverse regions, this study aspires to provide a global perspective on the state of acoustic quality in educational settings, shedding light on potential areas for improvement.

WEDNESDAY MORNING, 15 MAY 2024 ROOM 213, 8:00 A.M. TO 11:35 A.M.

Session 3aSP

Signal Processing in Acoustics, Acoustical Oceanography, Physical Acoustics, Computational Acoustics, and Underwater Acoustics: Bayesian and Machine Learning in Acoustics I

Paul J. Gendron, Cochair
ECE, University of Masschusetts Dartmouth, 285 Old Westport Rd., Dartmouth, MA 02747

Ning Xiang, Cochair
School of Architecture, Rensselaer Polytechnic Institute, 110 Eighth Street, Troy, NY 12180

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

Chair's Introduction—8:00

Invited Papers

8:05

3aSP1. Performance evaluation of decision trees and multilayer perceptrons in seabed classification. Diego Rios, Jack Tokuda (Dept. of Mathematical Sci., New Jersey Inst. of Technol., Newark, NJ), and Zoi-Heleni Michalopoulou (Dept. of Mathematical Sci., New Jersey Inst. of Technol., 323 M. L. King Boulevard, Newark, NJ 07102, michalop@njit.edu)

Accurate knowledge of the oceanic propagation medium, and, thus, seabed properties, is of paramount importance in source localization, especially of weak targets. In this work, we employ machine learning techniques to perform seabed identification and classification using impulse responses of different media; supervised learning is employed. We first design a decision tree architecture that relies on features selected from the impulse responses corresponding to the different media. Examples of these features are kurtosis, skewness, and strength of the signal. Training sets are created by identifying features from noisy signals and testing follows after extracting corresponding features from a different data set. Performance is evaluated as a function of Signal-to-Noise Ratio. A principal component analysis is also implemented for the investigation of the potential for dimensionality reduction. Subsequently, multilayer perceptrons are employed using identical data both for training and testing and the two machine learning techniques are compared; advantages and disadvantages of each are identified and discussed in this work. [Work supported by ONR.]

8:25

3aSP2. Parallel tempering in trans-dimensional Bayesian inversion for seabed geoacoustic models with many parameters per layer. Stan Dosso (School of Earth and Ocean Sci., Univ. of Victoria, School of Earth and Ocean Sci., University of Victoria, Victoria, BC V8W 2Y2, Canada, sdosso@uvic.ca), Charles W. Holland (Elec. and Comput. Eng., Portland State Univ., Portland, OR), Jan Dettmer (Earth, Energy, and Environment, Univ. of Calgary, Calgary, AB, Canada), and Yong-Min Jiang (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada)

Trans-dimensional (trans-D) Bayesian inversion is a powerful tool for estimating seabed geoacoustic models from ocean-acoustic data, combining quantitative model selection with parameter/uncertainty estimation. The approach applies reversible-jump Markov-chain Monte Carlo methods to sample probabilistically over the number of seabed layers and the corresponding geoacoustic parameters for each layer. Layers are added and removed during sampling, referred to as birth and death moves, respectively, changing the dimension of the model. However, the probability of accepting birth and death moves can approach zero for formulations that include many parameters per layer. This paper considers the use of parallel tempering to mitigate this degradation in efficiency. Parallel tempering employs a series of interacting Markov chains with successfully-relaxed acceptance criteria, achieved by raising the likelihood to powers of $1/T$, with T greater than or equal to 1 referred to as the sampling temperature. While only the $T = 1$ chain provides unbiased sampling, probabilistic interchange between chains provides a robust ensemble sampler that mixes more readily over the trans-D model space. The approach is illustrated for wide-angle reflection-coefficient inversion including compressional and shear parameters in the seabed model, resulting in a total of 5 unknown parameters per layer.

8:45

3aSP3. Probabilistic multiphysics inference with flexible coupling and data covariance estimation. Jan Dettmer (Earth, Energy, and Environment, Univ. of Calgary, Dept. of Geoscience, University of Calgary, Calgary, AB, Canada, jan.dettmer@ucalgary.ca), Pejman Shahsavari, and Jeremy Gosselin (Earth, Energy, and Environment, Univ. of Calgary, Calgary, AB, Canada)

We consider combining complementary information contained in multiple data types recorded from distinct physical processes interacting with the Earth's subsurface. Such multiphysics inference of non-invasive geophysical observations can improve the resolution of Earth structure and processes, but is plagued by many subjective choices that practitioners are commonly required to make. We present the method of probabilistic multiphysics inference that employs Bayesian statistics to overcome several requirements for subjective choices. To ensure appropriate data weights, full data covariance matrices are estimated during Markov chain Monte Carlo sampling. The layering structure of the subsurface is estimated with flexible coupling, where the number of homogeneous layers is treated as unknown and the number of geophysical parameters for each layer are unknown. The latter permits flexible coupling such that parameters for different physical processes are not required to share the same layering structure, which avoids over-parametrization. We consider two examples with elastic and electromagnetic waves. In the first example, the thicknesses of shallow (tens of meters) active and permafrost layers are better constrained by probabilistic multiphysics inference. The second example resolves cratonic structure, with reduced uncertainty of a sedimentary basin and for the depth of the lithosphere-asthenosphere boundary. [Work supported by an NSERC Discovery Grant.]

9:05

3aSP4. Uncertainty quantification for acoustical problems. Peter Gerstoft (Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093, gerstoft@ucsd.edu) and Ishan D. Khurjekar (Univ. of California, San Diego, La Jolla, CA)

Acoustical parameter estimation is a routine task in many domains and is typically done using signal processing methods. The performance of existing estimation methods is affected due to external uncertainty and yet the methods provide no measure of confidence in the outputs. Hence it is crucial to quantify uncertainty in the estimates before real-world deployment. Conformal prediction is a simple method to obtain statistically valid prediction intervals from an estimation model. In this work, conformal prediction is used for obtaining statistically valid uncertainty intervals for various acoustical parameter estimation tasks. We consider the tasks of DOA estimation and localization of one or more sources in an acoustical environment. The performance is validated on plane wave data with different sources of uncertainty including ambient noise, interference, and sensor location uncertainty, using statistical metrics. Results demonstrate that conformal prediction is a suitable and easy-to-use technique to generate statistically valid uncertainty quantification for acoustical estimation tasks.

9:25

3aSP5. Bayesian optimization for geoacoustic inversion. William F. Jenkins (Scripps Inst. of Oceanogr., UC San Diego, 9500 Gilman Dr., La Jolla, CA 92093, wjenkins@ucsd.edu), Peter Gerstoft, and Yongsung Park (Scripps Inst. of Oceanogr., UC San Diego, La Jolla, CA)

Geoacoustic inversion of high-dimensional parameter spaces is a computationally intensive procedure, often necessitating thousands of forward model evaluations to accurately estimate the geoacoustic environment, such as Markov chain Monte Carlo sampling. This study introduces Bayesian optimization (BO), an efficient global optimization technique, to estimate geoacoustic parameters with significantly fewer evaluations, typically on the order of hundreds. BO involves an iterative search within the parameter space to locate the global optimum of an objective function; in this study, the Bartlett power is used. BO consists of fitting a Gaussian process surrogate model to existing evaluations of the objective function, followed by selecting a new data point for evaluation using a heuristic acquisition function. The effectiveness of BO is showcased through its application to both simulated and real-world data from a shallow-water environment for multidimensional parameter space encompassing source location, array tilt, and seabed properties.

9:45–10:00 Break

10:00

3aSP6. Performance of a computational Bayesian approach for active localization of a mobile scatterer in a refractive ocean environment. Paul J. Gendron (ECE, Univ. of Masschusetts Dartmouth, Dartmouth, MA) and Abner C. Barros (ECE, Univ. of Masschusetts Dartmouth, 285 Old Westport Rd., Dartmouth, MA 02747, abarros1@umassd.edu)

Underwater active acoustic systems that employ limited aperture arrays suffer from reduced processing gain and poor angular resolution, thereby hindering inferential objectives such as localization and tracking. Accurate localization is further challenged by the short conventional coherence-time associated with mobile bodies and dynamic environments. A computational Bayesian approach is presented and expanded here, for joint inference of range, depth, and speed of a submerged mobile scatterer in a refractive and multipath environment [Barros and Gendron, JASA-EL, 2019]. The Gibbs-sampling based approach infers the joint posterior probability density (PPD) of the pressure field wave vectors associated with the angle/Doppler spread arrivals and then maps their joint PPD to the target state joint PPD through acoustic ray interpolation. Performance analysis at high frequencies and relevant ranges using simulated acoustic fields are presented. The posterior uncertainty is investigated as a function of aperture and SNR, and we summarize the PPD using posterior credible intervals. A bound on posterior variance due to uncertainty in sound speed is provided and explored using profiles from the HYCOM database.

10:20

3aSP7. A new seperation method for rotationg and static sound source localization. Ning Chu (Res. Ctr., Zhejiang Shangfeng Co., Renminxi Rd. No1818, Shao Xing, Zhejiang Province 310024, China, chuning1983@sina.com) and Ali Djafari (Res. Ctr., Zhejiang Shangfeng Co., Shao Xing, China)

Traditional sound source localization methods encounter significant challenges in simultaneously locating rotating and static sources. These challenges arise from the differing motion patterns of these two types of sound sources, and they are typically not situated on the same plane. To address this issue, a method based on Modal Composition Beamforming (MCB) and the equivalent source method is proposed for separating rotating and static sound sources that can fully utilize priori knowledge of the spatiotemporal properties of the sources. The proposed approach involves establishing a Rotating-Static Sources Power Propagation (R-S2P) model, utilizing the relationship between the beamforming output of equivalent sources and the actual beamforming output. Solving this model allows for matching the contributions of rotating and static equivalent sources. Simulations for three cases with different source strengths are presented, and the R-S2P model is solved using the Least Absolute Shrinkage and Selection Operator (LASSO). This method enables accurate separation and localization of rotating and static sources on different planes with varying relative intensities, even if the background noise is strong

10:40

3aSP8. Dissipation estimation for the impedance tube measurement using Bayesian inference. Ziqi Chen (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, chenz33@rpi.edu) and Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY)

Although air dissipation is usually neglected in the impedance tube measurement, it affects the accuracy of the measurement results. Multiple effects contribute to the sound dissipation in an impedance tube, like the boundary layer effect or relaxation effect. This work formulates a transfer function model incorporating different dissipation. Bayesian inference is applied to estimate the acoustic dissipation in an impedance tube. Critical parameters in dissipation models are estimated based on the transfer function measured from an empty impedance tube. Nested sampling is applied to get an accurate approximation of parameters. The results of the analysis demonstrate that the critical parameters estimated by the Bayesian method are accurate. The accurate estimated dissipation helps improve the accuracy of impedance measurement with the material under test.

11:00

3aSP9. Parameter estimation and sensitivities of a Bayesian source localization method in an urban environment. Aaron Meyer (US Army Engineer Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755, aaron.c.meyer@erdc.dren.mil), Adrian Doran, D. Keith Wilson, and Matthew J. Kamrath (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. The performance of conventional source localization algorithms often suffers 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. In this presentation, a new source localization algorithm based on the Wishart distribution signal model is described and tested on real acoustic data. The experimental data were collected within a mock urban environment as part of a NATO urban acoustics-seismics experiment in Walenstadt, Switzerland. Four acoustic arrays recorded signatures from gunshots and ground vehicles. The present study uses the measured signals to investigate the sensitivity and relative importance of the model parameters, including source and noise amplitudes, frequency band, prior signal sampling, and sensor-sensor correlation behavior. Results are presented in the form of source location probability maps and localization error metrics. Interpretations of the study, including strengths and weaknesses of the new method, are discussed.

Contributed Paper

11:20

3aSP10. Performance of an approximate Bayes factor based detector for environmentally informed active sonar. Kenneth Bowers (Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., Dartmouth, MA, kbowers@umassd.edu) and Paul J. Gendron (ECE Dept., Univ. of Masschusetts Dartmouth, Dartmouth, MA)

Developed here are a space of approximate Bayes Factor (BF) inference processors for high frequency broadband active sonar with short vertical arrays operating in shallow water ocean waveguides that exploit relatively depth invariant modes of propagation. The relevant information regarding the refractive media, rough surface and volume reverberation are incorporated to build the marginal likelihoods under each of the composite hypotheses of null and alternative. Acoustic scattering from a mobile object of interest under depth uncertainty characterizes the compound alternative hypothesis. Approximations are presented and inferences regarding the presence of the mobile body of interest are determined against a composite null hypothesis of reverberation and ambient acoustic noise. The approximate BF processor is shown to be a time-varying quadratic form of array observations over the beam-delay space. We demonstrate the sub-space processing of depth invariant modes at range and illustrate the BF inferential approach on a few representative waveguides. Performance in classic terms of probability of detection as a function of false alarm rate are presented. [This work supported by the Office of Naval Research.]

WEDNESDAY AFTERNOON, 15 MAY 2024 ROOM 201, 1:00 P.M. TO 1:45 P.M.

Session 3pAA

Architectural Acoustics: Soundscape Simulation: Opportunities and Challenges II

Catherine Guastavino, Cochair
CIRMMT & School of Information Studies, McGill University, McGill University, School of Information Studies, 3661 Peel, Montreal, H3A 1X1, Canada

Josep Llorca-Bofi, Cochair
Institute for Hearing Technology and Acoustics, RWTH Aachen University, Kopernikustrasse 5, Aachen, 52074, Germany

Andre Fiebig, Cochair
Engineering Acoustics, TU Berlin, Einsteinufer 25, Berlin, 10587, Germany

Contributed Papers

1:00

3pAA1. What is realistic enough? Design considerations for soundscape simulators. Richard Yanaky (Information Studies, McGill Univ. / Ctr. for Interdisciplinary Res. in Music Media and Technol., 3661 Peel St., Montreal, QC H3A 1X1, Canada, richard.yanaky@mail.mcgill.ca) and Catherine Guastavino (Information Studies, McGill Univ. / Ctr. for Interdisciplinary Res. in Music Media and Technol., Montreal, QC, Canada)

Urban soundscapes reflect life and behaviours in a city. Given the dynamic and ever-changing behaviours of people, this introduces many complexities for soundscape simulators. We focus on bottom-up simulators, wherein every aspect of an environment can potentially be simulated, including 3D models of spaces, people's behaviours, sound sources, weather, nature, lighting, and the physics behind them to create the illusion of navigating a real space. How much realism is necessary in such virtual environments though? This work discusses realism through the lenses of plausibility, hyperrealism, and ecological validity. Is it ok to *just* be a plausible reality? Should reality be exaggerated? How faithful should it be and from which perspective (e.g. matching physical or cognitive "realities")? These questions were considered during the development of our in-house soundscape simulator, City Ditty. City Ditty seeks to be operable by non-sound professionals and support integration into urban projects with minimal expertise and resources, thus encouraging more diverse urban professionals and city users to contribute to how their cities will sound through participatory approaches. Given these requirements, we discuss how suitable levels of realism can be attained to fit people's needs at technical, practical, and theoretical levels by considering these three lenses for design.

1:15

3pAA2. Case study: Computer modeling the noise produced by a future food court radiated to nearby existing offices. Hong Tong (MJM Acoust. Consultants inc., 753 Ste-Helene St., Longueuil, QC J4K 3R5, Canada, htong@mjm.qc.ca)

Lately, there's an increase in the co-existence of spaces of different usage with acoustical challenge. Throughout 2016 to 2018, the 2nd floor of the Complexe Les Ailes, in Montreal, underwent a fit-out to become a food court. At its center, there is an elliptical atrium where all levels are can be viewed. There are existing offices above the food court separated by a glass pane. The purpose of this study was to evaluate the noise generated by human activities in the future food court to the adjacent offices and recommend noise control elements if needed. Multiple sound samples were taken at existing food courts to quantify the sound level. Simulations were done with ODEON, a room acoustic software, to evaluate the sound level produced by different activities. On-site measurements were conducted to calibrate the existing conditions with the 3D ODEON model. Finally, noise reduction tests were undertaken to determine the sound levels that would be radiated in the offices by the activities in the food court. To reduce noise disturbance in the offices, acoustical treatment under the ceiling of the food court was recommended and the 3rd and 4th floor glass panes composition would have to be improved.

1:30

3pAA3. Comparing subjective similarity ratings and quantitative errors for the evaluation of free-field binaural panning techniques. Zane T. Rusk (The Penn State Univ., 104 Eng. Unit A, University Park, PA 16802, ztr4@psu.edu), Matthew Neal (Otolaryngol. and Comm. Disord., Univ. of Louisville and Heuser Hearing Inst., Louisville, KY), and Michelle C. Vigeant (The Pennsylvania State Univ., University Park, PA)

A free-field sound source can be accurately rendered over headphones via direct convolution with head-related impulse responses (HRIRs). Using binaural panning techniques, a free-field source can be approximated, which can be thought of as *HRIR reconstruction*. However, a loss of fidelity occurs related to the panning algorithm's spatial resolution. Previously, perceptual comparisons of noise bursts were made between panned sources and direct convolution with measured HRIRs. Reconstructions were generated using five different binaural panning methods, both with and without time-alignment of the HRIRs with separate application of the interaural time delay (ITD). To further explore the listening test results, the quantitative differences between the reconstructed and original HRIRs were investigated alongside the perceptual data. Several perceptually-motivated error metrics were evaluated, including errors in both ITD and interaural level difference. The fidelity of the reconstructed HRTF magnitude response was evaluated using metrics that leveraged auditory modeling steps. Both subjective and objective results support that HRIR time-alignment reduces the number of filters needed for high perceptual accuracy. A principal component-base rendering filter set produced the best subjective accuracy with only a small number of required filters (16-25). The ability to predict perceptually detectable degradation via quantitative errors will be discussed.

WEDNESDAY NOON, 15 MAY 2024 ROOM 215, 12:55 P.M. TO 2:00 P.M.

Session 3pAO

Acoustical Oceanography: Munk Award Lecture

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

Chair's Introduction—12:55

Invited Paper

1:00

3pAO1. Inference of geoacoustic model parameters from acoustic field data: Perspectives on Geoacoustic Inversion. Ross Chapman (Univ. of Victoria, University of Victoria, 3800 Finnerty Rd., Victoria, BC V8P5C2, Canada, chapman@uvic.ca)

Estimation of parameters of geoacoustic models from acoustic field data has been a central research theme in acoustical oceanography and ocean acoustics. During the past several decades, highly efficient numerical inversion techniques have been developed that provide model parameter estimates and their uncertainties based on statistical inference methods. However, the methods are model-based and the inversions are prone to errors related to model mismatch. In any event, the inversions can generate only effective models of the true structure of the ocean bottom, which is generally highly variable over relatively small spatial scales in range and depth. There are also questions about the theory for modelling sound propagation in porous sediment media that raise doubt about the validity of inversion results. In most inversions, a visco-elastic theory is used, but is this the most appropriate propagation model? Another question is about the impact of neglecting shear waves in geoacoustic models. Most inversions assume a fluid model of the ocean bottom. This paper revisits issues that have raised questions about limitations of geoacoustic inversion methods, and discusses the impact of various mitigation measures that have been applied. The paper concludes with musings about new inversion techniques based on machine learning.

Session 3pED

Education in Acoustics: Acoustics Education Prize Lecture

Chair's Introduction—1:00

Invited Paper

1:05

3pED1. Building the acoustics program at Brigham Young University. Scott D. Sommerfeldt (Brigham Young Univ., N283 ESC, Brigham Young University, Provo, UT 84602, scott_sommerfeldt@byu.edu)

The history of acoustics at Brigham Young University dates well back into the first half of the twentieth century. Nonetheless, at the end of the twentieth century the program was on the verge of dying off. The author was hired in the Department of Physics and Astronomy at that time, in the hopes of revitalizing the acoustics program. Various steps were taken at opportune times to continuously rebuild the program to where it has become well known again – both nationally and internationally. The acoustics program is characterized by both a very strong undergraduate program in acoustics, as well as a thriving graduate program that has produced over 70 graduate theses and dissertations and even more undergraduate theses and capstone papers. This presentation will overview the growth and development of the acoustics program since 1995, highlighting lessons learned that may be useful for others looking to develop or strengthen their programs.

Session 3pID

Interdisciplinary: Keynote Lecture: Vibrotactile Perception of Music

Stan Dosso, Chair
School of Earth and Ocean Sciences, University of Victoria, School of Earth and Ocean Sciences, University of Victoria, Victoria, V8W 2Y2, Canada

Chair's Introduction—2:15

Invited Paper

2:20

3pID1. Vibrotactile perception of music. Frank A. Russo (Toronto Metropolitan Univ., Dept. of Psych., Toronto, ON M5B 2K3, Canada, russo@torontomu.ca)

While music is primarily experienced through audition, many aspects of music can also be experienced through somatosensation. Vibrotactile stimulation occurs when a tactile stimulus displaces the skin at a specific carrier frequency. When the carrier is frequency modulated, it can create a complex waveform and when it is amplitude modulated, it can create a rhythmic pulse. These vibrotactile aspects of music have long been recognized as important by music performers, providing a valuable source of non-auditory feedback that may support musical expression. Critically, these same vibrotactile aspects of music may also support the perception of music and voice in individuals with hearing loss. This paper will provide an overview of research conducted over the last decade that has clarified the aspects of music and voice that may be perceived through somatosensation along with some insight into neural underpinnings of these perceptions. Specific features of music considered will include pitch, rhythm, and timbre. Also considered will be the manner by which sensitivity to these features may change as a function of hearing loss, experience, and the properties of stimulation.

3p WED. PM

Session 3pNS

Noise: Assorted Topics on Noise II

Aaron B. Vaughn, Cochair
Structural Acoustics Branch, NASA Langley Research Center, 1 NASA Drive, Hampton, VA 23666

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

Contributed Papers

1:00

3pNS1. Ongoing confirmation of an objective criterion predicting annoyance linked to wind turbines. William K. Palmer (76 Sideroad 33-34, RR 5, Paisley, ON N0G 2N0, Canada, trileaem@bmts.com)

The 2023 CAA Acoustics Week in Canada introduced a criterion based on an objective acoustic measure to predict annoyance subjectively identified by residents living in the vicinity of wind turbines. That evidence was gathered primarily at a site near constant speed, stall regulated wind turbines. This paper presents subsequent investigations confirming that the criterion is also effective at a site with variable speed pitch regulated wind turbines. The results arise from the analysis of over 400 days of sampling at a site 787 m from the nearest wind turbine, with 18 turbines within 3 km. Verification of the study data was shown by comparison to data collected by an acoustic contractor employed by the provincial Ministry of the Environment. That data was collected through over 70 recording periods to analyze times residents at the site identified annoyance during a Ministry audit. These samples were obtained through a Freedom of Information request purchase. Analysis of the samples confirms the annoyance criterion identified at the CAA 2023 conference applies also for a different turbine type to objectively predict the annoyance subjectively identified by residents. The implications of this confirmation will be discussed.

1:15

3pNS2. A review of the modelling of impulsive noise. Donal Finnerty (Aercoustics Eng. Ltd., 1004 Middlegate Rd #1100, Mississauga, ON L4Y1M4, Canada, donalf@aercoustics.com)

It is generally acknowledged that impulsive noise can be perceived to be more annoying than steady noise with the same equivalent continuous sound level. This increased level of annoyance is often reflected in the regulations of noise emissions, in the form of applied penalties or specific modelling and measurement requirements for impulsive sources. To ensure that facilities that emit impulsive noise comply with those regulations, accurate modelling of impulsive noise is necessary. The modelling of impulsive noise is commonly performed using the same software and methodologies established to predict the impact of steady noise, such as ISO 9613-2. This paper seeks to evaluate the effectiveness and accuracy of these propagation methods when predicting the impact of impulsive noise.

1:30

3pNS3. Ontario Class 4 classification. Giuseppe Garro (Noise, Vib., and Acoust., Stantec, 1331 Clyde Ave. #300, Ottawa, ON K2C 3G4, Canada, giuseppe.garro@stantec.com)

The Ministry of Environment, Conservation, and Parks introduced the Class 4 noise area classification for land-use planning applications in 2013. A Class 4 classification is used in special cases where new noise sensitive developments are proposed in proximity to existing, lawfully established stationary noise source(s). This infrequently used classification is typically applied for in-fill developments in urban settings where land-use compatibility concerns are common. This work summarizes the Class 4 design criteria and discusses the practical implementation of the classification in the context of land-use planning.

Session 3pPP

Psychological and Physiological Acoustics: Auditory Neuroscience Prize Lecture

Sarah Verhulst, Chair
Ghent University, Technologiepark 126, Zwijnaarde, 9052, Belgium

Chair's Introduction—1:00

Invited Paper

1:05

3pPP1. The sing-song of old man human ear. Christopher Shera (Departments of Otolaryngol. and Phys. & Astronomy, Univ. of Southern California, 1969 Allen Ave., Altadena, CA 91001, christopher.shera@usc.edu)

Otoacoustic emissions evoked from the inner ear are the barely audible, signature by-product of the delicate hydromechanical amplifier that evolved within its bony walls. Compared to the sounds evoked from the ears of common laboratory animals, otoacoustic emissions (OAEs) from human ears have exceptionally long delays, typically exceeding those of cats, guinea pigs, and chinchillas by a factor of two to three. This presentation asks "Why are human OAE delays so long?" and reviews efforts to find answers in the mechanical frequency selectivity of the inner ear. The road to understanding species differences in OAE delay has led to the identification of new invariances and the emergence of new questions.

Plenary Session and Awards Ceremony

Stan E. Dosso
President, Acoustical Society of America

Annual Membership Meeting

Presentation of Certificates to New Fellows

Ahmad T. Abawi – For contributions to modeling underwater sound propagation and scattering

William C. Alberts, II – For contributions to tactical infrasound and battlefield acoustics

Simone Baumann-Pickering – For contributions to passive acoustic monitoring of beaked whales

Ana M. Jaramillo – For service to the Society, outreach to Spanish speakers, and acoustics education

Jungmee Lee – For contributions to auditory processing of simple and complex signals

Michael L. Oelze – For contributions to quantitative ultrasound tissue characterization

Introduction of Award Recipients and Presentation of Awards

Science Writing Awards to Francesco Aletta, David George Haskell, and Ute Eberle

Walter Munk Medal of The Oceanography Society to N. Ross Chapman

Rossing Prize in Acoustics Education to Scott D. Sommerfeldt

Medwin Prize in Acoustical Oceanography to Julien Bonnel

William and Christine Hartmann Prize in Auditory Neuroscience to Christopher Shera

Silver Medal in Acoustical Oceanography to Stan E. Dosso

Wallace Clement Sabine Medal to Peter D'Antonio

R. Bruce Lindsay Award to Christopher Kube

Helmholtz-Rayleigh Interdisciplinary Silver Medal to D. Keith Wilson

Gold Medal to Ingo R. Titze

Vice President's Gavel to Ann Bradlow

President's Tuning Fork to Stan E. Dosso

Canadian Acoustical Association

Umberto Berardi

President, Canadian Acoustical Association

ACOUSTICAL SOCIETY OF AMERICA

Silver Medal in

Acoustical Oceanography

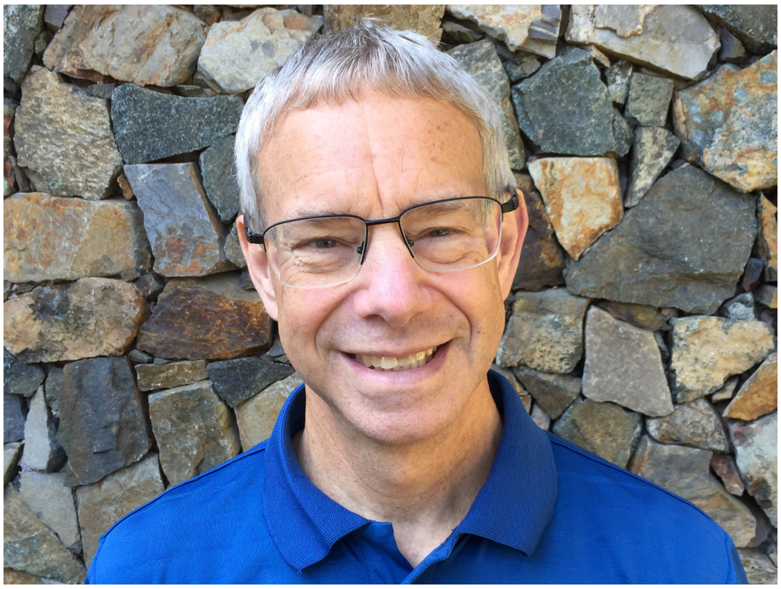
Stan E. Dosso

2023

The Silver Medal is presented to individuals, without age limitation, for contributions to the advancement of science, engineering, or human welfare through the application of acoustic principles, or through research accomplishment in acoustics.

PREVIOUS RECIPIENTS

Clarence S. Clay	1993
Herman Medwin	1997
D. Vance Holliday	2004
Robert C. Spindel	2009



ENCOMIUM FOR STAN E. DOSSO

. . . for contributions to Bayesian inference methods in ocean acoustics and marine geophysics.

15 MAY 2024 • OTTAWA, CANADA

Stan Dosso grew up in Victoria, British Columbia (BC), Canada, in a family with a brother and two sisters. His mother was a nurse, his father was a physics professor at the University of Victoria (UVic). The family lived a short walking distance from the university, where Stan decided to enroll for undergraduate studies in physics and mathematics. His first experience with ocean acoustics was while working toward his BSc, when he spent a summer work term supervised by Ross Chapman at what was then the Defence Research Establishment Pacific (DREP). Stan then went on to earn an MSc in physics at UVic, supervised by Ross, studying ocean-acoustic slope effects on the BC continental slope. Stan's time at UVic coincided with the appearance of a happy face painted on the dome of UVic's astronomy observatory. While this is considered an iconic sight on the campus today, at that time the painting was an unauthorized student-led initiative, a hallmark of the 1982 Physics class to which Stan belonged. More than 40 years later, despite my best forensic efforts, the details behind this story are still a bit fuzzy, but I like to believe this marks the start of Stan's influence at UVic.

Stan then moved to the University of British Columbia in Vancouver for a PhD in geophysics, studying geoelectromagnetic induction with Douglas Oldenburg, and developing a career-long interest in geophysical inverse theory.

Stan was hired at DREP in Victoria directly out of grad school and he worked there for 5 years, doing annual multi-week field camps on the polar ice pack. Stan always has been an avid outdoorsman, and I know he loved his time at sea and on the ice, despite seasickness interrupting data collection for his MSc, some close calls with polar bears, ice-floe breakups, and unsuccessful experiences at trying to brew coffee for colleagues. His research with DREP involved sea-ice seismo-acoustics, array element localization, and early work in geoacoustic inversion. In 1995, Stan was hired as an Ocean Acoustics Research Chair and Assistant Professor in the School of Earth and Ocean Sciences at UVic. In those early professional days, Stan and his wife Shelley raised three sons. With three young children, Stan on tenure-track, and Shelley working nights and weekends as a nurse, this was a very busy period, but among Stan's favorite times.

This led to a very successful and prolific career at UVic, where Stan continued his research in ocean acoustics and geophysical inversion problems. The over-arching theme of Stan's research is to quantify the information content of observational data to study physical systems that cannot be observed directly. His contributions in ocean acoustics cover the estimation of environmental parameters, such as of the seabed, water column and sea ice, but also source parameters, such as location or spectra. Stan notably wrote two seminal papers published in the *Journal of the Acoustical Society of America* in 2002 in which he formulated a numerical Bayesian approach to geoacoustic inversion. More recently, his flagship contribution has been the trans-dimensional, or transD, inversion method that he and then post-doc Jan Dettmer (now professor at University of Calgary) developed and introduced to the ocean acoustics community. When applied to seafloor geoacoustic inversion, transD notably enables researchers to learn the seabed parameterization, for example the number of sediment layers, from the data rather than fixing it arbitrarily, which otherwise biases the results.

Geoacoustic inversion has probably been Stan's main research area: he has worked with diverse types of ocean acoustic data including matched field, reflection coefficients, ray travel times, modal dispersion, interface-wave dispersion, reverberation, scattering, ambient noise, ship noise, and vector-sensor data. Quantifying the relative information content of different acoustic observables to estimate various seabed model parameters has been incredibly significant, a clear improvement over the sometimes ad-hoc methods that were used previously. In the past few years, Stan has collaborated with many acousticians to provide a detailed understanding of fine-grained sediment properties and their acoustic response. The use of transD methods was instrumental in gaining an improved understanding the seafloor of the New England Mud Patch, an area of high interest for our community. Stan's methods especially resulted in a clearer comprehension of the gradual transition between the mud and sand sub-bottom layers, a geological feature that was poorly understood before that.

Given the computationally intensive numerical methods employed, algorithmic research has also been a significant part of Stan's work. In his early work, Stan proposed efficient global optimizations methods, and later turned his attention to Bayesian sampling and error modeling. His current transD Bayesian algorithm provides a powerful approach to large-scale problems with many tens of unknown parameters. It is fair to say that Stan's contribution is bigger than just acoustics, and has impacted wave physics in general. His original work in acoustic inversion, including parallel tempering, covariance modeling, and transD birth-from-the-prior, have become well-accepted methods in the larger field of geophysical inversion. In short, I do believe Stan is an inverse problem genius who understands wave propagation, computational methods, and advances acoustical oceanography by using his theoretical skills to solve real world problems.

Impressively enough, Stan is much more than just an inversion nerd, he is also an outstanding educator and mentor. Stan served as the Director of the School of Earth and Ocean Sciences, a major department of UVic, from 2016 to 2021. Over the course of his career at UVic, Stan has been teaching both at the undergraduate and graduate levels and supervised many graduate students and postdocs, several of whom continued with very successful careers and won awards from the Acoustical Society of America (ASA).

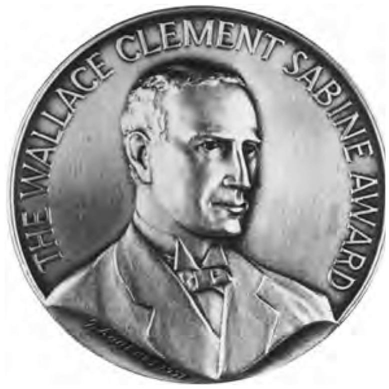
Stan is also very active in the ASA. His most notable contributions have been to chair the 176th ASA meeting in Victoria, to co-organize four ASA Schools, and to be elected to leadership positions including Vice President (2019-22) and President (2022-24).

In recognition of the significance and impact of Stan's research, he was recognized as an ASA Fellow in 2001, and was awarded the Medwin Prize in Acoustical Oceanography in 2004. Two years later Stan's teaching and research mentorship were recognized by UVic's Faculty of Science Teaching Excellence Award (2006).

For all these reasons, I cannot think of a more deserving recipient of the ASA Silver Medal in Acoustical Oceanography. Congratulation Stan, it is a well-deserved honor!

JULIEN BONNEL

WALLACE CLEMENT SABINE AWARD
OF THE
ACOUSTICAL SOCIETY OF AMERICA



Peter D'Antonio

2023

The Wallace Clement Sabine Award is presented to an individual of any nationality who has furthered the knowledge of architectural acoustics, as evidenced by contributions to professional journals and periodicals or by other accomplishments in the field of architectural acoustics.

PREVIOUS RECIPIENTS

Vern O. Knudsen	1957	Russell Johnson	1997
Floyd R. Watson	1959	Alfred C. C. Warnock	2002
Leo L. Beranek	1961	William J. Cavanaugh	2006
Erwin Meyer	1964	John S. Bradley	2008
Hale J. Sabine	1968	J. Christopher Jaffe	2011
Lothar W. Cremer	1974	Ning Xiang	2014
Cyril M. Harris	1979	David Griesinger	2017
Thomas D. Northwood	1982	Michael Vorländer	2018
Richard V. Waterhouse	1990	Gary W. Siebein	2020
A. Harold Marshall	1995		

SILVER MEDAL IN
ARCHITECTURAL ACOUSTICS

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 RECIPIENT

Theodore J. Schultz 1976

ENCOMIUM FOR PETER D'ANTONIO

. . .for contributions to theory, design, and application of acoustic diffusers.

15 MAY 2024 • OTTAWA, CANADA

Peter D'Antonio was born in Brooklyn, New York. His family introduced him to the music of the big bands and close vocal harmony groups, and he also played French horn in a competing parish drum and bugle corps. His love of music continued into the doo-wop era of the 1950's, when he formed an acapella group called the Dialtones, and in later years, he sang and played guitar in a cover band. He earned a B.S. in Physical Chemistry from St. John's University (1959-1963). His graduate studies were at the Polytechnic Institute of Brooklyn, where he graduated with a Ph.D. in Infrared Spectroscopy and a minor in X-ray crystallography. Following graduation, he began his professional career as a diffraction physicist at the Naval Research Laboratory in Washington, D.C. under the supervision of Nobel Laureate Dr. Jerome Karle. Peter studied the structure of matter in gaseous, amorphous, and crystalline phases, using electron, x-ray, and neutron diffraction, as well as electron and atomic force microscopy. He retired in 1996 following a distinguished career in fundamental structural research to focus on his passion for music composition, performance, recording, and a new career in architectural acoustics.

Peter began recording his compositions informally in his home in the early 1970s. As the music evolved and collaborators increased, there was a need for a soundproof recording studio, which he planned to locate in the basement of his residence. Never having designed a recording studio, his literature search revealed an article by Manfred Schroeder in the October 1980 issue of Physics Today, which described number-theoretic reflection phase grating (RPG) diffusors that were capable of uniformly scattering sound in concert halls. Inspired by these concepts and additional information on the acoustics of larger performance spaces he created a novel design for critical listening rooms. The design utilized a reflection-free zone, surrounding the listener to minimize interfering early reflections and simulate the initial time delay in larger spaces, and a diffuse field zone created with RPG diffusers on the rear wall to uniformly scatter incident sound and create an enveloping passive "surround" sound. Delving deeper into RPG theory, he surprisingly discovered that these diffusers were 2-dimensional periodic arrays similar to the 3-dimensional periodic crystal lattices he had been studying as a diffraction physicist. Peter was then able to design and model diffuser systems for recording studios using the Fraunhofer diffraction theory. The new design evolved into a commercial recording studio called Underground Sound.

Peter founded RPG Diffusor Systems in 1983 to evolve diffusive technology and manufacture commercial products. The first professional installation of the RPG diffusers was on the rear wall of the Oak Ridge Boys' Acorn Sound Recorders in Henderson, TN, in 1984. The new design became the standard in the recording industry.

Dr. D'Antonio was invited to join a team making acoustic measurements at Carnegie Hall for its 100th anniversary in 1989. These measurements resulted in the installation of RPG diffusers on the rear wall to remove a problematic slapback echo.

To verify the performance of diffusing surfaces, he designed and built the first experimental goniometer in 1993 to measure scattered polar responses at various angles of incidence. He and Trevor Cox developed a new diffusion coefficient from these polar responses, which was eventually standardized as ISO 17497-2 in 2012. Their collaboration also created three editions of the definitive text on sound absorbing and diffusing materials *Acoustic Absorbers and Diffusers: Theory, Design, and Application*. This notable effort is somewhat similar to the efforts made by Wallace Clement Sabine to quantify the acoustical properties of building materials as a major contribution to the field of architectural acoustics.

Dr. D'Antonio was invited to join the faculty at the Cleveland Institute of Music in 1991. During his tenure, he designed and experimentally evaluated a new variable acoustic modular performance shell (VAMPS) to provide variable local acoustics on stage. Following stage acoustic measurements at the Meyerhoff Symphony Hall in 2008, where Telarc was recording the Baltimore Symphony Orchestra, this new VAMPS design was installed to rave reviews from the orchestra and the conductor David Zinman.

Peter became Director of Research for a new company named RPG Acoustical Systems in 2017. He designed an Acoustical Research Center at the new facility incorporating a new experimental goniometer in addition to classical tools and the first virtual goniometer software (VIRGO), a Finite Element Model that predicts the polar responses and diffusion coefficient of a diffuser design. He also formed a company called REDI Acoustics with John Storyk and PK Pandy to create NIRO, the first AI Non-cuboid Iterative Room Optimizer software, to optimize critical listening rooms.

Beyond his professional accomplishments, Peter's original music library on Soundcloud and scholarly publications on Google Scholar serve as testaments to his enduring passion for both music and acoustics. His research methods have become the standards by which professionals and researchers around the world measure and model sound diffusion in rooms. His projects and publications established the state-of-the-art in the field of sound diffusion for decades. His willingness to discuss his research with students and practitioners in humble and forthright ways truly attests to his genuine and inspired desire to spread knowledge of architectural acoustics to the broadest possible audience. He has truly contributed in significant ways to the development of the "science of sound as it pertains to buildings" as defined by Wallace Clement Sabine.

Dr. D'Antonio constructively combined his passion for music and his diffraction physics experience to create an innovative career in architectural acoustics and pioneered the science of sound diffusion. His theories and innovations led to practical applications, transforming recording studios, educational facilities, worship spaces, performance venues, and home theaters. Peter was inducted into the recording industry's TECnology Hall of Fame in 2013. We are delighted and privileged that the Acoustical Society of America recognizes and honors Dr. Peter D'Antonio with this distinguished Wallace Clement Sabine Medal for contributions to the field of architectural acoustics.

NING XIANG
GARY W. SIEBEIN

R. BRUCE LINDSAY AWARD

ACOUSTICAL SOCIETY OF AMERICA

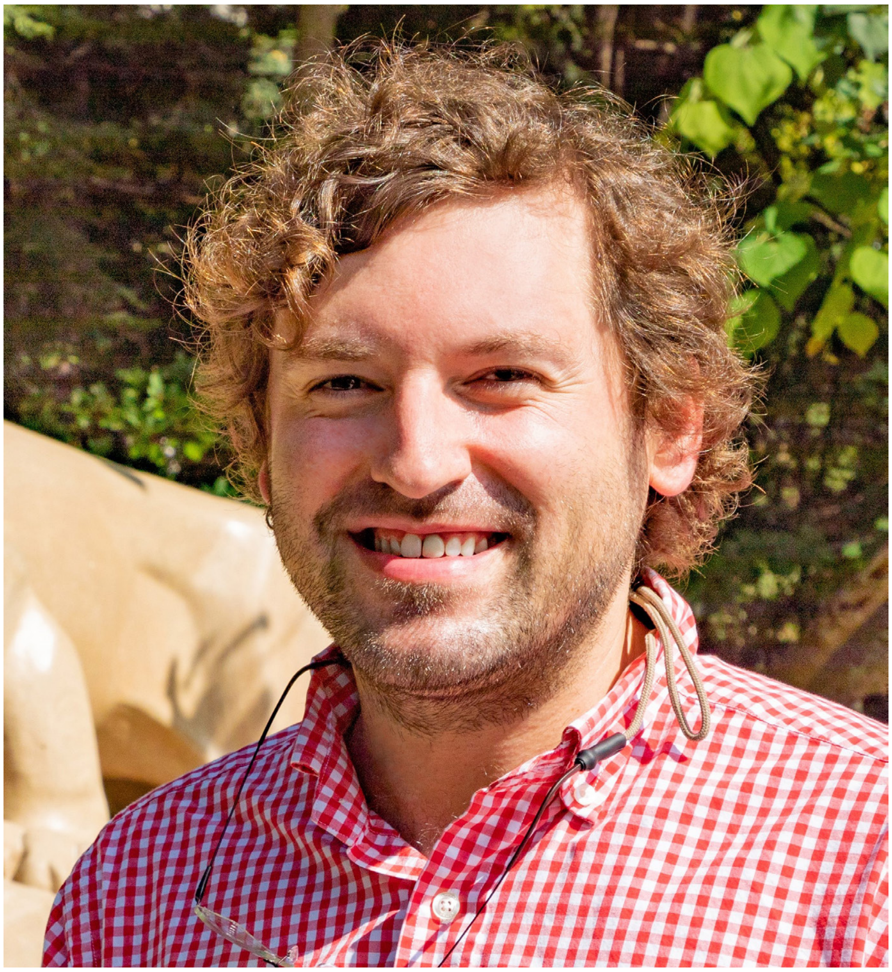
Christopher Kube

2024

The R. Bruce Lindsay Award (formerly the Biennial Award) is presented in the Spring to a member of the Society who is no more than 10 years post terminal degree on 1 July at the time of Award acceptance and who, during a period of two or more years immediately preceding the award, has been active in the affairs of the Society and has contributed substantially, through published papers, to the advancement of theoretical or applied acoustics, or both. The award was presented biennially until 1986. It is now an annual award.

PREVIOUS RECIPIENTS

Richard H. Bolt	1942
Leo L. Beranek	1944
Vincent Salmon	1946
Isadore Rudnick	1948
J. C. R. Licklider	1950
Osman K. Mawardi	1952
Uno Ingard	1954
Ernest Yeager	1956
Ira J. Hirsh	1956
Bruce P. Bogert	1958
Ira Dyer	1960
Alan Powell	1962
Tony F. W. Embleton	1964
David M. Green	1966
Emmanuel P. Papadakis	1968
Logan E. Hargrove	1970
Robert D. Finch	1972
Lawrence R. Rabiner	1974
Robert E. Apfel	1976
Henry E. Bass	1978
Peter H. Rogers	1980
Ralph N. Baer	1982
Peter N. Mikhalevsky	1984
William E. Cooper	1986
Ilene J. Busch-Vishniac	1987
Gilles A. Daigle	1988
Mark F. Hamilton	1989
Thomas J. Hofler	1990
Yves H. Berthelot	1991
Joseph M. Cuschieri	1991
Anthony A. Atchley	1992
Michael D. Collins	1993
Robert P. Carlyon	1994
Beverly A. Wright	1995
Victor W. Sparrow	1996
D. Keith Wilson	1997
Robert L. Clark	1998
Paul E. Barbone	1999
Robin O. Cleveland	2000
Andrew J. Oxenham	2001
James J. Finneran	2002
Thomas J. Royston	2002
Dani Byrd	2003
Michael R. Bailey	2004
Lily M. Wang	2005
Purnima Ratilal	2006
Dorian S. Houser	2007
Tyrone M. Porter	2008
Kelly J. Benoit-Bird	2009
Kent L. Gee	2010
Karim G. Sabra	2011
Constantin-C. Coussios	2012
Eleanor P. J. Stride	2013
Matthew J. Goupell	2014
Matthew W. Urban	2015
Megan S. Ballard	2016
Bradley E. Treeby	2017
Yun Jing	2018
Adam Maxwell	2019
Julien Bonnel	2020
Likun Zhang	2021
Meaghan A. O’Reilly	2022
Julianna C. Simon	2023



ENCOMIUM FOR CHRISTOPHER M. KUBE

...for contributions to the understanding of ultrasonic propagation and nonlinearity in polycrystalline materials.

15 MAY 2024 • OTTAWA, CANADA

Christopher Michael Kube grew up in Grand Island, the third largest city in Nebraska with a population of 50,000. Athletics, particularly running, was a big part of Christopher's growing up. He was a competitive runner, and after completing high school in Grand Island he attended nearby Hastings College on a track and cross-country scholarship. In between runs Christopher majored in Physics. For his senior project at Hastings College Christopher developed an acoustic levitation system using a $1000 budget and car audio amp borrowed from a friend, which opened his eyes to the field of acoustics.

Christopher's undergraduate experience with acoustics caught the attention of Professor Joseph Turner at the University of Nebraska-Lincoln (UNL) who recruited Christopher in 2008. He threw himself into the experimental challenges of his MS research project on ultrasonic measurement of stress in railroad rails, designing fixtures that attached magnetically and assembling the equipment necessary to make the measurements portable. The ultrasonic scattering response allowed Christopher to track the stress from thermal expansion surprisingly well over the length of a hot summer day. Over the course of this project, involving multiple trips to active rail lines in western Nebraska, Christopher came to the conclusion that the existing theory for inferring stress from ultrasound was inadequate. This motivated his interest in understanding and explaining how the polycrystalline nature of metals affects ultrasonic properties, which became the theme of his PhD research. Christopher's first major contributions in this area are notably a pair of papers published in the *Journal of the Acoustical Society of America (JASA)* in 2015 providing improved and accurate estimates for ultrasonic attenuation in polycrystals. Christopher's PhD research also produced very original work on estimating acoustoelasticity coefficients in polycrystalline and anisotropic solids. These parameters determine how stress affects ultrasound and conversely enable ultrasonic nondestructive evaluation (NDE) of *in situ* stress. Unquestionably the most significant result of Christopher's time at UNL was meeting a fellow graduate student in Turner's group, Andrea Arguelles, now wife, partner, collaborator and department colleague at Penn State, where they welcomed their son Jack in December during the same week as the Sydney meeting.

After his PhD in 2014 Christopher continued research in ultrasonics and nondestructive evaluation (NDE) at the U.S. Army Research Laboratory in Maryland. Notable among his achievements is a completely new theory for quantifying ultrasonic scattering from nonlinear inclusions. This work opens possibilities for NDE measurements based on localized nonlinearity of scatterers, whether they are impurities, welds or grains. In 2016 he proposed an "anisotropy index", a number that characterizes acoustic anisotropy that is applicable to any crystalline solid. In 2019 Christopher joined the Engineering Science and Mechanics department at Penn State University, where he is also a member of the Acoustics program. Although he remains an avid Cornhusker, his career so far at PSU has been tremendously productive. Not only has Christopher continued developing theories of nonlinear elastic waves and ultrasound in polycrystalline materials, but he has branched out into new challenging areas. Specifically, his initial research at the Applied Research Laboratory (ARL) on applications of ultrasonic NDE in additive manufacturing (AM) has blossomed in multiple directions. One example is application of ultrasonics to monitor laser generated melt pools, thereby providing a unique tool for studying solidification and microstructure formation in AM. Christopher's rapid rise to prominence in this important and evolving acoustics and materials science topic is reflected by the fact that he leads a Department of Energy project on ultrasonics for *in situ* material property monitoring in AM involving Carnegie Mellon University, the University of North Texas, and Westinghouse Electric Company. This level of leadership responsibility in a multi-institution effort is rare for an assistant professor. Other projects in ultrasonic applications in AM have been funded by the Navy and 3M and his work in this area has been recognized by his selection for US Air Force Laboratory (AFRL) Summer Faculty Fellowships. Christopher is particularly successful in establishing productive scientific collaborations, an aspect of research that he impresses on his students. This success is partly explained by Christopher's congenial and affable nature but is mainly attributable to the respect he commands from his peers.

Christopher has a prolific publication record with 47 journal articles to date. The ASA is his preferred venue, with 22 papers in JASA and 7 *in JASA Express Letters*. The number of fundamental papers on ultrasonic attenuation and nonlinear effects in polycrystalline materials is quite remarkable. Chris is also active in the ultrasonics and nondestructive testing communities. He serves as an associate editor of *Ultrasonics*, was invited to be editor for a special issue of *Nondestructive Testing and Evaluation International* on additive manufacturing and is now a member of the journal's editorial board.

His research contributions in ultrasonics have been recognized by the American Society of Nondestructive Testing through the Young NDT Professional Award in April 2020 and the Research in Nondestructive Evaluation Outstanding Journal Article, 2015. It is no surprise that Christopher is also an excellent teacher, as evidenced by the 2020 award of the Hastings College Outstanding Alumni Award for Teaching. Paraphrasing one eminent member of ASA, Christopher has an outstanding balance of intellectual ability and patience with a natural inquisitiveness, and quoting another, succinctly, "he is a class act."

Christopher's history of involvement with the Acoustical Society of America is spectacular. He has been a member since 2009 when as an MS student he attended and delivered a talk at the 158th meeting. Fast forward to 2023 when ASA members elected him to chair the Physical Acoustics Technical Committee. This trajectory is remarkable considering that Christopher is an assistant professor and it speaks volumes about his recognition and reputation within the Society. Christopher is also a member of the Structural Acoustics and Vibration Technical Committee and has reviewed for JASA and JASA-EL as well as many other journals. He has chaired and organized special sessions, e.g., Frontiers of Resonant Ultrasound Spectroscopy and its Applications and Nonlinear Acoustics in Solids for the fall 2022 and 2023 meetings. Two of his undergraduate students received the ASA Robert W. Young Award. ASA is clearly Christopher's home professional society, and he has proven himself to be a good citizen in terms of his service within both the national and international acoustics communities. His productivity in terms of the considerable number of substantial journal articles over the past eight years since receiving his PhD is extraordinary for a theoretician. Christopher is a true acoustician whose career is on a remarkable trajectory, embodying the spirit of this award. Therefore, we are proud to present Christopher the 2024 R. Bruce Lindsay Award.

ANDREW N. NORRIS

Helmholtz-Rayleigh Interdisciplinary Silver Medal
in
Computational Acoustics, Physical Acoustics, and Engineering Acoustics

ASA
ACOUSTICAL SOCIETY
OF AMERICA

D. Keith Wilson

2024

The Silver Medal is presented to individuals, without age limitation, for contributions to the advancement of science, engineering, or human welfare through the application of acoustic principles, or through research accomplishment in acoustics.

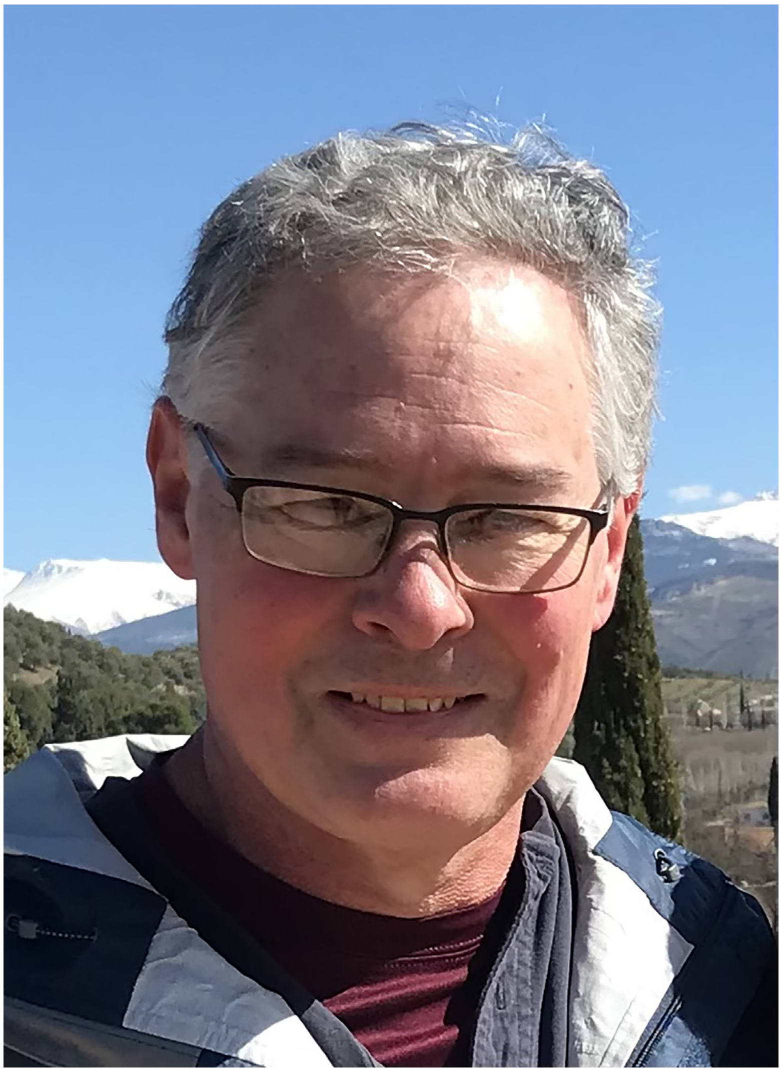
PREVIOUS RECIPIENTS

Helmholtz-Rayleigh Interdisciplinary Silver Medal

Gerhard M. Sessler	1997
David E. Weston	1998
Jens P. Blauert	1999
Lawrence A. Crum	2000
William M. Hartmann	2001
Arthur B. Baggeroer	2002
David Lubman	2004
Gilles A. Daigle	2005
Mathias Fink	2006
Edwin L. Carstensen	2007
James V. Candy	2008
Ronald A. Roy	2010
James E. Barger	2011
Timothy J. Leighton	2013
Mark F. Hamilton	2014
Henry Cox	2015
Armen Sarvazyan	2016
Blake S. Wilson	2017
Kenneth S. Suslick	2018
Barbara G. Shinn-Cunningham	2019
Michael R. Moldover	2021
George L. Augspurger	2022
Vera A. Khokhlova	2023

Interdisciplinary Silver Medal

Eugen J. Skudrzyk	1983
Wesley L. Nyborg	1990
W. Dixon Ward	1991
Victor C. Anderson	1992
Steven L. Garrett	1993



CITATION FOR D. KEITH WILSON

...for contributions to computational acoustics, atmospheric acoustics, and national defense and security

15 MAY 2024 • OTTAWA, CANADA

D. Keith Wilson earned an M.S. in Electrical Engineering from the University of Minnesota in 1987, where he first became interested in acoustics under Professor Robert Lambert, and a PhD in Acoustics from Pennsylvania State University in 1992, where his advisor was Professor Dennis Thompson. In his PhD dissertation he formulated theoretical principles of acoustic tomography of the atmosphere and carried out its experimental implementation. Since Keith's pioneering work, acoustic tomography has been used in several countries for remote sensing of the atmosphere and, also, to monitor temperature and air flow in closed environments such as boiler rooms and large halls. Keith was awarded the R. Bruce Lindsay Award from the ASA in 1997 for developing acoustic tomography of the atmosphere.

Upon completion of his PhD, Keith joined the Woods Hole Oceanographic Institution as a research fellow (1991-1993) under Dr. George Frisk and then became a member of the research faculty at the Pennsylvania State University Meteorology Department (1993-1995) under Professor John Wyngaard. The latter appointment and his years as a PhD student were particularly beneficial because they provided Keith with the opportunity to study atmospheric boundary layer (ABL) meteorology. Later, Keith used this knowledge to express the variances and outer length scales of temperature and wind velocity fluctuations in terms of the meteorological parameters of the ABL. His approach for modeling atmospheric turbulence has been widely used for near-ground sound propagation, auralization of flying aircraft, source localization, and the effect of turbulence on sonic boom disturbances. Many acousticians credit Keith for bringing ABL meteorology to atmospheric acoustics.

In 1995 Keith joined the U.S. Army Research Laboratory (ARL), near Washington, DC, as a physical scientist. Although his work at the ARL was rewarding and productive, small-town life in New England was attractive from a family perspective. So, Keith joined the Cold Region Research and Engineering Laboratory (CRREL), part of the U.S. Army Engineer Research and Development Center (ERDC), in Hanover, NH in 2002, where he has been working ever since.

Keith has made other outstanding contributions to atmospheric acoustics. He employed the relaxation model to describe sound interactions with the ground, which is usually modelled as a rigid-porous material. The theory developed is now known as *Wilson's relaxation model*, which is one of the main approaches for sound interactions with the ground. The ongoing revision of the ANSI S1.18 *Method for determining ground impedance* will recommend using Wilson's model.

Keith was among the first to *quantify uncertainties in outdoor sound propagation*. He used sophisticated methods for sampling random parameters of the problem such as stratified, Latin-hypercube, and importance sampling. Keith was also the first to recognize the potential in machine learning methods for fast predictions of outdoor sound propagation. His results have been used in predicting absorption/transmission properties of sound insulating media, wind farm noise, and infrasound propagation.

Keith has contributed significantly to modern computational atmospheric acoustics. In addition to machine learning and quantification of uncertainties, he developed a finite-difference time-domain algorithm for solving linearized fluid dynamics equations, which has been widely used in atmospheric acoustics and other fields. Also, Keith collaborated on the most advanced version of the parabolic-equation code, which correctly accounts for the temperature and wind velocity profiles, atmospheric turbulence, impedance ground, terrain elevations, wide-angle propagation, vector character of the wind velocity, and uncertainty.

Keith has made important contributions to battlefield acoustics and sensing. His work on national defense and security has been recognized by many distinguished awards from ARL, CRREL, and ERDC, including the prestigious Superior and Meritorious Civilian

Service Awards and recognition as a Fellow of the Military Sensing Symposia. One of his most notable contributions was a theoretical model for the performance of acoustic arrays operating in the atmosphere, including both the effects of turbulence and signal-to-noise ratio. Another important contribution is the development of the software Environmental Awareness for Sensor and Emitter Employment (EASEE), which models atmospheric and terrain impacts on sensor coverage. The EASEE software has been used by all five DoD services and several intelligence organizations for applications such as installation security, border protection, routing of aircraft, identifying coverage gaps in networks, and optimization of sensor types and locations.

Keith keeps a very busy schedule, managing from six to eight projects a year, involving several researchers. This requires extensive administrative work and travel. Yet Keith is remarkably productive as a researcher. People who know him closely are truly amazed that he can accomplish so many tasks. In addition to long hours, he works productively in airports and hotel rooms. Keith's hobbies include growing dahlias, listening to 1960's and classical music, and reading about history and physics. Keith's wife, Nancy, a music teacher, has provided unwavering support throughout his career, especially when he worked late or was away for meetings.

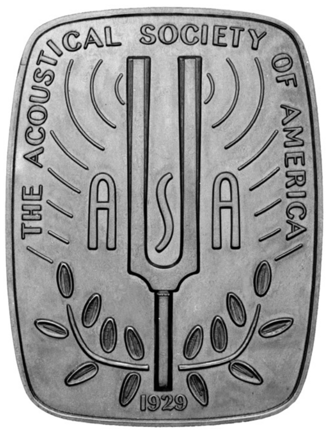
It is safe to say that ASA is Keith's *home*. He attends almost every ASA meeting with multiple presentations, including numerous invited talks. He has organized many special sessions. Most of his peer-reviewed articles are in *JASA* and *JASA EL*. He has been an Associate Editor for *JASA* and *JASA EL* for more than 20 years and served as a Chair of the Publication Policy Committee for six years.

Two of Keith's contributions to the ASA have already enormously benefited the Society and will continue doing so for years to come. He was the Founding Editor of *JASA EL* and served as its Editor in 2005-2009, leading the transition of *Acoustics Research Letters Online (ARLO)*, a former rapid review and publication Letters section of *JASA,* into a stand-alone publication under the *JASA EL* name. In 2018, Keith founded the new ASA Technical Specialty Group Computational Acoustics which has since been granted full status as a TC; the first new TC in 30 years. The creation of TCCA has also resulted in the new sections entitled *Computational Acoustics* in *JASA* and *JASA EL.* These sections have become increasingly popular and have attracted new authors (and readers) who would otherwise publish elsewhere.

In recognition of the above-mentioned contributions and achievements, the ASA honors Keith Wilson with the Helmholtz-Rayleigh Interdisciplinary Silver Medal in Computational Acoustics, Physical Acoustics, and Engineering Acoustics.

VLADIMIR E. OSTASHEV
KEITH ATTENBOROUGH
R. DANIEL COSTLEY

Gold Medal



Ingo R. Titze
2024

The Gold Medal is presented in the spring to a member of the Society, without age limitation, for contributions to acoustics. The first Gold Medal was presented in 1954 on the occasion of the Society's Twenty-Fifth Anniversary Celebration and biennially until 1981. It is now an annual award.

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R. Bruce Lindsay	1963
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Richard H. Bolt	1979
Harry F. Olson	1981
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Martin Greenspan	1983
Robert T. Beyer	1984
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Judy R. Dubno	2020
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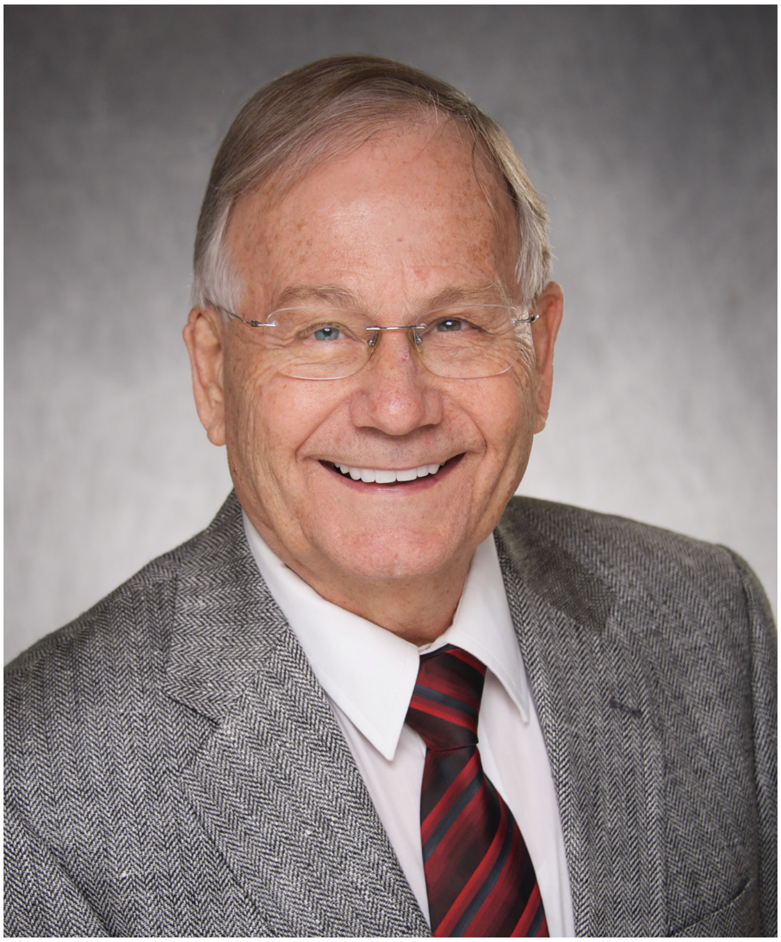

ENCOMIUM FOR INGO R TITZE

...for contributions to understanding human voice production and the development of clinical applications.

15 MAY 2024 • OTTAWA, CANADA

Ingo R. Titze was born in East Germany. He and his family moved to West Germany as refugees to the small town of Werdohl, where they lived for ten years, and then emigrated to the United States, settling in Salt Lake City, UT. It was here that Ingo attended high school and later earned Bachelor's and Master's degrees in Electrical Engineering from the University of Utah.

Following graduation, Ingo was employed for two years by North American Aviation (later North American Rockwell) in Tulsa, OK, where he worked on radar absorptive materials for nose cones of aircraft and applications for space exploration. He then spent two years working at the Boeing Company in Seattle, WA. Ingo eventually pursued a Ph.D. in Physics at Brigham Young University (BYU) in Provo, UT with the goal of combining his scientific curiosity with his lifelong passion for singing.

Ingo received his Ph.D. in 1973, and then served a short stint as Instructor of Physics at Pomona College before accepting his first appointment as Assistant Professor at the University of Petroleum and Minerals in Saudi Arabia. From 1976 to 1979, Ingo was an Assistant Professor at Gallaudet College and then moved to the University of Iowa (UI) in 1979 where he was named a UI Foundation Distinguished Professor in 1995. Ingo retired from the University of Iowa in 2019 and is currently engaged in research at the Utah Center for Vocology at the University of Utah. He also serves as the Chair of the board of directors for the National Center for Voice and Speech, a non-profit research, training, and dissemination enterprise begun in 1990.

Ingo was elected a Fellow of the ASA in 1983 and was awarded the ASA Silver Medal in Speech Communication in 2007. For more than four decades he has contributed to our scientific knowledge of voice production by painstakingly applying the principles of physics to understanding the aerodynamic, mechanical, and acoustic nature of vocal fold vibration. His work has greatly expanded our knowledge of the intricacies of how humans generate voiced sounds for purposes of communication and musical expression, while simultaneously laying the foundation for innovative clinical treatments of voice disorders. More recently Ingo has applied his vast knowledge of the human voice to understanding sound production in the animal world.

Among many landmark findings published in several hundred journal articles, Ingo's research legacy is his refinement of the "myoelastic-aerodynamic theory of phonation." Although the theory was so named by Janwillem van den Berg in 1958, it was Ingo who demonstrated mathematically how self-oscillation depends on the phase relations between the aerodynamic driving pressures acting on the vocal folds and the velocity components of the tissue movement. He showed that these conditions are facilitated by an aerodynamically induced biomechanical wave in the mucosal tissue of the vocal folds. Another of Ingo's major theoretical insights was the effect of the acoustic reactance of the vocal tract on the self-oscillation of the vocal folds. His legacy is further bolstered by investigations into how the fundamental frequency of vocal fold vibration and vocal intensity are controlled, how certain aspects of the laryngeal system differ across males and females, and the establishment of the concept of phonation threshold pressure.

Ingo's explication of the physical nature of the voice has resulted in far-reaching effects on voice science, voice care and therapy, and voice training. Through computational modeling, he has developed a system that simulates the biomechanics and acoustics of voice production allowing researchers to conduct virtual experiments with a wide range of parametric variation. Ingo's clear explanations of the complementary mechanisms of self-oscillation have strongly influenced the direction of voice research over the past 30 years, and his research has facilitated developments in speech synthesis, speaker identification, voice quality, and speech transformation. His work has also provided a theoretical basis for voice therapy techniques, guidance for laryngological interventions (e.g., surgery, injections, etc.), and informs voice coaches and trainers as they educate singers, entertainers, teachers, and other professional voice users. In addition, Ingo played a pivotal role in founding a new subfield called "vocology," which is dedicated to the care and treatment of voice disorders and the training of professional voice users.

Ingo has educated hundreds of students, clinicians, physicians, voice teachers, and professional vocalists about the science of the voice through his presentations in courses, conferences, workshops, and summer programs. Ingo is also an exquisite mentor who motivates students by sharing his own excitement about the next new thing that might be learned about how the voice works. His relentless enthusiasm is contagious. His book *Principles of Voice Production* was written specifically for a diverse audience so that highly theoretical research on the mechanics of voice production could be understood by those with less technical backgrounds. His more advanced book, *The Myoelastic Aerodynamic Theory of Phonation* is a classic exposition of theories and research of phonatory mechanics, aerodynamics, modeling, and simulation.

Ingo also developed the Summer Vocology Institute (SVI) that offers coursework for students and practicing professionals in voice science, instrumentation, and voice habilitation. The summer of 2024 will mark the 25th year of the SVI, with Ingo having served as the primary instructor. He has also taken leadership roles in organizing many conferences, served as an associate editor for the National Association of Teachers Singing Journal from 1986 to 2023, and reviewed hundreds of manuscripts for academic journals including JASA and JASA-EL. Ingo has served in several roles for the National Institutes of Health. He was on the Sensory Disorders and Language Study Section from 1986-1990, served as Chair of the Voice Panel during the planning of a National Strategic Research Plan in 1988, was a member of the task force charged with reorganizing the behavioral research study section in 1998-1990, and was a member of the prestigious Advisory Council for the National Institute on Deafness and Other Communication Disorders (NIDCD) from 2000 to 2004. In 2015, Ingo founded and served as President of the Pan American Voice Association, whose mission is to advance the scientific and interdisciplinary study of vocalization across species in all countries of the Western Hemisphere through research, training, and dissemination of knowledge.

At the 1993 fall meeting of the ASA in Denver, Colorado, Ingo served the Society in a unique way. At the plenary session held in the Boettcher Concert Hall at the Denver Center for the Performing Arts, he presented a show called ``Voices of People and Machines.“ It was open to the public and attracted hundreds of students from the Denver area schools, The show was part education, part music, part comedy, and culminated with Ingo (himself a fine tenor) singing an operatic duet with a numerically simulated voice coupled with an animated face called “Pavarobotti.“ It was a technically complex presentation to execute and posed a high risk of failure, but he and his team made it a success. There would be few other ASA plenary sessions in which one of our members took acoustics straight to the public quite as boldly as Ingo did at that ASA meeting in 1993.

In sum, Ingo Titze’s research and educational contributions, and the development of Vocology, have strongly shaped and directed theoretical, experimental, and clinical voice science over the past four decades. He loves learning, he loves teaching what he learns to others, and he inspires all those around him to pursue and produce new knowledge. He is a most deserving recipient of the Gold Medal of the Acoustical Society of America.

BRAD H. STORY
RONALD C. SCHERER
JOE WOLFE

Session 4aAA

Architectural Acoustics: Acoustics in Mass Timber

Evelyn Way, Cochair
Maxxon Corp, 920 Hamel Road, Hamel, MN 55340

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

Chair's Introduction—8:00

Invited Papers

8:05

4aAA1. The project design includes a mass timber structure. Now what? Jessica S. Clements (Acoust. Studio, Newcomb & Boyd, LLP, 303 Peachtree Ctr. Av NE, Ste. 525, Atlanta, GA 30303, jclements@newcomb-boyd.com)

Mass timber comes with any number of acoustical design challenges depending on the project type and use. The many advantages in sustainability and cost mean that mass timber design is here to stay. As an acoustical consultant what does this mean for your project? This presentation will review some of the typical concerns you might have or questions to consider asking of the design team. We will review some typical pitfalls, touch on available resources, and discuss a few potential solutions to consider.

8:25

4aAA2. Design considerations for mass timber in commercial buildings. David Manley (DLR Group, 6457 Frances St., Omaha, NE 68106, dmanley@dlrgroup.com)

Mass timber commercial buildings can significantly shift the building and design industry to a more sustainable path, but integrating architectural, engineering, and acoustic systems within this relatively new superstructure presents many challenges for designers. This talk will discuss DLR Group's integrated design approach and mass timber research including architectural design, structures, and acoustics.

8:45

4aAA3. Listening to the structure: Mass-timber construction at the Groton Hill Music Center. Carl P. Giegold (Threshold Acoust. LLC, 2622 Grant St., Evanston, IL 60201, cgiegold@thresholdacoustics.com), Laurie Kamper, Brandon Cudequest, and Matt Skarha (Threshold Acoust. LLC, Chicago, IL)

Historically, Western music spaces were built with massive masonry-bearing walls. While these designs offer structural and acoustical benefits, today's construction culture is conscious of more fluid forms and embodied carbon, which influences how consultants think about the design of spaces for music. The recently completed Groton Hill Music Center is unique for its use of mass timber's sculptural capabilities to shape performance space acoustic volumes forthrightly, with little additional finish treatment. The center features studio classrooms for students of all ages, an orchestral rehearsal space, a 300-seat recital hall for soloists and small ensembles, and a grand 1000-seat concert hall that opens to view for a 500-seat lawn audience. Structure and architecture are largely unified, with one's acoustic experience defined by the sculpted mass timber structure. The comparatively light-weight timber/concrete composite structure represents a significant and somewhat acoustically risky reduction in mass to provide an acoustically warm environment. This presentation will explore how the structure was optimized to provide stiffness and multi-scale diffusion, taking the ideals of classic concert hall design, and embracing contemporary construction technologies and acoustical analyses.

9:05

4aAA4. Mass timber acoustic design: Data informed design. Denis Blount (Acoust., Audioviusal, Arup, 1191 Second Ave., Ste. 400, Seattle, WA 98101, denis.blount@arup.com) and Ben Loshin (Acoust., Audioviusal, Arup, Seattle, WA)

Acoustic isolation is a fundamental challenge for mass timber construction, and solutions that work are critical for decarbonizing the built environment. While a broad range of acoustic laboratory test data exists for mass timber assemblies, field test data and flanking paths data are scarce. This talk will cover motivation for leveraging mass timber as a construction material (i.e. sustainability, embodied carbon, biophilic design, etc.), and inherent acoustic challenges. Additionally, it will address Arup's current practices and lessons learned from project work and research collaborations. The information presented will be drawn from project case studies, field test flanking data, and analytical studies aimed at developing the next generation of mass timber products and solutions. Case studies will focus on building types where mass timber has posed significant acoustics and vibration design challenges (e.g., healthcare, modular).

4a THU. AM

Analytical studies include novel inventions to solve vibration and room acoustic challenges. Field test data includes flanking field tests validated against numerical models.

9:25

4aAA5. Prediction of sound and vibration levels from heavy-hard impacts in a cross laminated timber—Concrete hybrid building. Joshua Brophy (Architectural Acoust., Acentech, 33 Moulton St., Cambridge, MA 02138, jbrophy@acentech.com) and Aedan Callaghan (Pliteq Inc., Toronto, ON, Canada)

Heavy-hard weight impacts in fitness spaces often result in elevated noise and vibration levels, which can lead to complaints in sensitive adjacencies. Previous works have demonstrated a method to predict the resulting noise and vibration levels in fitness space adjacencies. This method involves the analysis of an *in situ* transfer function in combination with laboratory force data and sound pressure level measurements from a shotput dropped on a calibrated reference rubber impact sheet. It was originally developed for use in concrete structures, which are typical for large multi-residential, commercial, and mixed-use buildings that commonly feature fitness amenity rooms or commercial gyms. The growing popularity of mass timber as a sustainable building material provides an opportunity to expand upon the previous research by examining the validity of the prediction method in another structural type. This paper serves as a case study of heavy-hard impacts in a residential hybrid mass timber structure, where weight drops from a fitness center at the top level of the concrete podium were disturbing apartment residents occupying the first CLT level directly above.

9:45–10:00 Break

10:00

4aAA6. Impact sound insulation performance of mass timber floors. Jianhui Zhou (School of Engineeing, Univ. of Northern BC, 499 George St., Wood Innovation and Design Ctr., Prince George, BC V2L 1R5, Canada, jianhui.zhou@unbc.ca) and Mohammad Hossein A. Jafari (School of Engineeing, Univ. of Northern BC, Prince George, BC, Canada)

Mass timber floors are being used increasingly in both mass timber and hybrid timber buildings due to its dry and rapid construction. Bare mass timber structural slabs have relatively low impact sound insulation performance. Though certain floating floor assemblies on mass timber slabs can provide adequate single number ratings, such assemblies are mainly effective in the middle to high frequency range. This paper will provide an overview of the research on the impact sound insulation perforamnce of mass timber floors with exposed ceiling and floating floor assembies conducted at the Universtiy of Northern British Columbia. The test data of bare mass timber floors, with continous floating concrete toppings, with raised discrete floating floor assemblies will be presented and compared. The effect of interlayer, concrete thickness and excitation sources on the single number ratings and frequency spectra will be discussed. Recommendations for future reseach will be given based on the discussion.

10:20

4aAA7. Vibroacoustic performance of a mass timber cassette floor through mock-up tests. Jianhui Zhou (School of Engineeing, Univ. of Northern BC, 499 George St., Wood Innovation and Design Ctr., Prince George, BC V2L 1R5, Canada, jianhui.zhou@unbc.ca), Chenyue Guo, Mohammad Hossein A. Jafari (School of Engineeing, Univ. of Northern BC, Prince George, BC, Canada), and Brant York (Intelligent City Inc., Vancouver, BC, Canada)

Though cross laminated timber slab floors are being used increasingly in mass timber construction, solid mass timber slab floors are limited to short-to-medium span applications. The sound insulation performance of solid mass timber slab floors is often achieved through assemblies on the floor surface as exposed wood ceilings are often prefered by achitects and occupants. A mass timber cassete floor system has been recently designed and tested for its structural, vibrational and acoustical performance. This paper reports the vibroacoustic performance of the proprietary floor system through mock-up building tests. In particular, the natural frequency and mode shapes of the floor under 200 Hz was measured by experimental modal tests. The effect of floating concrete topping on the dynamic properties of the floor was assessed by experimental modal testing. The impact sound insulation performance of the bare cassette floor was first measured, and then with floating concrete toppings. The airborne sound insulation performance of a partition wall was measured to reveal the effect of flanking sound casued by the exposed continous mass timber ceiling. Detailed results will be presented in the full paper.

10:40

4aAA8. Use of a micro acoustic chamber for rapid iterative design, teaching, and demonstration in the development of innovative lower-carbon mass timber assemblies. Mark Fretz (Architecture, Univ. of Oregon, 70 NW Couch St., 105A White Stag Bldg., Portland, OR 97209, mfretz@uoregon.edu), Jason Stenson (Architecture, Univ. of Oregon, Portland, OR), and Dale Northcutt (Architecture, Univ. of Oregon, Eugene, OR)

Mass timber construction offers a range of benefits to the design industry, including the potential for sustainability and reduced building embodied carbon, biophilia, speed of assembly, and prefabrication. However, these benefits can create acoustical challenges for sound isolation between spaces. Thus, the University of Oregon's Institute for Health in the Built Environment, in collaboration with the TallWood Design Institute, has developed a micro acoustic chamber as a conceptual testbed, educational resource, and demonstration tool for design professionals and students, to develop new and innovative mass timber acoustic assemblies, understand acoustic implications of material choices in composite form, and comparatively analyze the performance of acoustic detailing decisions more rapidly at a small scale. The chamber hosts specimens that are 813 mm $\times$ 813 mm and up to 300 mm thick, far smaller than a typical laboratory acoustic floor/ceiling assembly test chamber, creating known limitations; however, test samples can be cost and material efficient, allowing relative performance differences between assemblies to be evaluated quickly and easily for initial design feedback. This presentation will share how the use of a micro acoustic chamber has been integrated into design education, student, and industry pilot research to accelerate the development of mass timber acoustic solutions.

11:00

4aAA9. Cross laminated timber geometry and joinery implications on low-frequency sound transmission: FEA and Laser Doppler Vibrometry studies. Tomás I. Méndez Echenagucia (Univ. of Washington, 3950 University Way NE, Gould Hall, Seattle, WA 98105, tmendeze@uw.edu), Andrew A. Piacsek (Phys., Central Washington Univ., Ellensburg, WA), and Nicolaas B. Roozen (KU Leuven, Leuven, Belgium)

Cross Laminated Timber (CLT) slabs are becoming more widespread in North American construction in large part because of the environmental advantage of lower embodied carbon when compared to concrete and steel slabs. This advantage is significantly reduced by the use of high embodied carbon layers in the mass timber floor assemblies, such as concrete or gypsum screeds, that are used to reduce the direct sound transmission. This paper presents a study on the potential to reduce low-frequency sound transmission in cross-laminated timber floors by means of geometric stiffness. A laser doppler vibrometry (LDV) experiment on two different geometries is presented. One sample has a flat geometry, the other has a folded plate design. The folded plate sample is built with the same CLT panels used on the flat one, but is made out of 12 custom parts assembled into a domical shape. Both samples were installed on a 6×7 foot glulam frame, subjected to shaker excitations in the low frequencies on their top side, and the vibration on their bottom sides measured with the LDV. The resulting data is used to calibrate an FEA model, and subsequently study the sound transmission implications of the sample's boundary conditions and joinery.

11:20

4aAA10. Acoustic implications of thin mass timber panels. Evelyn Way (Maxxon Corp, 920 Hamel Rd., Hamel, MN 55340, eway@maxxon.com)

As mass timber construction becomes more widely adopted, there is an incentive to use thinner, lighter panels. Research sponsored by the American Wood Council and WoodWorks explores the relative performance of 5-ply CLT and 5-ply equivalent panels to 3-ply CLT and 3-ply equivalent panels both bare and with topping slabs. In addition to the comparison of thicker and thinner panels, assemblies using resilient mats and gypsum concrete topping slabs were also developed to meet the IBC residential code minimum performance. Results and acoustic analysis will be presented along with a discussion of the non-acoustic considerations of the move to thinner mass timber structural panels.

11:40

4aAA11. Calculation of sound transmission parameters for wood-frame and mass timber assemblies. Jason Smart (American Wood Council, 74 Driftwood Court, Heathsville, VA 22473, jsmart@awc.org)

While much of it is proprietary, a significant amount of the data from sound transmission tests performed on mass timber assemblies has been shared in the public domain. Previous efforts at using this data to develop empirical models for estimating single-number ratings of floor assemblies have been limited in scope and have not always yielded accurate results. In order to provide an additional means of demonstrating compliance with code requirements, the American Wood Council (AWC) has developed calculation-based analysis approaches for deriving STC and IIC ratings for assemblies constructed with wood-based framing and mass timber. As required by the International Building Code (IBC), these analysis procedures are based on comparisons of data from floor/ceiling assemblies having STC and IIC ratings as determined by the test procedures set forth in ASTM E90 and ASTM E492, respectively. The models presented in this report can be used to estimate STC and IIC ratings of: a.) light-frame floor/ceiling assemblies framed with either sawn lumber, wood I-joists, or wood trusses and b) mass timber floor/ceiling assemblies constructed with either cross-laminated timber (CLT), dowel-laminated timber (DLT), or veneer-based mass timber. Descriptions of the models are provided. Validation and application of the models are also discussed.

Session 4aAB

Animal Bioacoustics: Acoustic Ecology, Biological Soundscapes, and Animal Vocal Communication and Physiology II

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

Contributed Papers

8:30

4aAB1. An open-source acoustic detector for beluga whales, with evaluations in the Western Canadian Arctic. Fabio Frazao (Comput. Sci., Dalhousie Univ., 6050 University Ave. Halifax, NS B3H 1W5, Canada, fsfrazao@dal.ca), Marie-Ana Mikus, Valeria Vergara (Raincoast Conservation Foundation, Sidney, BC, Canada), William D. Halliday (Wildlife Conservation Society Canada, Whitehorse, YT, Canada), and Mike Dowd (Mathematics & Statistics, Dalhousie Univ., Halifax, NS, Canada)

Belugas (*Delphinapterus leucas*) face threats from various sources, including noise from oil and gas exploration, vessel traffic, and other human activities. Here we present the development of a deep learning-based acoustic detector to automatically detect the species, and measure its performance when applied to study sites in the western Canadian Arctic. We used over 20,000 individually annotated beluga vocalizations to train deep learning models in the binary task of classifying 3-second audio clips into containing beluga vocalizations or not. Approximately 7,000 annotated vocalizations were reserved for testing, and models were evaluated on their ability to correctly label audio clips of two lengths: 3 s and 60 s. The average F1 score (across 10 models) on 3 s clips was 0.82 with a standard deviation of 0.027, with the best model achieving 0.86. When applied to 60 s clips, the best model achieved an F1 score of 0.96. We used the trained classifier to build a detector that processes longer recordings and will make it available as an open-source tool.

8:45

4aAB2. Quantifying northern bottlenose and sperm whale acoustic behavioural responses to anthropogenic noise in Baffin Bay, Canada. Kimberly Franklin (Oceanogr., Dalhousie Univ., Dept. of Oceanogr. Dalhousie University, LSC Rm. 3631, 1355 Oxford St., Halifax, NS B3H 4R2, Canada, kimberly.j.franklin@dal.ca), William D. Halliday (Wildlife Conservation Society Canada, Whitehorse, YT, Canada), David R. Barclay, and Sarah Fortune (Oceanogr., Dalhousie Univ., Halifax, NS, Canada)

Marine mammals rely on their auditory system for a myriad of life functions (e.g., navigating, foraging, socializing) and consequently, are vulnerable to loud human activities (e.g., vessel traffic, fishing, military activities). These activities can impede communication, cause behavioural disturbances, and can even cause injuries. As the Arctic warms and sea ice coverage decreases, more opportunities for human activities are arising. How noise impacts the acoustic behaviour of Arctic marine mammals is unclear. In October 2022 and 2023 controlled noise exposure experiments were conducted using military sonar (source level of 176.4 dB re 1 μPa) on northern bottlenose and sperm whales in Baffin Bay while they were foraging around vessels actively fishing. Hydrophone suction-cup biologgers (DTAGs; $n = 5$, $\sim$72 cumulative hours) were used to capture vessel and sonar noises, and whale vocalizations before, during, and after the noise exposure periods. Using a click detector, focal whale clicks were identified and quantitatively compared to received noise levels. This information will then be used to determine noise thresholds for acoustical behavioural responses. These results will support risk-mitigation strategies for the Department of National Defence Canada and Fisheries and Oceans Canada, as well as address Inuit concerns about the effects of military sonar on marine mammals.

9:00

4aAB3. Narwhal and beluga seasonal presence in Barrow Strait and Eclipse Sound, Nunavut from 2013–2019. Eva C. Hidalgo Pla (Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093, echidalg@ucsd.edu), Kait Frasier, John Hildebrand, and Joshua Jones (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA)

We detected narwhal and beluga echolocation clicks in acoustic recordings between 2013 and 2019 in two Canadian Arctic locations: central Barrow Strait and at the eastern entrance of Eclipse Sound. We determined the presence of narwhals and belugas in the region in long-term acoustic recordings, using a semi-automated process for echolocation click detection and differentiation between the two monodontid species in large acoustic datasets. Additionally, we compare mean daily sea-ice concentrations in the recording areas with the acoustic presence of the species. This work provides insights into the trends in monodontid presence between sites and across the acoustic record. Our findings also reveal instances when echolocation clicks coincided with the presence of commercial ships transiting past the recording locations, exposing animals to underwater radiated noise. This study contributes to our understanding of the seasonal movements of belugas and narwhals in Nunavut waters, revealing information on multi-year trends in their habitat utilization, relationships with sea-ice, and exposure to underwater noise from ships.

9:15

4aAB4. Diel and tidal cycles in summer habitat use by St. Lawrence Estuary Belugas through a pluriannual passive acoustics monitoring network. Samuel Giard (Pêches et Océans Canada, 850 Rte. de la Mer, Mont-Joli, QC G5H 3Z4, Canada, samuel.giard@dfo-mpo.gc.ca), Yvan Simard (Pêches et Océans Canada, Mont-Joli, QC, Canada), Nathalie Roy (Mont-Joli, QC, Canada), Florian J. Aulanier (Fisheries and Oceans Canada, Mont-Joli, QC, Canada), and Véronique Lesage (Pêches et Océans Canada, Mont-Joli, QC, Canada)

A network of 10 passive acoustics monitoring stations is used to examine patterns of habitat use at diel and tidal timescales by St. Lawrence Estuary beluga during summer 2018 and 2019. An occurrence index of vocal activity within the preferred frequency band of communications for belugas is used as a proxy for presence at the stations. Diel and tidal patterns of activity and mean residency time are extracted from statistics of hourly occurrence timeseries at the 10 stations. Spatially, diel and tidal occurrence levels of beluga communication sounds show patterns of variation that differ among the stations, but tend to be locally stable from one summer to next. Mean residency time at the 9 Estuary stations vary between 4 to 15 hours and most have an occurrence maximum during early morning. The Saguenay fjord station shows a distinct profile, with a mean residency time of 30 hours and high level of activity at evening and night. This work underlines

the ability of passive acoustics, through continuous monitoring at high spatio-temporal resolution, to reveal the complexity of the habitat use by this confined marine mammal population and understand its responses to diel and tidal forcings of the ecosystem.

9:30

4aAB5. Cetacean presence and ambient sound level analysis in Bermuda: A comparison to historical records. Dawn Parry (K. Lisa Yang Ctr. for Conservation Bioacoustics, Cornell Lab of Ornithology, Cornell Univ., 159 Sapsucker Woods Rd., Ithaca, NY 14850, dmp295@cornell.edu), Jean-Pierre Rouja (The Nonsuch Expeditions & Station-B, St. David's, Bermuda), and Holger Klinck (K. Lisa Yang Ctr. for Conservation Bioacoustics, Cornell Lab of Ornithology, Cornell Univ., Ithaca, NY)

Marine Protected Areas (MPAs) are established to conserve nature while also preserving ecosystem services and cultural values. Many ongoing MPA discussions are centered on offshore islands, including Bermuda, a British Overseas Territory in the North Atlantic Ocean, which is currently assessing a management plan to protect 20% of its EEZ. Marine mammals are essential targets to monitor as part of effective MPA management because they are keystone species. In addition, as human activities raise ocean noise levels worldwide, monitoring ambient sound levels is essential to evaluating impacts on marine ecosystems. We collected nine months of passive acoustic data (at 250 kHz sampling rate and with 24-bit resolution) between December 2021 and September 2022, approximately 24 nm south of Bermuda, close to a location where previous studies were conducted in 1966 and 2013–2014. The cetacean presence and ambient sound levels derived from our dataset will be compared to these historical datasets. Our comparative analysis focused on humpback whales (*Megaptera novaeangliae*), fin whales (*Balaenoptera physalus*), minke whales (*Balaenoptera acutorostrata*), blue whales (*Balaenoptera musculus*), sei whales (*Balaenoptera borealis)*, and beaked whales (family Ziphiidae). Results will illustrate how Bermuda's offshore underwater soundscape has changed over nearly six decades.

9:45

4aAB6. Using sensors of opportunity to estimate the density of blue and fin whales. Kerri D. Seger (Appl. Ocean Sci., 2127 1/2 Stewart St., Santa Monica, CA 90404, kerri.seger.d@gmail.com), Danielle V. Harris (Univ. of St. Andrews, St Andrews, United Kingdom), and Jennifer Miksis-Olds (Univ. of New Hampshire, Durham, NH)

The "Combining global OBS and CTBTO recordings to estimate abundance and density of fin and blue whales", or CORTADO project, is using data from two bottom-sensor types to implement a suite of methods for estimating density of fin and blue whales. While previous studies have demonstrated the utility of Ocean Bottom Seismometers (OBS) and Comprehensive Test Ban Treaty Organization (CTBTO) hydroacoustic data, the techniques have not yet been developed to the point where they can be routinely applied by the wider research community. The primary CORTADO goal is to provide a set of software tools and training materials to expand access and usability of large, historic datasets. This talk will focus on using the CTBTO sensors for estimating density with a bearing method. This technique uses the bearing of and the signal-to-noise ratio for each detected call to estimate how many animals are producing a set of calls at any given time over the detection range of the sensor at all bearings. Applying this tool to several CTBTO sensors, we can compare the amount of blue and fin whales over time as well as spatially across the northern and southern coverage areas of the sensor array.

10:00–10:15 Break

10:15

4aAB7. Of bats and robots. Rolf Müller (Mech. Eng., Virginia Tech, ICTAS II, 1075 Life Sci. Cir, (Mail Code 0917), Blacksburg, VA 24061, rolf.mueller@vt.edu), Ben Westcott (Elec. and Comput. Eng., Virginia Tech, Blacksburg, VA), and Chi Nnoka (Mech. and Aerosp. Eng., Univ. at Buffalo, The State Univ. of New York, Buffalo, NY)

Some bat species feature joint adaptations to their biosonar sensing and flight systems that allow them to navigate and hunt in dense vegetation. Understanding how biosonar sensing and flight are connected in these species poses a challenge: Kinematics recordings of bat-flight can document the system outputs, but due to various limitations, especially on processing, analysis of bat flight has been limited to minimal numbers of flights and hence has had difficulty to capture the natural variability in flight maneuvers. On the input side of the bats' flight-control system, it is necessary to understand the stimulus ensemble that guides flight in dense vegetation. Navigation in these cases must be based on clutter echoes, i.e., signals consisting of many unresolved components. Since the biosonar inputs into a bat are difficult to record without heavy interference with the animals' behaviors. Hence, biomimetic sonar robots can be used to collect stimuli from natural environments and their recreations. With their superior abilities to find patterns, deep-learning offer an opportunity to cut through the complexity of the input and output data of bat flight control and elucidate how clutter echoes are interfaced with the high-dimensional flight kinematics of bats to create the animals' exceptional maneuvering capabilities.

10:30

4aAB8. A robotic platform for integrating biomimetic sonar and AI. Ben Westcott (Elec. and Comput. Eng., Virginia Tech, Blacksburg, VA 24061, bwestcott@vt.edu), Ibrahim M. Eshera (Elec. and Comput. Eng., Virginia Tech, Blacksburg, VA), and Rolf Müller (Mech. Eng., Virginia Tech, Blacksburg, VA)

To gain a first-hand view of the biosonar inputs that a bat receives when operating in a complex natural environment, a bioinspired sonar head to mimic the auditory periphery of the bats' biosonar system is currently under development. The sonar head seeks to replicate the static geometric complexity of the baffles that emit the biosonar pulses and receive the returning echoes in horseshoe bats. Furthermore, some of the dynamic deformations of these structures are mimicked using tension-driven soft-robotics actuated by a set of 12 motors (two for the noseleaf and five for the each pinna). The periphery has been interfaced with acoustic emission and reception systems as well as a modular system control system for the acoustic input and output as well as the motors. Besides low-level control, the sonar head has onboard computing resources that can support deep-learning inference in the loop with acoustic data acquisition and pulse generation. All functional components have been integrated into a custom-designed shell that allows operating the system in natural outdoor environments. A wireless user interface allows remote control over the system so that the tasks of carrying and orienting the sonar head and controlling its function can be shared between two individuals.

10:45

4aAB9. Using a biomimetic sonar bat robot to explore the sensory world of bats in the field and in the laboratory. Nicholas Rock (Phys., Univ. of Rhode Island, 2 Lippitt Rd., Kingston, RI 02881, nicholasrock@uri.edu) and Rolf Müller (Mech. Eng., Virginia Tech, Blacksburg, VA)

Bat species such as those belonging to the families of horseshoe and Old World leaf-nosed bats make their navigation decisions based on echoes that are often received from dense vegetation. To gain an understanding of how

these animals accomplish navigation based on such complicated clutter echoes, a biomimetic sonar system that replicates the auditory periphery of bats has been used as synthetic observer to probe the sensory stimulus ensemble of the bats in a flight tunnel as well as in the field. Both laboratory and field experiments were carried out in Brunei on the island of Borneo. For the flight tunnel, the biomimetic sonar robot has been used to record a large number of pulse and echo pairs in conjunction with an array consisting of 28 ultrasonic microphones. In this experiment, the sonar head was positioned at points spaced on a 3d grid that covers the tunnel's volume. This data set will be used to train a deep neural network to predict the inputs to the ears of a bat maneuvering in the tunnel. In the field experiments, the sonar head was carried off road in various tropical-forest habitats to record echo samples in conjunction with precise GPS location references.

11:00

4aAB10. An experimental array setup to study the integration of biosonar and maneuvering flight in bats. Chi Nnoka (Dept. of Mech. and Aerosp. Eng., Univ. at Buffalo, The State Univ. of New York, 804 Furnas Hall, Buffalo, NY 14260, chinnoka@buffalo.edu), Yihao Hu (Dept. of Elec. and Comput. Eng., Virginia Tech, Blacksburg, VA), Ulmar Grafe (Dept. of Biology, Universiti Brunei Darussalam, Bandar Seri Begawan, Brunei Darussalam), Javid Bayandor (Dept. of Mech. and Aerosp. Eng., Univ. at Buffalo, The State Univ. of New York, Buffalo, NY), and Rolf Müller (Dept. of Mech. Eng., Virginia Tech, Blacksburg, VA)

Understanding biosonar-based flight control in bats remains a challenge for fundamental bioacoustics and holds promise for engineered flight-control systems, especially for highly maneuverable drones. To accomplish this, detailed data on bat flight is required that captures the variability across flight situations, individuals, and species. Hence, a flight tunnel has been constructed that provides enough space for natural flight maneuvers. Reconstructing the detailed geometry of freely maneuvering bats requires capturing a flying bat from many different angles to ensure freedom from occlusions. Ideally, this would also be done without placing artificial markers on the bat. Hence, a flight tunnel has been instrumented with arrays of 50 high-speed video cameras and 28 ultrasonic microphones that were all integrated within a modular canvas setup. The optical properties of each camera have been determined from multiple images taken of a panel with a calibration grid seen from different orientations. The spatial relationship between the cameras was estimated based on a large set of images containing random points created with an LED. Reconstructing the 3D geometries of the bats from large numbers of high-speed video frames requires an automated method due to the problem's complexity. To this end, deep-learning methods are currently under development.

11:15

4aAB11. A robotic platform for mimicking the flapping flight of bats. Logan Kelley (Mech. Eng., Virginia Tech, 635 Prices Fork Rd., Blacksburg, VA 24061, loganckelley@vt.edu), Jungsoo Park, Alexander Leonessa, and Rolf Müller (Mech. Eng., Virginia Tech, Blacksburg, VA)

To understand how bats control their wing movements based on biosonar inputs, it is important to identify the key kinematic synergies that need to be replicated on a robotic prototype so its wingbeat patterns and aerodynamic capabilities matches its biological counterpart. Attempting to replicate the approximately 20 degrees of freedom in each bat wing is challenging since more mechanical degrees of freedom increase mass and reduce reliability. Our proposed approach, is based on recordings of the kinematics of maneuvering bats that are obtained from high-speed camera data. The first two synergistic actions to implemented in the robot were wing flapping and folding that work together to increase net lift generation. During straight flight, these synergies follow a fixed synchronized pattern allowing a single-actuator robot model to mimic key bat flight characteristics. By recording the robot prototype's flight kinematics with the same high-speed camera setup, we can compare the flight kinematics of bats and flapping-flight robots inspired by them, quantifying differences in complexity and performance. The goal is a bioinspired robotic platform enabling detailed study of bat flight biomechanics and control strategies. This will provide new biological insights and advance flapping-flight robot development by determining kinematic complexity required for bat-like performance.

Session 4aBAa

Biomedical Acoustics, Structural Acoustics and Vibration, Computational Acoustics, Acoustical Oceanography, and Physical Acoustics: Wave Propagation in Complex Media: From Theory to Applications I

Pierre Bélanger, Cochair
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Guillaume Haiat, Cochair
Multiscale modeling and simulation laboratory, CNRS, Laboratoire MSMS, Faculté des Sciences, UPEC, 61 avenue du gal de Gaulle, Creteil, 94010, France

Chair's Introduction—7:55

Invited Paper

8:00

4aBAa1. The acoustic trap theory and its application to lung ultrasound. Libertario Demi (Information Eng. and Comput. Sci., Univ. of Trento, Via sommarive 9, Trento, Italia 38123, Italy, libertario.demi@unitn.it)

In this lecture, the acoustic trap theory, as first introduced in 2016 in the context of lung ultrasound (DOI: 10.7863/ultra.15.08023), will be illustrated. Recent publications focusing on the design and validation of lung mimicking phantoms (DOI: 10.1038/s41598-017-13078-9, 10.1121/10.0001797, 10.1121/10.0007482) will be reviewed in the light of this theory. Moreover, its relevance to the monitoring and diagnosis of lung diseases by means of quantitative lung ultrasound spectroscopy will be discussed. To this end, results obtained within published (DOI: 10.1109/TUFFC.2020.3012289, 10.1121/10.0001723) and ongoing multicenter clinical studies, performed with dedicated multi-frequency imaging modalities implemented on open research scanners, will be presented and looked at through the lens of this theory.

Contributed Paper

8:20

4aBAa2. Poroelastic model of the lungs at low frequencies predicted by Biot's theory. Arife Uzundurukan (Ctr. de Recherche Acoustique-signal-humain, Université de Sherbrooke, Ctr. de Recherche acoustique-signal-humain, Université de Sherbrooke, Sherbrooke, QC J1K 2R1, Canada, arife.uzundurukan@usherbrooke.ca), Sébastien Poncet (Ctr. de Recherche Acoustique-signal-humain, Université de Sherbrooke, Sherbrooke, QC, Canada), Daria C. Boffito (Dept. of Chemical Eng., Polytechnique Montréal, Montréal, QC, Canada), and Philippe Micheau (Ctr. de Recherche Acoustique-signal-humain, Université de Sherbrooke, Sherbrooke, QC, Canada)

Biot's theory of poroelastic wave propagation inherently lends itself to elucidate the characteristics of a biphasic medium comprising solid and fluid components, such as biological tissues. One of the intricately complex biological tissues that remains poorly understood is the lungs since their properties diversify significantly through their pore geometries affected by inspiratory positive airway pressure (IPAP) and applied frequency range. One hypothesizes that the vibroacoustic behaviour of the lungs can be predicted by Biot's theory, as the nature of the lungs aligns with the principles of the theory at low frequencies. This study aims to analytically investigate the vibroacoustic behaviour of the lungs, considering 10 and 20-cm H_2O IPAP. Utilizing a fractional derivative formulation, one predicts the complex-valued shear wave speed, as well as the fast and slow compression wave speeds, for frequencies ranging from 5 to 100 Hz. A 3D digital thorax twin study using these predicted wave speeds, particularly at 28 Hz and 20 cm H_2O IPAP, is validated against experimental data from the literature. Consequently, applying Biot's theory provides a valuable framework for understanding the dynamic vibroacoustic behaviour of the lung tissues in response to varying IPAP and low frequencies.

Invited Paper

8:35

4aBAa3. 2D boundary-condition-free nonlinear inversion technique applied to optical shear vibration induced microelastography. Elijah E. W. Van Houten (Univ. of Sherbrooke, University of Sherbrooke, Sherbrooke, QC, Canada, Sherbrook, QC, Canada, elijah.van.houten@usherbrooke.ca), Sajad Ghazavi, Guillaume Fle, Hari s. Nair, Boris Chayer (Lab. of Biorheology and Medical Ultrason., Univ. of Montreal Hospital, Montreal, QC, Canada), Ruchi Goswami, Salvatore Girardo, Jochen Guck (Max Planck Inst. for the Sci. of Light & Max-Planck-Zentrum für Physik und Medizin, Erlangen, Germany), and Guy Cloutier (Lab. of Biorheology and Medical Ultrason., Univ. of Montreal Hospital, Montréal, QC, Canada)

Optical microelastography (OME) has emerged as a new technique for quantifying cellular mechanical properties. However, accurately reconstructing viscoelastic properties at the microscale level from noisy 2D displacement fields remains a challenge. This study introduces a 2D boundary-condition-free nonlinear inversion (2D-NoBC-NLI) approach, addressing challenges of interpreting noisy data and deducing full-field 3D displacements from 2D measurements. OME requires vibrating the cell and mapping the shear modulus based on wave-induced displacements within the cell. The shear modulus distribution is recovered via a coupled adjoint field NLI reconstruction to allow 2D-NoBC-NLI. Validation was conducted through numerical simulations at 36 kHz on a homogeneous sphere of 75 μm diameter and an assigned viscoelastic modulus, G*, of $800 + i150$ Pa. The same reconstruction approach was also applied to experimental data obtained from polyacrylamide (PAAm) microbeads of the same diameter. Results demonstrated relative differences from true simulated values of 0.7% and 45% for storage and loss moduli, respectively, with a coefficient of variation under 1% for homogeneous regions. When applying this method to PAAm microbeads, viscoelastic reconstructions showed the potential of OME under experimental conditions. These findings highlight the accuracy of 2D-No BC-NLI reconstruction in OME for precise microscale characterization and mapping of the viscoelastic cell structure.

Contributed Papers

8:55

4aBAa4. Signal detrending in elastographic wave data with large motion artifacts through a 2D Whittaker smoother. Rosen P. David (Physiol. & Biomedical Eng., Mayo Clinic College of Medicine and Sci., Medical Sci. Bldg., 321 3rd Ave SW, Rochester, MN 55902, Rosen.David@mayo.edu), Azra Alizad (Radiology, Mayo Clinic College of Medicine and Sci., Rocheter, MN), and Mostafa Fatemi (Physiol. & Biomedical Eng., Mayo Clinic College of Medicine and Sci., Rochester, MN)

A variety of advanced shear wave elastography techniques require the measurement of frequency-resolved phase velocity to properly characterize wave signals where factors such as tissue viscosity or guided wave effects are present. However, when applying these techniques in a dynamic environment where extrinsic motion cannot be limited, nonlinear and nonstationary trends can be expected in the motion signal used for phase velocity estimation. Because such extrinsic motion signals can contaminate a broad range of frequencies, they can produce significant errors in the phase velocity estimates. In this study, we propose a spatiotemporal detrending approach for elastography wave motion signals built from a 2D Whittaker smoother. This smoother has its origins in nonparametric regression and is well suited for fitting nonlinear curves of arbitrary shape. The resulting detrending filter was applied to Lamb wave signals from clinical ultrasound bladder vibrometry measurements. We found that the proposed detrending filter allowed for phase velocity estimates to be made in acquisitions that are otherwise be too corrupted by large motion. This works was supported by a grant from the NIH (DK131685)

9:10

4aBAa5. Surfacic characterization of soft tissues biomechanical properties using impact-based methods: A comparative study. Arthur Bouffandeau (CNRS, MSME UMR CNRS 8208, 61, Ave. du Général de Gaulle, Créteil 94010, France, arthur.bouffandeau@u-pec.fr), Giuseppe Rosi (Université Paris-Est Créteil, Creteil, France), Sabine Bensamoun (Université de technologie de Compiègne-UTC, Biomechanics and Bioengineering UMR CNRS 7338, Compiègne, France), Charles-Henri Flouzat-Lachaniette (INSERM U955, IMRB Université Paris-Est, Créteil, France), Vu-Hieu Nguyen (Université Paris-Est Créteil, Créteil, France), and Guillaume Haiat (CNRS, MSME UMR CNRS 8208, Créteil, France)

Various methods have been developed to assess the skin stiffness in order to assist the clinician for therapeutic monitoring or these diagnoses. The objective of this work was to compare the performances of a new acoustical characterization method based on impact analysis with those of two existing approaches, namely (i) a suction device, the Cutometer®, and (ii) a digital palpation tool, the MyotonPro®. This new impact analysis method is based on the analysis of the temporal variation of the force obtained during the impact of an instrumented hammer on a cylindrical punch placed in contact with a soft tissue mimicking phantom (Technogel®). The performances of the three aforementioned techniques (sensitivity and resolution) were assessed using homogeneous or bilayer structures with various thickness and rigidity, formed by different soft tissues mimicking gel pads. The impact analysis based method (IBAM) was two times better than the other two approaches in terms of reproducibility and sensitivity. The axial resolution of the IBAM was around 20 times better compared to the two other methods. The results open the way for the development of a cheap, non-invasive and objective method that could be used in the future in the cosmetic industry and in dermatology. This project has received funding from the projects OrthAncil (ANR-21-CE19-0035-03) and from the project OrthoMat (ANR-21-CE17-0004).

9:25

4aBAa6. The effect of cavitation-induced pressure on bubble cloud density in histotripsy. Adam Maxwell (Biomedical Eng. and Mech., Virginia Tech, Dept. of Urology, University of Washington, Seattle, WA 98195, amax38@u.washington.edu) and Eli Vlaisavljevich (Biomedical Eng. and Mech., Virginia Tech, Blacksburg, VA)

Tissue ablation in histotripsy is achieved by generating clouds of cavitation or boiling bubbles in the tissue. The rate of tissue ablation increases for clouds with a greater number density of nucleated bubbles. Therefore, controlling density may offer a mechanism to improve treatment efficacy. Experiments demonstrate several trends in density with ultrasound frequency, transducer f-number, and other variables. This presentation will describe one potential mechanism governing the density of cavitation bubbles nucleated during a focused ultrasound pulse. In particular, the rapid expansion of a cavitation bubble generates a partial cancellation of the incident pressure in the vicinity of the bubble, mitigating potential nucleation of other bubbles. We demonstrate this effect through a single-bubble numerical model, and further evaluate dependences of the pressure field with nuclei size, pressure amplitude, pulse frequency, and medium properties. The single-bubble model is further extended to consider the evaluate the effect of transducer focusing, and the combined pressure fields of multiple bubbles during cavitation nucleation. The predictions from simulation show good agreement with experimentally reported trends with frequency and transducer f-number, supporting the role of the mechanism in limiting the nucleation density in histotripsy.

9:40–9:55 Break

Invited Papers

9:55

4aBAa7. Ultrasound propagation in cancellous bone: Dependence on mechanical, material, and structural properties. Keith A. Wear (Ctr. for Devices and Radiological Health, Food and Drug Administration, Bldg 62 Rm. 2114, 10903 New Hampshire Ave., Silver Spring, MD 20993, keith.wear@fda.hhs.gov)

Critical skeletal sites for osteoporotic fractures include femur and vertebrae. However, many ultrasound devices designed for the management of osteoporosis instead target the calcaneus, which is far more accessible to ultrasound than femur or vertebrae. Calcaneus-based devices have been shown to be very effective for predicting osteoporotic fractures of the hip. MicroCT-based finite element models support targeting of calcaneus based on similar biomechanical structure-function relations in calcaneus compared with femoral neck, greater trochanter, proximal tibia, and vertebra. Finite element analysis can be used to relate these mechanical properties to ultrasonic properties such as broadband ultrasound attenuation, speed of sound, and broadband ultrasound backscatter. As an ultrasound beam propagates through cancellous bone, it loses energy to absorption, longitudinal-shear scattering, and longitudinal-longitudinal scattering. The relative roles of these mechanisms can be demonstrated in phantom models.

10:15

4aBAa8. Competing effects of ultrasound absorption and scattering in porous bone. Brett A. McCandless (Mech. and Aerosp. Eng., North Carolina State Univ., Raleigh, NC), Kay Raum (Charite Univ. Hospital, Berlin, Germany), and Marie Muller (MAE, North Carolina State Univ., 911 Oval Dr., Eng. Bldg. III, Raleigh, NC 27606, mmuller2@ncsu.edu)

The mechanisms and effects of ultrasonic attenuation in porous cortical bone are poorly understood, and it is necessary to better understand them to evaluate bone porosity noninvasively using ultrasound. A finite-difference time domain numerical study was conducted in which ultrasound propagation was simulated in human femur cross-sections obtained via scanning acoustic microscopy. The effect of absorption on overall attenuation was studied by varying the nominal absorption level attributed to the solid matrix. Ultrasound pulses were emitted with a central frequency of 8 MHz in through-transmission and backscattering configurations. From these data, the respective extinctions lengths due to overall attenuation, scattering, and absorption were obtained. Two regimes seem to exist depending on the nominal absorption value. At low absorption values, scattering dominates overall attenuation, scattering and absorption appear to have a synergistic effect on overall attenuation, and the diffusion constant decreases with increasing average pore diameter. At high absorption values, absorption dominates overall attenuation, the extinction length for scattering increases with pore diameter, and the diffusion constant increases with increasing average pore diameter. These regimes affect how ultrasound parameters, such as the extinction lengths due to scattering, absorption, and overall attenuation, should be used to evaluate the porosity of cortical bone.

10:35

4aBAa9. Electrical potentials in bone induced by ultrasound propagation. Mami Matsukawa (Doshisha Univ., 1–3, Tatara Miyakodani, Kyotanabe, Kyoto 610-0321, Japan, mmatsuka@mail.doshisha.ac.jp), Shouta Kitajima (Doshisha Univ., Kyotanabe, Japan), and Atsushi Hosokawa (Dept. of Elec. and Comput. Eng., National Inst. of Technol., Akashi College, Akashi, Japan)

Bone with complex geometry is hard, heterogeneous and anisotropic tissue, which makes ultrasound diagnosis very difficult. Therefore, the simulation of wave propagation is often performed to understand the complicated wave propagation in bone, where we find solid-liquid coexisting conditions. In addition to the diagnosis, one of the recent topics of bone studies are ultrasound fracture healing. The low intensity pulsed ultrasound (LIPUS) is popular as a fracture treatment technique in the field of orthopedic surgery, although how bones detect high frequency ultrasound is still under discussion. In this study, we focus on the weak piezoelectricity of bone as one of the key properties for the LIPUS treatment. According to the piezoelectric finite difference time-domain (PE-FDTD) method, we have investigated ultrasound propagation and generation of electrical potentials in bone. First, we verified the PE-FDTD method by comparing the simulated electric fields with the experimental data obtained by an ultrasound receiver using bone as the piezoelectric element. Next, we have tried to understand the wave propagation and generation in a real bone model (the radius of a 66-year-old woman). The generation of electrical potentials in the cancellous bone was also studied by simulation and experiments. The use of human bone to fabricate a model was permitted by the ethical committee at Doshisha University.

10:55

4aBAa10. Fast and accurate transcranial ultrasound simulation using the asymptotic model of the Civa Healthcare platform. Sylvain Chatillon (LIST, CEA, Institut CEA LIST CEA Saclay, Bât. Digiteo - 565, Gif sur yvette 91191, France, sylvain.chatillon@cea.fr), Andrew Drainville (LabTAU INSERM 1032, Lyon, France), John Snell, David Moore, Frederic Padilla (Focused Ultrasound Foundation, Charlottesville, VA), and Cyril Lafon (LabTAU INSERM 1032, Lyon, France)

To ensure its efficacy and safety, transcranial ultrasound therapy treatment planning requires accurate pressure field simulations and phase law corrections. Despite their long computation time and high memory usage, full numerical methods are often used since they are considered more accurate than semi-analytical methods. This work will present the so-called “pencil method“, a fast asymptotic model embedded in the CIVA HealthCare simulation platform. It allows computation in harmonic and impulse mode and the consideration of complex configurations, including solid obstacles, considering, at each interface, refractions and reflections with or without mode conversion of the acoustic field. This model was successfully compared to a recent collaborative work by Aubry *et al*. that presented a set of numerical benchmarks for transcranial propagation, to allow comparisons between various modeling tools. It was used to investigate the influence of parametric variation of skull material properties on the quality of acoustic focusing through the human skull. Its ability to predict the thermal rise at the intracranial target was validated against experimental data obtained *ex-vivo* through human skulls. Finally, works in progress will be shared about its connection to the open-source Kranion software developed at the FUS Foundation to facilitate comparison between clinical and simulated data.

11:15

4aBAa11. Validation of mSOUND using a fully heterogeneous skull model. Jeff J. Bell (Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, jjb7481@psu.edu), Lu Xu, Hong Chen (Washington Univ. in St.Louis, St Louis, MO), and Yun Jing (Acoust., Penn State Univ., State College, PA)

Transcranial ultrasound has found an increasing number of applications in recent years, including the treatment of neurological conditions through thermal ablation and neuromodulation. Ensuring the success and safety of such treatments necessitates precise numerical simulations of transcranial ultrasound, a pivotal aspect of treatment planning involving phase correction. Addressing this demand, an open-source wave solver named mSOUND (https://m-sound.github.io/mSOUND/home) was developed specifically for modeling focused ultrasound in heterogeneous media. A recent intercomparison study (J. Acoust. Soc. Am. 152, 1003–1019, 2022) scrutinized mSOUND alongside other wave solvers like k-Wave, demonstrating its accuracy in modeling wave propagation through a homogeneous skull. This study extends the assessment to evaluate mSOUND's accuracy in modeling wave propagation through a fully heterogeneous skull, utilizing CT images of an *ex vivo* human skull. The obtained results are systematically compared with those from k-Wave, revealing a high level of agreement.

11:30

4aBAa12. Simulation-corrected focusing to the vertebral canal. David Martin (Physical Sci., Sunnybrook Res. Inst., 2075 Bayview Ave., Toronto, ON M4N 3M5, Canada, dave.martin@mail.utoronto.ca), Rui Xu (Medical Phys. and Biomedical Eng., Univ. College London, London, United Kingdom), and Meaghan O'Reilly (Physical Sci., Sunnybrook Res. Inst., Toronto, ON, Canada)

Phased arrays have long been used to deliver focused ultrasound to the brain, but applications in the spinal cord remain comparatively unexplored. This is largely due to the aberrating effect of vertebral bone on the incoming wavefront, where variable density and complex geometry distort the pressure field in the canal. For controlled focusing, phase and amplitude corrections must therefore be calculated in advance. Here, we validate a previously-developed ray acoustics model for transvertebral focusing with a bilateral spine-specific array. Benchtop trials were conducted with *ex vivo* human vertebrae and ray acoustics beamforming was compared to a geometric baseline and hydrophone-corrected gold standard. Planar shift in the 90% contour was evaluated via raster scans of the sagittal and coronal planes. Ray acoustics correction reduced mean sagittal shift from 1.76 ± 0.79 mm to 1.63 ± 0.86 mm and reduced mean coronal shift from 0.99 ± 0.54 mm to 0.76 ± 0.63 mm, while hydrophone correction produced mean sagittal and coronal shifts of 1.40 ± 0.83 mm and 0.53 ± 0.47 mm, respectively. Large variance in simulation-corrected results is hypothesized to stem from nonuniform attenuation and intervertebral acoustic windows, a possibility that will be explored in future *in silico* and benchtop work.

Session 4aBAb

Biomedical Acoustics and Physical Acoustics: Droplets Strike Back I

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Chair's Introduction—9:00

Invited Papers

9:05

4aBAb1. Acoustic vaporization of perfluorohexane droplets is induced by heterogeneous nucleation at 1.1 MHz. Rashmi Ramesh, Chloë Thimonier (Laboratoire d'Imagerie Biomédicale, Sorbonne Université, Paris, France), Stéphane Desgranges (Avignon Université, Avignon, France), Vincent Faugeras (Ecole Normale Supérieure, Paris, France), François Coulouvrat (Sorbonne Université, Paris, France), Justine Laurent (ESPCI, Paris, France), Guillaume Marrelec (Laboratoire d'Imagerie Biomédicale, Sorbonne Université, Paris, France), Christiane Contino-Pépin (Avignon Université, Avignon, France), Wladimir Urbach, Christophe Tribet (Ecole Normale Supérieure, Paris, France), and Nicolas TAULIER (Laboratoire d'Imagerie Biomédicale, Sorbonne Université, 15 rue de l'école de médecine, Paris 75006, France, nicolas.taulier@sorbonne-universite.fr)

We investigated the ADV of perfluorohexane (PFH) nano and micro-droplets at a frequency of 1.1 MHz, at conditions where there is no superharmonic focusing. Our experiments were performed on suspensions of droplets in glycerol to avoid sedimentation. The ADV pressure threshold was defined as the pressure for which there is half a chance to observe a vaporization event. Our experiments showed that the threshold depends on the number of insonified droplets and the data were fitted by a statistical model, that is independant on the mechanism leading to ADV. The fit allows the derivation of the threshold in the case of a single insonified droplet. We further observed that the value of the ADV pressure threshold of a single PFH droplet decreases as its radius increases. In addition, at a constant droplet radius, the threshold did not vary when the PFH volume inside the droplet decreases, thanks to the encapsulation of an increasing number of water droplets. These results are incompatible with a model of homogeneous nucleation (where a nucleus can appear anywhere into the PFH volume). However, we developed a heterogeneous nucleation model, where the nucleus appears at the inner droplet surface, that successfully predicts our experimental ADV results.

9:30

4aBAb2. Predicting the spontaneous vaporisation of superheated perfluorocarbon droplets. Luca Bau, QIANG WU (Univ. of Oxford, Oxford, United Kingdom), Nicholas Ovenden (Mathematics, Univ. College London, London, United Kingdom), and Eleanor P. Stride (Univ. of Oxford, Inst. of Biomedical Eng., Oxford OX3 7DQ, United Kingdom, eleanor.stride@eng.ox.ac.uk)

Superheated perfluorocarbon droplets have been widely explored as agents for ultrasound imaging and therapy, as well as for other applications such as radiation dosimetry. Submicrometre, or "nano" droplets offer a number of potential advantages over microbubbles, e.g., longer circulation half-lives, higher surface area-to-volume ratio and the ability to perfuse the microvasculature more easily. A key challenge in the use of nanodroplets, however, is the need to avoid spontaneous vaporisation whilst keeping the energy required for acoustic activation within the range of pressures that can be used safely in humans. This is especially important for imaging applications. Perfluoropropane (C_3F_8) microbubbles can be condensed to form liquid nanodroplets that offer a good trade-off between thermal stability and acoustic vaporisation threshold. Anecdotal reports, however, suggest that C_3F_8 droplets can spontaneously vaporise, and may therefore pose a potential safety risk, especially if bubble coalescence occurs. The aim of this study was to build on recent theoretical models of droplet vaporisation and investigate the probability of vaporisation as a function of temperature and interfacial tension. The results are compared with experimental measurements of vaporisation rates for different droplet formulations.

4a THU. AM

Contributed Papers

9:55

4aBAb3. Non-equilibrium activation dynamics of superheated phase-change contrast agents. Nicholas Ovenden (Mathematics, Univ. College London, Dept of Mathematics, University College London, London WC1E 6BT, United Kingdom, n.ovenden@ucl.ac.uk), Luca Bau, and Eleanor P. Stride (Univ. of Oxford, Oxford, United Kingdom)

Phase-change contrast agents, such as superheated perfluorocarbon droplets, potentially offer exciting new opportunities in diagnostic and therapeutic applications of ultrasound. An important advantage of droplets over, e.g., microbubble agents is that they can be "activated" (vaporised) by a reduction in ambient pressure, initiated by an externally applied ultrasound pulse. Controlling the timing and location of droplet activation, however, is vitally important to avoid safety concerns. Current theoretical models for droplet vaporisation assume that a critical stable bubble embryo forms that is in thermal and mechanical equilibrium. Model predictions are, however, inconsistent with some experimental observations. The aim of this study is to develop a non-equilibrium model for droplet activation wherein initial bubble growth rate and vapour pressure are dependent upon the bubble nucleus size. We demonstrate how incorporating nonequilibrium effects into theoretical models of droplet vaporisation dynamics enables better agreement of the droplet expansion with experimental data and thus provide greater insight into control of the activation process.

10:10

4aBAb4. Microcavitation dynamics of vaporized perfluorocarbon nanodroplets captured with an acoustical camera. Mark Burgess (Dept. of Medical Phys., Memorial Sloan Kettering Cancer Ctr., 321 E 61st St., New York, NY 10065, burgesm1@mskcc.org) and Jeffrey A. Ketterling (Radiology, Weill Cornell Medicine, New York, NY)

Perfluorocarbon nanodroplets are on-demand cavitation nuclei for numerous biomedical applications. These nanodroplets remain in a metastable state until an ultrasound pulse of sufficient negative pressure initiates conversion from a liquid droplet into a gaseous micro- or nano-bubble in a process termed acoustic droplet vaporization (ADV). Extensive research has gone into understanding their interaction with ultrasound at varying acoustic and ambient conditions. Recent studies have detected intra- and post-excitation collapse emissions indicative of inertial cavitation. This study aims to further understand the post-ADV dynamics of perfluorocarbon nanodroplets with an acoustical camera technique. Nanodroplets were activated with a low-frequency ultrasound pulse (5 MHz) while being insonified with a high-frequency (35 MHz) probing wave. The side scattered emissions from the probing wave were captured with another matched high-frequency transducer. The amplitude modulation of these emissions is proportional to the radial strain; thus, a relative radius-time curve can be extracted from the scattered signal after filtering and enveloping. Results show that this technique can capture the radial growth and collapse curves of these newly formed bubbles and aligns well with intra- and post-excitation emissions indicative of inertial cavitation. This approach could be useful to understand post-ADV dynamics of various droplet formulations.

10:25–10:50 Break

Invited Paper

10:50

4aBAb5. Peak positive pressure matters in acoustic droplet vaporization. Samuele Fiorini, Anunay Prasanna, Gazendra Shakya (ETH Zurich, Zurich, Switzerland), and Outi Supponen (ETH Zurich, Sonneggstrasse 3, ETH Zurich - D-MAVT - IFD - ML H31, Zurich 8092, Switzerland, outis@ethz.ch)

We demonstrate the significance of the positive pressure component of an ultrasound wave in acoustic droplet vaporization. Theory and acoustic simulations reveal that the distorted compression part of the incoming wave, which includes a broad spectrum of frequencies above the fundamental, presents a pronounced shift in phase when focusing within the droplet and crossing the focal point. The extent of this so-called Gouy phase shift is sufficient to change the sign of the compression phase of the wave, actively creating a tension region in the droplet bulk in the same location where we experimentally observe vapor nucleation. A sign reversal of the rarefaction component of the ultrasound wave is, on the other hand, not expected. The extent of distortion of the incoming ultrasound wave can influence the occurrence of droplet vaporization, even at constant peak negative pressure, which may partially explain the broad range of vaporization threshold values reported in literature. The results suggest that vaporization can be achieved exploiting ultrasound waves with high peak positive pressures and reduced peak negative pressures.

Contributed Papers

11:15

4aBAb6. Gold nanoparticle-coating reduces the acoustic pressure threshold for nanodroplet vaporization. Nishita Mistry, Ruchika Dhawan (Dept. of Elec. Eng., Indian Inst. of Technol. Gandhinagar, Gandhinagar, Gujarat, India), and Karla Patricia Mercado-Shekhar (Dept. of Biological Sci. and Eng., Indian Inst. of Technol. Gandhinagar, Academic Block 6/207, Indian Inst. of Technol. Gandhinagar, Gandhinagar, Gujarat 382355, India, karlamshekhar@iitgn.ac.in)

Nanodroplets with a low-boiling-point perfluorocarbon (PFC) may vaporize spontaneously at physiological temperatures. Using a high-boiling-point PFC, e.g., perfluorohexane (PFH), can enable ultrasound-triggered vaporization for applications in imaging and drug delivery. However, PFH requires relatively high ultrasound (US) pressures for vaporization compared to low-boiling-point PFCs, making its use challenging. We investigated the feasibility of reducing the vaporization threshold by gold-coating lipid-encapsulated PFH nanodroplets (Au-PFH-NDs). We synthesized Au-PFH-NDs with 200 nm mean particle size and 5.12×10^{-4} pg mass of gold-coating per nanodroplet. B-mode images of the emulsion perfused in a flow phantom were used to determine the pressure threshold for ND vaporization upon exposure to 2 MHz focused US (f-number of 1.27, 0.5% duty cycle). The pressure threshold for Au-PFH-NDs (3.97 ± 0.63 MPa) was significantly lower ($p < 0.05$) than that of NDs without gold-coating (7.07 ± 0.02 MPa), indicating that gold-coating reduced the vaporization

pressure threshold. In addition, the pressure threshold of the Au-PFH-NDs was not significantly different ($p > 0.05$) from that of perfluoropentane (PFP) NDs (4.16 ± 0.01 MPa). These results suggest that the Au-PFH-NDs can be vaporized at similar pressures as PFP NDs, but are more stable at physiological temperatures. These findings are the first step towards employing gold-coated PFC nanodroplets with a lower vaporization pressure threshold for multimodal imaging and drug delivery.

11:30

4aBAb7. Droplets for acoustic vaporization: Formulations, properties and applications. Gaio Paradossi (Dept. of Chemical Sci. and Technologies, Univ. of Tor Vergata, Dept. of Chemica Sci. and Technologies, Via della Ricerca Scientifica 1, Rome, RM 00133, Italy, gaio.paradossi@uniroma2.it) and Fabio Domenici (Dept. of Chemical Sci. and Technologies, Univ. of Tor Vergata, Rome, RM, Italy)

The Acoustic Droplet Vaporization (ADV) (1), a phase-change of the droplets core from liquid to vapor phase upon ultrasound irradiation, burst renewed interest in droplet emulsions, a traditional topic of colloidal science, opening up innovative applications in biomedicine. Droplets undergoing ADV share similar liquid cores, typically perfluorocarbons (PFCs), however, the nature of the shells can be polymeric (2) or lipidic (3, 4). Such difference imparts to phase-change droplets diverse acoustic and mechanical behaviors (5). In this contribution we present some results concerning a general strategy for the formulation of polymer or lipid shelled submicron droplets. Other key points for the use of ADV-responsive in biomedical applications will be addressed. Recently we have extended the concepts of ADV to radiation responsive droplets for dosimetry in cancer treatment with hadronic radiation. Results on this activity will be also reported (6). References: (1) *J. Acoust. Soc. Am.* (2004) **116** (1): 272–281; (2) *Chem. Commun.*, 2013, 49, 5763; (3) *JoVE*, *169* (2021); (4) *Langmuir* (2019), 35, 10116–10127; (5) *Phys. Chem. Chem. Phys.*, 2016, 18, 8378; (6) *COCIS*, *49*, 118–132 (2020).

THURSDAY MORNING, 16 MAY 2024 — ROOM 211, 9:00 A.M. TO 10:15 A.M.

Session 4aCA

Computational Acoustics, Structural Acoustics and Vibration and Engineering Acoustics: Validation and Verification

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

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

Chair's Introduction—9:00

Invited Papers

9:05

4aCA1. Validation and verification of models for setting SONAR system requirements. Jennifer Cooper (Johns Hopkins Univ. Appl. Phys. Lab., 11100 Johns Hopkins Rd., Mailstop 8-220, Laurel, MD 20723, jennifer.cooper@jhuapl.edu), Jean Dougherty, and Marina Johnson (Johns Hopkins Univ. Appl. Phys. Lab., Laurel, MD)

When determining requirements for a SONAR system as a whole, typically very high-level requirements such as number of detections and total false alarms allowed per unit time must be converted into specific requirements for system components. These requirements may include array requirements like hydrophone count and position. Validation of probability of detection via sea test is fraught with complications, not least of which is environmental uncertainty. As a precursor, it is generally advisable to approach and attempt to validate models for each term in the SONAR equation one at a time. The approaches to validation vary depending on the term and can range in complexity from comparison to analytical solutions, to comparison with alternate available models, up through data collections at sea. Each of these approaches comes with its own challenges and limitations, which are discussed.

9:25

4aCA2. Data and information management for acoustics research. Amanda Hanford (Penn State Univ., PO Box 30, State College, PA 16802, ald227@psu.edu), Tyler P. Dare (Penn State Univ., State College, PA), Keith Rice (Penn State Univ., University Park, PA), and Andrew S. Wixom (Penn State Univ., State College, PA)

A critical part of verification and validation in academic research includes the important consideration of data and information management. As researchers grapple with escalating volumes of data, effective data management becomes imperative for optimizing operational processes and ensuring reproducibility and archivability. Data management involves the organization, storage, and retrieval of information to support research advancements and strengthen the foundation of decision making and scientific knowledge. Key components also include data operations, which involves the orchestration of data workflows, and data quality management, which focuses on maintaining accurate and reliable data. This talk explores the multifaceted aspects of data management, emphasizing its significance in ensuring data quality, accuracy and accessibility.

Contributed Papers

9:45

4aCA3. Predicting sound transmission loss in particle-reinforced polymer composites using machine learning: A comparative study with experimental and theoretical results. Jonty Mago (Automotive Health Monitoring Laboratory/Ctr. for Automotive Res. and Tribology, Indian Inst. of Technol. Delhi, Hauz Khas, New Delhi, Delhi 110016, India, jontymago@gmail.com), Sunali Jaish (Automotive Health Monitoring Laboratory/Ctr. for Automotive Res. and Tribology, Indian Inst. of Technol. Delhi, New Delhi, Delhi, India), Ashutosh Negi (School of Interdisciplinary Res., Indian Inst. of Technol. Delhi, New Delhi, Delhi, India), J. S. Bolton (Ray W. Herrick Laboratories/School of Mech. Eng., Purdue Univ., USA, West Lafayette, IN), and S Fatima (Automotive Health Monitoring Laboratory/Ctr. for Automotive Res. and Tribology, Indian Inst. of Technol. Delhi, New Delhi, Delhi, India)

In this study, the use of machine learning (ML) models to predict sound transmission loss (STL) in particle-reinforced polymer composites has been examined. The work included extensive literature research and data extraction for training various ML models, focusing on their effectiveness in accurately predicting STL, which is crucial for evaluating acoustic performance. The method involves processing data and applying it to different ML algorithms, with the models calibrated and tested for reliability. A key aspect is comparing these models' predictions with actual experimental results and theoretical models based on the mass law in acoustics. The findings reveal the potential and limitations of ML in materials science, showing their accuracy in predicting STL and comparing them with traditional theories. This research advances the use of data-driven methods in developing and assessing acoustic materials, significantly impacting materials science and machine learning.

10:00

4aCA4. Sound propagation modeling in forests using the transmission line matrix method: Comparison with experimental *in situ* measurements. Quentin Goestchel (School of Oceanogr., Univ. of Washington, 1503 NE Boat St., Seattle, WA 98115, qgoestch@uw.edu), Gwenaël Guillaume, Ecotière David (UMRAE, CEREMA, Univ Gustave Eiffel, F-67035 Strasbourg, France, Strasbourg, France), and Gauvreau Benoit (UMRAE, Univ Gustave Eiffel, CEREMA, F-44344 Bouguenais, France, Nantes, France)

We present efforts to adapt the time-domain Transmission Line Matrix (TLM) method for modeling sound propagation in forests. It is relevant to professionals and researchers in environmental acoustics and to those interested in studying sound propagation in outdoor environments using numerical modeling techniques. The study aims to demonstrate the applicability of the method in complex media where sound waves undergo multiple reflections combined with ground effects during their propagation from a sound source to a receiver point. The TLM method is used to numerically model sound propagation in a 3D forest geometry generated based on a real tree distribution case. Data from an experimental campaign conducted in the rainforest of French Guyana are used as a reference. The results show an encouraging comparison between them and the *in situ* measurements, even if the hypotheses made to adapt the input data in the model were strong. We find that more measurement points and more specifications (ground impedance measurements, canopy density, etc.) about the experimental site are needed for further simulations, even if obtaining them presents technical difficulties.

10:15

4aCA5. Predicting excess attenuation: An investigation of turbulence in a littoral environment. Andrea Vecchiotti (Dept. of Eng., East Carolina Univ., Washington, DC), Diego Turo (Mech. Eng., The Catholic Univ. of America, 620 Michigan Ave., N.E., Washington, Washington, DC 20064, turo@cua.edu), Matthew Stengrim, Jeff Foeller, Teresa J. Ryan (Dept. of Eng., East Carolina Univ., Greenville, NC), and Joseph Vignola (Mech. Eng., The Catholic Univ. of America, Washington, DC)

This work presents an effort to effectively represent turbulence in a numerical model of atmospheric sound propagation in a near-shore environment. Model results are compared with a set of atmospheric transmission loss measurements made in a pitch-catch configuration with a stationary source 500 m from shore. High resolution temperature profiling and scanning Doppler LIDAR wind profiling measurements in the source-to-receiver direction are concurrent with the acoustic transmission loss measurements at the shoreline. These meteorological measurements inform the input parameters used in the parabolic equation. The numerical predictions are compared to the measured transmission loss values to evaluate model performance with a variety of different approaches to implement turbulence parameters.

Session 4aID

Interdisciplinary, Student Council, Physical Acoustics, and Education in Acoustics: Graduate Studies in Acoustics Poster Session

Pratik A. Ambekar, Cochair
Univ. of Washington, 1013 NE 40th St, Seattle, WA 98105

John A. Case, Cochair
Graduate Program in Acoustics, Penn State, 201 Applied Science Building, University Park, PA 16802

Marissa Garcia, Cochair
Natural Resources and the Environment, Cornell University, 159 Sapsucker Woods Road, Cornell Lab of Ornithology - K. Lisa Yang Center for Conservation Bioacoustics, Ithaca, NY 14850

All posters will be on display and all authors at their posters from 10:00 a.m. to 11:00 a.m.

Contributed Papers

4aID1. Acoustics at UMass Dartmouth. David A. Brown (Elec. and Comput. Eng., UMass Dartmouth, 151 Martine St., UMass Dartmouth, Fall River, MA 02723, dbAcousticsdb@gmail.com), John R. Buck, and Paul J. Gendron (Elec. and Comput. Eng., UMass Dartmouth, Dartmouth, MA)

The University of Massachusetts Dartmouth has a long history spanning 5 decades of graduate courses offerings and research in underwater acoustics, transduction and signal processing leading to M.S. and Ph.D. degrees in Electrical Engineering. Collaborations between Marine Science, Physics, Mechanical, Bioengineering, and Electrical Engineering departments offer many interdisciplinary opportunities in Acoustics. Courses include Fundamentals of Acoustics, Vibrations, Underwater Acoustics, Electroacoustic Transducers, Medical Ultrasonics, Sonar, Digital Signal Processing, Array Processing, Random Signals, Information Theory, Communications, Detection and Estimation. Many unique facilities including an Underwater Acoustic Test Tank, Open-Ocean water access, and Unmanned Underwater Vehicles support both undergraduate and graduate projects. Research focuses include transducers and transduction science, materials characterization, calibration, array and sonar signal processing, animal bioacoustics, communications, detection and estimation, active and passive sonar funded by Office of Naval Research and Industry.

4aID2. Graduate acoustics at Brigham Young University. Kent L. Gee, Micah Shepherd, Brian E. Anderson (Phys. & Astronomy, Brigham Young Univ., Provo, UT), Tracianne B. Neilsen (Phys. & Astronomy, Brigham Young Univ., N251 ESC, Provo, UT 84602, tbn@byu.edu), Matthew S. Allen, and Jonathan D. Blotter (Mech. Eng., Brigham Young Univ., Provo, UT)

Graduate studies in acoustics at BYU prepare students for industry, research, and academia by complementing in-depth coursework with publishable research. Coursework provides students with a foundation in acoustical principles, practices and measurement skills, including a experimental techniques and technical writing. Labs across the curriculum cover calibration, directivity, scattering, absorption, laser Doppler vibrometry, experimental methods for dynamic structures, lumped-element mechanical systems, equivalent circuit modeling, arrays, filters, room acoustics, active noise control, and near-field acoustical holography. Recent thesis and dissertation topics include active noise control, directivity, room acoustics, energy-based acoustics, time reversal, nondestructive evaluation, vibration and acoustics of aerospace vehicles, biomedical applications, flow-based acoustics, voice production, aeroacoustics, sound propagation modeling, nonlinear propagation, high-amplitude noise analyses, machine and deep learning applied to ambient noise level prediction, crowd noise interpretation, and underwater acoustic source localization, and ocean environment classification. Graduate students are expected to present research at professional meetings and publish in peer-reviewed acoustics journals. Graduate students often serve as peer mentors to undergraduate students on related projects and may participate in field experiments to gain additional experience. @BYUAcoustics

4aID3. The Graduate Program in Acoustics at Penn State. Andrew Barnard (Graduate Program in Acoust., Penn State, 201C Appl. Sci. Bldg., University Park, PA 16802, barnard@psu.edu) and Daniel A. Russell (Graduate Program in Acoust., Penn State Univ., University Park, PA)

The Graduate Program in Acoustics at Penn State offers graduate degrees (M.Eng., M.S., Ph.D.) in Acoustics, with courses and research opportunities in a wide variety of subfields. Our 820 alumni are employed around the world in military and government labs, academic institutions, consulting firms, and consumer audio and related industries. Our 40 + faculty from several disciplines conduct research and teach courses in structural acoustics, nonlinear acoustics, architectural acoustics, signal processing, aeroacoustics, biomedical ultrasound, transducers, computational acoustics, noise and vibration control, acoustic metamaterials, psychoacoustics, and underwater acoustics. Course offerings include fundamentals of acoustics and vibration, electroacoustic transducers, signal processing, acoustics in fluid media, sound and structure interaction, digital signal processing, experimental techniques, acoustic measurements and data analysis, ocean acoustics, architectural acoustics, noise control engineering, nonlinear acoustics, outdoor sound propagation, computational acoustics, biomedical ultrasound, flow induced noise, spatial sound and three-dimensional audio, and the acoustics of musical instruments. Distance education students pursuing the M.Eng. degree join resident students in a hybrid classroom environment. This poster highlights faculty research areas, laboratory facilities, student demographics, successful graduates, and recent enrollment and employment trends.

Session 4aMU

Musical Acoustics: General Topics in Musical Acoustics II

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

Contributed Papers

9:00

4aMU1. The applications of dynamic time warping in the source separation of percussive sounds. Christopher Grabow (Penn State, 205 Hallowell Bldg, State College, PA 16802, cng5270@psu.edu) and Tyler P. Dare (Penn State, State College, PA)

Music source separation (MSS) is the process of splitting various components of a musical piece into individual tracks. This process combines the fields of acoustics and machine learning to extract useful data from music, which assists in a variety of music information retrieval tasks. In the past decade, many methods have been employed to perform MSS with varying levels of success. This research explores the use of dynamic time warping (DTW) for MSS tasks in the time domain. DTW is an algorithm that performs a temporal alignment of two time series to measure their similarity. It is unique in that the algorithm will minimize the Euclidean distance between the two sequences by stretching or compressing them to optimize similarity. This makes DTW a distinctive method for MSS, as it operates entirely in the time domain and classifies sounds without the interference of time warping. The research performed focuses only on the separation of transient, percussive sounds. Measurements taken with a drum kit and a selection of digital drum sounds served as the foundation for tests of the algorithm. The results of this research illustrate the potential of DTW in time domain MSS applications.

9:15

4aMU2. An investigation into the directivity, spectra, and modes of a Tambourine. Emma Todd (Brigham Young Univ., 5 Heritage Halls #2204, Provo, UT 84602, et444@byu.edu), Hanna Pavill, Jacob B. Hales, Matt Coleman, and Micah Shepherd (Brigham Young Univ., Provo, UT)

As an instrument, the tambourine has been used by various cultures for thousands of years. However, despite this rich history, there remains little literature that investigates the acoustical properties of the tambourine itself and an even greater lack in research that moves beyond discussing the instrument's membrane alone. As a tambourine's membrane is only one part of a larger whole, this is problematic; in fact, the membrane is not usually considered the instrument's main radiator as many tambourines completely lack a membrane and players often actively damp the membrane while playing. Therefore, in order to develop a more in-depth and foundational understanding of the tambourine's physics, we will discuss the nature of the instrument's directivity, analyze how its sound spectra varies across diverse playing styles, and present the various modes associated with its wooden frame and metal jingles. The modes have been estimated using finite element techniques.

9:30

4aMU3. The sound radiation and directivity of glockenspiel bars. Hanna Pavill (Brigham Young Univ., N284 ESC, Provo, UT 84602, pavillh@byu.edu) and Micah Shepherd (Brigham Young Univ., Provo, UT)

The directivity of a musical instrument describes the predominant directions that sound radiates from that instrument when it is being played. The shape of an instrument, as well as the note and playing style, can greatly affect its sound radiation and directivity. The glockenspiel is a percussion instrument made of pitched metal bars of uniform thickness laid out in a keyboard pattern and set on a frame. It produces sound when the player strikes the bars with a mallet. Uniform beam theory is often used to describe the modal behavior of the individual glockenspiel bars. However, when a single bar is stuck, the other surrounding bars and support frame influence the sound radiation and directivity. Therefore, glockenspiel measurements were performed using a directivity measurement system which has previously been used to measure musical instrument directivity. To eliminate strike-to-strike variance, an automatic striking device was used. This work will compare the sound radiation and directivity of a glockenspiel bar in the standard configuration, to an individual glockenspiel bar outside of the frame.

9:45

4aMU4. Further investigation of the vibration characteristics and sound radiation of Balinese gamelan gongs. Dallin T. Harwood (Brigham Young Univ., N283 ESC, Provo, UT 84602, dth37@byu.edu), Hanna Pavill, Emma Todd, Samuel D. Bellows, and Micah Shepherd (Brigham Young Univ., Provo, UT)

Balinese gamelan gongs are percussion instruments of special interest because of their unique shape and sound. Unlike a Chinese tam-tam, the gongs are thick and deep, with a protruding dome in the center and long edges that sharply wrap around the circumference of the gong. When struck, the larger gongs are designed to produce a strong beating pattern. Recent work in both the analytical modeling and high-resolution measurements of the gong has advanced our understanding of the physics of this suite of instruments. However, there is still much to be understood about the influence of the shape of the gong and what boundary conditions to expect from an instrument of the dimensions described. This paper will present measurements and discussion of the nature of both the gong's boundary conditions and its back cavity. Both will be used to support the continued development of theoretical models of the gong.

10:00–10:15 Break

10:15

4aMU5. The development of an automated striking device for repeatable percussion strikes. Jacob Sampson (Utah Valley Univ., 800 W University Parkway, Orem, UT 84058, jacobbsampson@gmail.com), Hanna Pavill, and Micah Shepherd (Brigham Young Univ., Provo, UT)

Many high-quality directivity systems involve a single microphone arc which rotates around a sound source. Since multiple data captures are necessary as the arc rotates around a sound source, the source should emit the sound in a steady and repeatable way at all arc positions. To obtain directivity measurements of a struck percussion instrument, it is crucial to be able to strike the instrument during each microphone arc position as consistently as possible. However, a human player typically cannot strike the instrument in the exact same way for many consecutive measurements. To overcome the shortcomings of human players, an automatic striking device was developed to aid in the study of the directivity of percussion instruments. The device consists of a striking implement holder attached to a stepper motor that

is controlled by an Arduino. The striking implement holder allows the user of the striking device to use different-sized mallets/drumsticks depending on the instrument that is being measured. The automatic striking device has been used to obtain directivity measurements on several instruments including a glockenspiel, cymbal, and drum. The details of the automatic striking device will be presented along with a subset of directivity results.

10:30

4aMU6. Ultrasound tongue imaging of vowel spaces across pitches in singing. May Pik Yu Chan (Linguist., Univ. of Pennsylvania, Dept. of Linguist 3401-C Walnut St., Ste. 300, C Wing University of Pennsylvania, Philadelphia, PA PA 19104-6228, pikyu@sas.upenn.edu) and Jianjing Kuang (Linguist., Univ. of Pennsylvania, Philadelphia, PA)

One important technique in singing is vowel modification: the adjustment of the resonance space based on the sung pitch for more efficient voice production. We explore whether vowel modification is a learned technique for enhanced acoustics, or if it is a necessary articulatory adjustment for high pitch production. 16 participants without vocal training participated in a singing experiment with ultrasound tongue imaging. Participants were asked to sing sets of English vowels across their comfortable pitch range rising by semitone in a steady tempo, resembling a vocal warm up exercise. Participants sang 5 sets of vowels in total, each set consists of 5 target vowels ([i], [ɛ], [æ], [ɑ], [u]) in randomized order with 1 filler ([ɔ]) closing each breath group. Images of tongue position were splined using DeepLabCut. Preliminary results show that untrained singers tend not to adjust their tongue positions by pitch, though cases of tongue lowering occasionally occurred, particularly for the participants who sing a wide pitch range. In contrast, additional pilot data from 2 trained operatic singers showed gradual tongue adjustments across their pitch range, neutralizing vowel contrasts at their highest pitches. We discuss findings with respect to vowel-pitch interaction, drawing implications on theories of voice production.

10:45

4aMU7. Analysis and interpretation of complex vibrato patterns: A novel parametric approach to genre-specific singing performance. Theodora I. Nestorova (Schulich School of Music, McGill Univ., 555 Sherbrooke St. W., Montréal, QC H3A 1E3, Canada, theodora.nestorova@mail.mcgill.ca), Gary Scavone (Music Res., McGill Univ., Montreal, QC, Canada), Ian Howell (Ann Arbor, MI), and Josh Gilbert (READS Lab, Harvard Univ., Boston, MA)

A complex, under-researched phenomenon, vibrato exists in a variety of musical styles and genre contexts. However, currently accepted acoustical analysis methods presume the vibrato is uniform, consistent, and persistent. This Western Art Music lens disregards significant stylistic characteristics of many genres with non-normative vibrato features. Therefore, a new system considering both regularity and shape of vibrato metrics in many genres over time is essential. 2 cross-genre songs and 1 exercise task were disseminated to 15 professional Operatic, Musical Theater, and Jazz singers. 16 pitch segments from each singer were subjected to sinusoidal extraction, $f_{;o}$ band-pass filtering, and an FFT LTAS in Praat. Each sample's mean half extent, pitch, and vowel was calculated and assessed using standard deviation, Coefficient of Variation (CV%), linear/polynomial/non-linear regression techniques in R. The results, corroborated by a perceptual survey distributed to professional singing pedagogues, indicated that vibrato variability predictably distinguishes performed genres. The CV% well characterizes vibrato variability and is higher in the Musical Theater and Jazz singing samples. A novel model – 4 parameter logistic s-curve regression – is proposed as optimal representation of multi-phasic vibrato with complex shapes. Such novel vibrato models may be employed to acoustically examine complex vibrato patterns in many instruments beyond singing.

11:00

4aMU8. Short-term retention of popular music in older adults: Support for a plasticity theory of implicit music knowledge acquisition. Annabel J. Cohen (Psych., Univ. of PE, Dept. of Psych., University of PE, 550 University Ave., Charlottetown, PE C1A 4P3, Canada, acohen@upei.ca), Corey Collett (Psych., Univ. of PE, Charlottetown, PE, Canada), and Kristen Gallant (Psych., Univ. of Waterloo, Waterloo, ON, Canada)

Based on our Plasticity Theory of Implicit Music Knowledge Acquisition (PTIMKA), we tested the hypothesis that adolescence is a sensitive period for acquiring musical information and lifelong musical grammar. Older adults ($N = 27$, mean age $= 65.7$ years; $SD = 6.7$) identified artist, title, and year of popularity and rated their familiarity for short excerpts of 36 songs popular between 1962 and 2021. Knowledge and familiarity were greater for music popular during the participants' adolescence. A subsequent surprise retention task required participants to choose which of 2 excerpts had been presented in the first task; for each of 36 trials, targets and foils represented the same era of popularity. Retention (dõ) scores and confidence in retention judgment were higher for the songs popular during adolescence, even though targets of all eras had just been presented in the previous 15 minutes. It is argued that popular music styles congruent with an adolescence-established grammar were more accurately encoded than styles violating this grammar, as would songs popular before or after adolescence. Prior data from younger adults showing trajectories opposite to those for older adults as a function of decade of popularity are further consistent with this interpretation and with PTIMKA. (Work supported by NSERC)

Session 4aNS

Noise, Physical Acoustics and Engineering Acoustics: Methods for Community Noise Testing and Analysis I

Alexandra Loubeau, Cochair
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Aaron B. Vaughn, Cochair
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Duncan Halsead, Cochair
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Chair's Introduction—8:30

Invited Papers

8:35

4aNS1. Dose error correction using simulation extrapolation for community noise dose-response modeling. Aaron B. Vaughn (NASA Langley Res. Ctr., 2 N Dryden StMS 463, Hampton, VA 23681-2199, aaron.b.vaughn@nasa.gov), Nathan B. Cruze, Matthew Boucher, and William Doebler (NASA Langley Res. Ctr., Hampton, VA)

The objective of this work is to provide a framework to account and correct for dose error in dose-response modeling due to measurement uncertainty. Error in noise measurements, especially in the case of limited monitoring locations in a community, can lead to an attenuation or misestimation of parameters in dose-response models. This error can result in overpredicted annoyance at lower doses and underpredicted annoyance at higher doses. Simulated data in the present work are based on previous NASA community studies and incorporate a notional design for future studies with the X-59 aircraft. Several populations of different annoyance response sensitivities are included. Simulation extrapolation (SIMEX) is used to correct for the dose error in a simple, fully pooled logistic regression. Results indicate the negative impact of attenuation is greatly diminished for all amounts of dose error considered, regardless of a population's annoyance sensitivity. Therefore, SIMEX can help produce a more accurate dose-response relationship.

8:55

4aNS2. The influence of atmospheric variability on predicted sonic boom metrics. Shane V. Lympany (Blue Ridge Res. and Consulting, 29 N Market St., Ste. 700, Asheville, NC 28801, shane.lympany@blueridgeresearch.com) and Juliet A. Page (Blue Ridge Res. and Consulting, Asheville, NC)

As part of the Quesst mission, NASA will fly the X-59 aircraft over selected communities to survey community responses to low sonic booms. Previously, we developed a Kalman filter method to estimate the loudness metrics experienced by each survey participant during each flight. The Kalman filter fuses acoustic measurements with predictions from PCBoom, a sonic boom propagation model. PCBoom requires vertical profiles of the temperature, humidity, and wind to propagate sonic booms through the atmosphere and predict the loudness metrics at the ground. Prior to each X-59 flight, NASA will launch weather balloons to measure the vertical atmospheric profile. However, the atmosphere changes continually with geographic location and time. The purpose of this work is to determine where and when to launch these weather balloons to minimize the uncertainty in the predicted loudness metrics caused by atmospheric variability. We analyzed the effect of atmospheric variability on the predicted Perceived Level in three communities in different climate zones in the United States. To achieve acceptable uncertainty in the Perceived Level, weather balloons should be launched from at least two different sites within the survey area within one hour of each X-59 flight.

9:15

4aNS3. Turbulence-induced variability of a far-field Falcon-9 sonic boom measurement. Kaylee Nyborg (Dept. of Phys. and Astronomy, Brigham Young Univ., BYU Dept. of Phys. and Astronomy, N284 ESC, Provo, UT 84602, kaylee.nyborg@byu.net), Mark C. Anderson, and Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Although aircraft sonic booms have been researched for decades, sonic booms from rocket booster landings are relatively recent events that are less studied. However, as booster landings become more prevalent, a better understanding of the associated sonic booms and community impact is needed. Atmospheric turbulence distorts sonic boom waveforms, affecting both the rise time and peak pressure, and can have a large impact on human perception metrics. As an initial investigation into the impact of turbulence on sonic boom

measurements, measurements were made of the reentry sonic boom during the SpaceX Falcon-9 SDA Tranche 0B mission at Vandenberg Space Force Base, California. An 11-microphone linear array spanning 150 m (500 ft) recorded the boom 8.5 km from the landing pad. This paper discusses results from this measurement, including variability of human perception metrics. For example, Perceived Level varied by up to 8 dB across the array. [Work supported through a NASA Graduate Research Fellowship.]

Contributed Papers

9:35

4aNS4. Development of a weather robust microphone configuration for sonic boom measurments. Jesse Blaine (Phys., Brigham Young Univ., C110 ESC, Brigham Young University, Provo, UT 84602, jlatch@byu.edu), Mark C. Anderson, and Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

This paper discusses ongoing development and testing of ground-based, weather-robust microphone measurement systems in conjunction with preparation for community testing of NASA's X-59 low-boom aircraft. Prior efforts [Anderson *et al*., Proc. Mtgs. Acoust. **42**, 040005 (**2022**)] resulted in the refinement of a ground-plate setup by investigating varied windscreen and plate diameter and thickness. The most recent goal has been to make the setup more compact and easier to manufacture. This paper discusses the results of additional tests performed on updated designs meant to find the balance between compactness and performance. Anechoic chamber testing was performed to show ground plate and windscreen performance and to compared results against prior versions. Outdoor tests included measurements during different wind conditions and over different ground surfaces to examine low-frequency wind noise reduction and ground impedance effects. Results discussed during the presentation suggest the new design strikes an acceptable compromise between compactness, manufacturing ease and robustness, and performance. [Work supported by NASA Langley Research Center through Analytical Mechanics Associates]

9:50

4aNS5. Investigating sonic booms using impulse metrics. Avery K. Sorrell (Dept. of Phys. and Astronomy, Brigham Young Univ., N281 Eyring Sci. Ctr., Provo, UT 84602, averysorrell137@gmail.com), Mark C. Anderson, Kaylee Nyborg, and Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

As an impulsive sound, a sonic boom exhibits traits similar to other transient acoustic impulses, such as a single strike of a sledgehammer strike or an exploding firework. Although several metrics exist to assess the nature and potential acoustic hazards of these other impulsive sounds, these metrics are rarely applied to sonic booms. This paper investigates the use of several metrics commonly applied to impulsive sounds, such as B and D durations or the kurtosis of the pressure waveform, for sonic booms recorded during previous NASA flight test campaigns, including CarpetDIEM (Carpet Determination in Entirety Measurements) and QSF18 (Quiet Supersonic Flights 2018). Understanding the behavior of these metrics in urban (QSF18) and rural (CarpetDIEM) environments adds to the larger body of knowledge concerning sonic boom measurements and properties and may be useful in quantifying sonic boom variability in communities.

10:05–10:20 Break

Invited Paper

10:20

4aNS6. Abstract withdrawn.

Contributed Papers

10:40

4aNS7. Assessment of noise complaints as an indicator of long-term effects of aircraft noise on communities surrounding aerodromes. Colin Novak (Mech. Automotive and Mater. Eng., Univ. of Windsor, 401 Sunset Ave., Windsor, ON N9B 3P4, Canada, novak1@uwindsor.ca) and Julia Jovanovic (Mech. Automotive and Mater. Eng., Univ. of Windsor, Windsor, ON, Canada)

Aircraft noise is considered by many as the most burdensome part of aircraft operations. Communities neighbouring airports often express concerns about the possible health effects of chronic exposure to noise. To quantify the long-term effects of environmental noise exposure, researchers often use the metric of annoyance. Annoyance is widely considered to be the most well-corroborated health effect of aircraft noise and a moderating factor for other suspected health effects. Annoyance can also be correlated to average cumulative noise levels, such that higher levels of chronic noise likely evoke higher levels of annoyance within the population. Annoyance data is typically collected using extensive surveys and/or interviews, which are costly and time-consuming. In the absence of annoyance data, complaints are often used as a proxy for annoyance. This research used complaint, noise and annoyance data to demonstrate that complaints do not equate to annoyance, nor do they correlate to cumulative noise metrics. Thus, while complaint data may prove useful in analyzing short-term response to operations, it should not be relied upon for the assessment of long-term impacts from aircraft noise exposure, nor should it be used to direct noise and annoyance mitigation initiatives.

10:55

4aNS8. Aircraft noise contours—From everything for everyone to nothing for anyone. Colin Novak (Mech. Automotive and Mater. Eng., Univ. of Windsor, 401 Sunset Ave., Windsor, ON N9B 3P4, Canada, novak1@uwindsor.ca) and Julia Jovanovic (Mech. Automotive and Mater. Eng., Univ. of Windsor, Windsor, ON, Canada)

Aircraft noise exposure contours were originally intended as a tool to facilitate compatible land use in the vicinity of airports. The simple concept was for authorities to use historic data and reasonable predictions for near future demands to create a map that identified areas of high aircraft noise impacts. Municipalities and planners would refer to this map to determine appropriate zoning around the airport, restricting noise sensitive development in high noise exposure areas. With time, different demands and applications of the noise contours emerged. Some stakeholders demanded longer term forecasts to allow for planning farther into the future. Some co-opted contours as a PR tool to suggest reduced impacts on communities. Some began using noise contours as a tool to protect against encroachment. Others began to look to noise contours as a representation of daily acoustic conditions in areas surrounding the airport. In Canada, a lack of oversight and guidance for the selection of inputs parameters for noise models, make it

such that noise contours have become a product illustrating whatever their creator intends, lacking objectivity and scientific rigour. This research demonstrates how noise contours can be designed to achieve desired results by modulating different input parameters.

11:10

4aNS9. What pandemic era travel restrictions revealed about aircraft noise, complaints and annoyance—SONICC 2020. Julia Jovanovic (Mech. Automotive and Mater. Eng., Univ. of Windsor, 401 Sunset Ave., Windsor, ON N9B 3P4, Canada, jovano11@uwindsor.ca) and Colin Novak (Mech. Automotive and Mater. Eng., Univ. of Windsor, Windsor, ON, Canada)

At the start of COVID-19 travel restrictions, Toronto Pearson International Airport experienced an approximate 80% reduction in traffic. This gave an unprecedented opportunity to investigate the impacts that a drastic reduction in aircraft noise would have on the communities surrounding the airport. Using the results of the Survey of Noise Impacts on Canadian Communities (SONICC) distributed in the summer of 2020, this research evaluated pre-pandemic and amidst pandemic aircraft noise annoyance in neighbourhoods surrounding Pearson Airport. The research investigated the effects of air traffic reduction on noise levels, complaint behaviour and annoyance. Complaint volumes correlated closely to the number of operations, experiencing a significant reduction. Despite the notable reduction in complaints, many complainants continued to vigorously complain, and some locations even experienced an increase in complaints. Pre-pandemic compared to amidst pandemic annoyance experienced reductions proportional to the average reduction in noise exposure. Despite significant reductions in noise, 33% of pre-pandemic highly annoyed (HA) respondents, remained highly annoyed, suggesting that anything short of a complete halt of operations would result in severe annoyance amongst a portion of the population.

11:25

4aNS10. Antiquated Canadian Aircraft Noise Guidance—Setting the grounds for encroachment and adverse community impacts. Julia Jovanovic (Mech. Automotive and Mater. Eng., Univ. of Windsor, 401 Sunset Ave., Windsor, ON N9B 3P4, Canada, jovano11@uwindsor.ca) and Colin Novak (Mech. Automotive and Mater. Eng., Univ. of Windsor, Windsor, ON, Canada)

A comprehensive research project at the University of Windsor entitled *Prediction and Management of Aircraft Noise Annoyance Around Canadian Airports* reviewed multiple components of Canada's TP 1247 Land Use in the Vicinity of Aerodromes. Using noise, complaints and survey data from Toronto Pearson International Airport, researchers evaluated the Noise Exposure Forecast (NEF) system as a tool for aircraft noise annoyance prediction and management. The NEF metric and its correlation to annoyance was examined as well as applicability of the NEF 30 threshold for the onset of significant annoyance. Further, the guideline for expected community response to noise was tested. The research found that most components of the NEF system are antiquated and in need of revision. A lack of prompt action to update Canadian guidelines for aircraft noise is setting the grounds for conflicts between stakeholders with competing interests and increasing the risks for greater noise impacts on future communities.

11:40

4aNS11. Investigation of speed and altitude effects on sound exposure level calculations for multiple helicopters. Mary L. Houston (NASA Langley Res. Ctr., NASA Langley Res. Ctr., M.S. 461, Hampton, VA 23601, Mary.L.Houston@NASA.gov), Kyle Pascioni (NASA Langley Res. Ctr., Hampton, VA), and James Stephenson (DEVCOM AvMC, Hampton, VA)

International and United States Federal regulations evaluate helicopter acoustic emissions for certification purposes as well as community noise impact assessments. The Sound Exposure Level (SEL) is a commonly used metric that represents the loudness of a single event, with a penalty for duration. For example, a flyover lasting ten seconds can have the same SEL as a five-second duration flyover that is 6 dB higher in peak level. Ten helicopters spanning the light and medium weight classes were modeled using flight test data. They were then virtually flown using the Advanced Acoustic Model at multiple speeds and altitudes. SEL and A-weighted SEL (SELA) were calculated at ground-based observer locations for each condition. Scaling laws with altitude are determined, and cruising airspeeds which result in the minimum SEL are found. Implications for future land-use planning will be discussed, and the relationships are reduced to general or vehicle-specific operational guidance for pilots.

Session 4aPA

Physical Acoustics and Signal Processing in Acoustics: Acousto-Optic Sensing

Carl R. Hart, Cochair
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Oleg A. Sapozhnikov, Cochair
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Yangfan Liu, Cochair
Purdue University, Ray W. Herrick Laboratories, 177 S, Russell Street, West Lafayette, IN 47907

Chair's Introduction—8:00

Invited Papers

8:05

4aPA1. Experimental studies with parallel phase-shifting interferometry. Yasuhiro Oikawa (Waseda Univ., Tokyo, Japan, yoikawa@waseda.jp), Risako Tanigawa (Waseda Univ., Tokyo, Japan), Mariko Akutsu (Railway Tech. Res. Inst., Tokyo, Japan), Kenji Ishikawa (NTT, Atsugi, Japan), and Denny Hermawanto (National Res. and Innovation Agency, Jakarta, Indonesia)

One of the most popular of devices currently used for acousto-optic sensing is the laser Doppler vibrometer (LDV), an interferometer originally designed to measure the vibration velocity of objects. LDVs are a popular way to measure and visualize acoustic fields, however it requires scanning the laser, which limits the measurement sound field. Recently, as the camera captures 2D images of the sound field on thousands of pixels simultaneously, the sound fields that can be measured are no longer limited to those that can be repeated. A high-speed camera captures 2D sound fields with tens or hundreds of thousands of frames per second, making it possible to film a slow-motion video of propagating sound in real time. The parallel phase-shifting interferometry (PPSI) with polarized high-speed cameras has been developed and introduced in the acoustics field as well. It has demonstrated impressive visualizations of airborne acoustic phenomena due to its high sensitivity and spatiotemporal resolution. It has also found many applications in recent years. This paper presents our experimental studies, especially measurement of aerodynamic sound, sound radiated from fast-moving sources or musical instruments, determination of acoustic center, etc.

8:25

4aPA2. Optical method for *direct* temperature measurement of high intensity focused ultrasound heating. Ghanem Oweis (Mech. Eng., American Univ. of Beirut, Bliss St., AUB, Bechtel Bldg. 408, Beirut 1107-2020, Lebanon, goweis@aub.edu.lb) and Hussein Daoud (Mech. Eng., American Univ. of Beirut, Beirut, Lebanon)

The acoustic pressure field or heat deposition from the passage of an ultrasonic wave can produce a perceptible alteration in the medium's optical refractive index. With proper placement of the light source and imaging camera, the optical effect can be recorded. This talk describes an optical temperature measurement methodology dubbed laser-ray-bundle (LRB) of the heat deposition from a high intensity focused ultrasound (HIFU) transducer in a transparent tissue phantom. The method does *not* require calibration, and *directly* converts the thermo-optic signal to temperature without recourse to any iterative computations. LRB boasts of excellent temporal and spatial resolutions and high temperature sensitivity better than any fine wire thermocouple. Furthermore, it is completely non-invasive. The LRB method is demonstrated with the heat deposition from single millisecond long HIFU pulses in a PDMS tissue phantom or from a train of single pulses with very good performance. The method represents a new capability for quickly assess a HIFU transducer output, and for laboratory and preclinical studies of heating near tissue interfaces including bone, air, or fat. A preliminary experimental setup for measuring the acoustic pressure field is presented.

8:45

4aPA3. Deeper, faster, and colorful photoacoustic imaging in life sciences. Junjie Yao (Duke Univ., 100 Sci. Dr., Hudson Hall Annex 261, Durham, NC 27708, junjie.yao@duke.edu)

Photoacoustic imaging (PAI) is an increasingly powerful technique for multi-scale anatomical, functional, and molecular imaging by acoustically detecting the optical absorption contrast in biological tissues. I will focus on several technological advancements in PAI that have collectively enabled fast, deep, and high-sensitivity biomedical applications and discoveries in life sciences. First, PAI has overcome the penetration limit by utilizing advanced internal light delivery techniques, allowing for super-deep (>10 cm) imaging. This

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breakthrough has extended the applicability of PAI to internal organ imaging in large animal models and humans. Second, innovative scanning technologies and deep-learning models have significantly accelerated PAI, enabling imaging speeds that are more than 1000 times faster while maintaining a large field of view and high spatial resolution. This enhancement facilitates the monitoring of highly dynamic biological processes at the microscopic scale, such as functional brain activities and glassfrog transparency. Lastly, PAI has greatly benefited from the genetically encoded switchable or tunable near-infrared photoacoustic-specific probes. By incorporating these probes, the sensitivity and specificity of PAI have been improved by more than 1000 times, enabling highly sensitive detection of malignant cancer, tissue hypoxia, and neuronal activities.

9:05

4aPA4. On the sensitivity and noise of acousto-optic sensing: Exploring the detection limits. Kenji Ishikawa (NTT, 3-1 Morinosato-Wakamiya, Atsugi 243-0198, Japan, ke.ishikawa@ntt.com)

Acousto-optic sensing has garnered increasing attention in recent years due to its non-contact nature, gaining importance in a range of acoustic measurement applications. Although its potential is widely recognized, the signal-to-noise ratio often requires enhancement in practical applications, which can impede further development. While more sensitive measurement methods have been evolving, the discussion around noise and detection limits in acousto-optic sensing remains limited. This presentation will highlight the author's recent findings on sensitivity and noise in acousto-optic sensing. It aims to illuminate the achievable detection limits through a detailed analysis of optical and acoustic noises. The insights provided will offer a deeper understanding of the complexities involved and potentially guide future advancements in this field.

9:25

4aPA5. Acousto-optic sensing: A scattering interpretation. Samuel A. Verburg (Dept. of Elec. and Photonics Eng., Tech. Univ. of Denmark, Ørsteds Plads, Bldg. 352, Kgs. Lyngby 2800, Denmark, saveri@dtu.dk) and Efren Fernandez-Grande (Dept. of Elec. and Photonics Eng., Tech. Univ. of Denmark, Kgs Lyngby, Hovedstaden, Denmark)

Acousto-optic sensing consists in using (laser) light as the sensing element in order to measure, visualise and study acoustic phenomena in a remote, non-invasive manner. One of the most common sensing techniques involves measuring acoustically-induced phase shifts of an optical probing beam by means of interferometry. Arguably, the main fundamental limitation of such sensing technique is that it provides projections of the acoustic field, instead of acoustic quantities at specific locations. In this study we examine the theory of spontaneous light scattering in transparent media, and use this framework to formulate a general acousto-optic sensing problem. Preliminary results based on numerical simulations indicate that light scattering could be applied to the measurement of acoustic pressure fluctuations. Some of the main challenges associated with the principle—mainly related to the weaknesses of the acousto-optic effect in air and relatively low signal-to-noise ratio—as well as the main fundamental differences from conventional interferometry techniques are discussed in this study.

9:45–10:00 Break

10:00

4aPA6. Sensing viscous acoustic flow: Using spider silk to hear. Jian Zhou, Junpeng Lai (Mech. Eng., Binghamton Univ., Vestal, NY), and Ronald Miles (Mech. Eng., Binghamton Univ., 85 Murray Hill Rd., Mech. Eng. Dept, Binghamton, NY 13902, miles@binghamton.edu)

Measurements obtained with a laser vibrometer show that a single strand of spider silk captures sound with almost full fidelity across a broad frequency range from infrasound to ultrasound, surpassing the performance of any known microphone or ear. The high responsivity of the silk to acoustic particle velocity enables the orb-weaving spider to hear sounds from meters away by utilizing its web—a hearing mechanism distinct from that of other animals that rely on an organ within their body. Laser-based measurements have also revealed that small creatures like mosquitoes can hear far-field sounds, similar to the spider, by sensing acoustic particle velocity rather than pressure. These findings offer insights for designing novel acoustic flow detectors and have led to the development of a recently commercialized flow microphone.

10:20

4aPA7. Laser interferometer measurement and reconstruction of supersonic projectile acoustic pressure fields. Matthew G. Blevins (U.S. Army Engineer Res. and Development Ctr., 2902 Newmark Dr., Champaign, IL 61822, matthew.g.blevins@usace.army.mil) and Gregory W. Lyons (DEVCOM Army Res. Lab., Adelphi, MD)

Sensing acoustic fields with a laser interferometer offers several advantages over conventional techniques: the probe beam does not disturb the medium in which it operates, the interferometer optics can be placed far from the source, and interferometry typically has a wider bandwidth than microphones. These advantages are particularly useful in situations where precise measurements of high-amplitude shock-generating sources are required. In this work we use a laser interferometer to study the generation and propagation of shock wave signatures of ballistic projectiles. A laboratory experiment was conducted to collect interferometer data from a variety of supersonic projectiles over a range of standoff distances. Simultaneously-captured microphone data is compared to the interferometer measurements, and techniques for inversion and reconstruction of the sound pressure field are discussed.

10:40

4aPA8. Deconvolution of spatially averaged shock wave forms. Carl R. Hart (U.S. Army Engineer Res. and Development Ctr., Cold Regions Res. and Eng. Lab., 72 Lyme Rd., Hanover, NH 03755, carl.r.hart@erdc.dren.mil) and Michael J. White (U.S. Army Engineer Res. and Development Ctr., Construction Eng. Res. Lab., Champaign, IL)

Noncontact optical methods (e.g., shadowgraphy, schlieren, and interferometry) allow visualization of propagating shock waves, avoid sensor-field interaction, and retain the potential for very high bandwidth. The heterodyne Mach-Zehnder interferometer produces quantifiable noncontact samples of the accumulated phase shifts induced by a shock wave, as viewed through its beam projection. This instrument is commercially available as a Laser Doppler Vibrometer (LDV). For shocked wave fields in air obeying spherical symmetry, point values for density and overpressure can be inferred. An important shortcoming for inference quality is the finite spatial dependence of the probe beam cross section, the profile shape itself serving to limit both the spatial resolution and the measurement bandwidth. We propose to model both of these effects of spatial averaging by constructing a model waveform and a model beam profile, convolving them, and comparing the result to synthesized LDV output. Mismatch between the convolved result and spatially averaged measurement can be quantified by a cost function, which is minimized via gradient descent.

10:55

4aPA9. Microphone diaphragm thermal noise floor measurement using a laser vibrometer. Morteza Karimi, Junpeng Lai (Mech. Eng., Binghamton Univ., Vestal, NY), Johar Pourghader (Mech. Eng., Binghamton Univ., Binghamton, NY), and Ronald Miles (Mech. Eng., Binghamton Univ., 85 Murray Hill Rd., Mech. Eng. Dept, Binghamton, NY 13902, miles@binghamton.edu)

The response of a microphone to thermal noise can be extremely difficult to predict and can adversely affect the performance of a given design. The direct measurement of the thermal noise floor of a microphone diaphragm could greatly improve our understanding of the parameters that affect the noise performance. A laser vibrometer can serve as a powerful tool, enabling precise and non-intrusive measurements of the diaphragm thermal noise. The experimental setup for measuring the diaphragm thermal noise using a laser vibrometer is described. Having both the noise floor, and the response to sound for a given microphone, the sound pressure-referred noise floor can be estimated. Measured and predicted results are presented for a variety of microphones. These measured results describe how design parameters, such as the size of the diaphragm, the dimensions of the backside volume, and overall geometry, impact the thermal noise floor measurements.

11:10

4aPA10. Development of an arrayed ultrasonic sensor using surface plasmon resonance (SPR). Kota Dezao (Doshisha Univ., 1-3 Tataramiyakotani, Kyotanabe-shi, Kyoto-fu 610-0321, Japan, kouta.nakiri@gmail.com), Shouta Kitajima, and Mami Matsukawa (Doshisha Univ., Kyotanabe, Kyoto, Japan)

Surface plasmon resonance (SPR) ultrasonic sensors are non-resonant and expected to show flat frequency response over a wide frequency range. Then, the sensors seem suitable for the detection of photoacoustic waves. We have developed a simple glass prism SPR sensor with thin Ag layer (deposited by an electron beam evaporator) and checked the frequency response as a function of Ag area diameter. The diameter of 1 mm was sufficient to obtain better frequency response than that of the calibrated needle transducer in the range of 2.5 to 7.0 MHz. By covering a very thin Au film on the Ag layer, we succeeded in longer working time (>6 months) of the sensor. An array-type SPR sensor (5×5, diameter of each sensor 1 mm) has also been fabricated to evaluate pressure distribution of ultrasonic waves. The response of each sensor was similar, telling the high reproducibility of the sensor fabrication.

11:25

4aPA11. Sound-field image denoising using deep neural network considering physical characteristics of sound. Daisuke Urata (Waseda Univ., 2-19-3, Haruhino, Asao-ku, Kawasaki, Kanagawa 215-0036, Japan, daisuke.u.0530@gmail.com), Yasuhiro Oikawa (Waseda Univ., Tokyo, Japan), and Kenji Ishikawa (NTT, Atsugi, Japan)

This paper presents a method of denoising utilizing deep neural networks (DNN) in sound field images measured by optical methods. Optical sound measurement has attracted much attention because of its capability to visualize sound field with high spatial resolution from a distance, which is difficult with conventional microphones. However, optical methods have the issue of the measured sound-field images often being heavily distorted by noise. This is caused by the weak phase of light changed by sound. Conventionally, noise-reduction filters considering the physical properties of sound have been known to be used for sound-field denoising. In this background, we propose a DNN-based sound-field denoising method that takes into account the physical properties of sound. In the DNN, a network architecture originally used for denoising natural images is employed, and we integrate loss functions based on the Helmholtz equation. Additionally, a 2D sound field dataset obtained from numerical acoustic simulation with random parameters is used during the training. Numerical and experimental data comparison experiments showed that the proposed DNN-based sound-field denoising outperformed the previous non-DNN methods.

11:40

4aPA12. Acoustic ground effects simulations from asteroid disruption via the "Pulverize It" method. Brin Bailey (Phys., Univ. of California, Santa Barbara, University of California, Santa Barbara, Broida Hall, Santa Barbara, CA 93106, brittanybailey@ucsb.edu), Alexander N. Cohen, Philip Lubin (Phys., Univ. of California, Santa Barbara, Santa Barbara, CA), Darrel K. Robertson (NASA Ames Res. Ctr., Moffett Field, CA), Mark Boslough (Univ. of New Mexico, Albuquerque, NM), Sasha Egan (Energetic Mater. Res. and Testing Ctr., New Mexico Inst. of Mining and Technol., Socorro, NM), and Elizabeth Silber (Sandia National Labs., Albuquerque, NM)

Our simulations show that PI ("Pulverize It"), a NASA Phase II NIAC study, is an effective multi-modal approach for planetary defense that can operate in extremely short interdiction modes (with intercepts as short as hours prior to atmospheric entry) as well as long interdiction time scales with months to years of warning. The basic process is complete disruption of the threat via fragmentation. In long warning time cases, the fragment cloud spreads enough to miss Earth, resulting in no ground effects. In cases where the warning time is short, the fragments (typically <10 m in diameter) will enter Earth's atmosphere where their energy is dissipated in a series of ground-level optical pulses and de-correlated shock waves, mitigating any significant damage. We investigate the acoustic ground effects through a set of simulation codes that model the interaction of asteroid fragments with the Earth's atmosphere following threat interception. Even in the short warning time cases where the fragments enter the atmosphere, our simulations show that threats mitigated by the PI method produce vastly less damage on the ground when compared to the same unfragmented case, yielding shock wave over-pressures under 3 kPa. Our simulations support the proposition that threats sized 20 m—1000 m in diameter can be effectively mitigated through fragmentation, resulting in acoustic ground effects that are below estimated damage thresholds and yield short- and long-term non-lethal effects.

Session 4aPP

Psychological and Physiological Acoustics and Speech Communication: Interactions Between Voice and Speech Perception

Etienne Gaudrain, Cochair
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McGill University, 2001 McGill College, Montreal, H3A 1G1, Canada

Invited Papers

8:00

4aPP1. Voice perception as a bridge between psychoacoustics and speech intelligibility. Etienne Gaudrain (Lyon Neurosci. Res. Ctr., CNRS, Ctr. Hospitalier Le Vinatier - Bâtiment 462 - Neurocampus, 95 bvd Pinel, Lyon 69000, France, etienne.gaudrain@cnrs.fr)

While clinical assessments of speech intelligibility focus on the reception of phonetic to lexical cues by a listener, in natural situations, speech also contains cues that inform the listener about the identity of the talker, or their emotional state. These *indexical* cues contribute to the access of phonetic information, either through talker normalisation, or by promoting the segregation of competing talkers. They can also directly affect the interpretation of the lexical content, e.g., through prosody. In other words, indexical cues play a crucial role in everyday communication. Yet, they are rarely considered in clinical evaluations of hearing impairment. Through various studies involving child and adult hearing aid and cochlear implant users, it appears that some voice cues are more affected by hearing loss than others, and this could have consequences on the development and remediation strategies. These studies also highlight the central role that voice perception study could play in connecting psychoacoustics to communication sciences. As voice literally carries the linguistic information in speech, it sits at the forefront of auditory processing. The mechanisms underpinning hearing loss thus tend to directly affect the neural representations of voice properties. As voice manipulation technologies are becoming more precise and more widely available, new opportunities arise to link peripheral auditory deficits to speech processing difficulties in populations with impaired hearing.

8:20

4aPP2. A mechanistic investigation of processing costs associated with cross-talker acoustic-phonetic variability. Sahil Luthra (Psych., Carnegie Mellon Univ., 5000 Forbes Ave., Pittsburgh, PA 15213, sahil.bamba.luthra@gmail.com)

The acoustic realization of speech sounds may differ substantially across talkers, such that (for example) a sound interpreted as “s” in one voice might be considered a “sh” in another. In this talk, I will review select neurobiological and computational data illustrating how listeners cope with this variability. I will also review behavioral data showing how talker variability can hinder processing, with slower and/or less accurate processing when the listening environment contains multiple voices compared to one voice. I argue that at least three mechanisms underlie these multi-talker processing costs. First, multi-talker processing costs may arise because talker changes disrupt a listener’s ability to attend to the target talker. Second, I make the novel claim that talker changes might impose processing penalties by inducing uncertainty about whether the extant acoustic-to-phonetic mapping is appropriate, motivated by recent data that when multi-talker processing costs are attenuated by a preceding carrier phrase, this manifests as slowed responses to single-talker speech rather than improved processing of mixed-talker speech. Finally, talker variability costs may result from making on-the-fly adjustments to the mapping between acoustics and phonetic categories. Overall, this talk highlights the ways in which listeners contend with and are hindered by talker variability.

8:40

4aPP3. What do voices contribute to acoustic context effects in speech perception? Christian E. Stilp (Psychol. and Brain Sci., Univ. of Louisville, 317 Life Sci. Bldg., Louisville, KY 40292, christian.stilp@louisville.edu)

All perception takes place in context, and speech perception is no exception. Surrounding sounds form a context that guides speech perception. This is particularly true when earlier sounds inform the perception and/or recognition of later sounds. Several lines of research have advanced models where listeners use preceding context to account for characteristics of the talker’s voice, such as vocal tract properties, articulatory maneuvers, or the speaker’s identity. Conversely, research with nonspeech sounds and/or nonhuman animals

has advanced alternative models where perception is responding more to signal acoustics than to vocal characteristics. In this talk, I will review recent evidence from our research on acoustic context effects and talker normalization when hearing speech and nonspeech (music) sounds. Parallel patterns of performance across speech and music domains are conducive to, but not definitive evidence of, acoustic context being utilized according to general signal acoustics, with speech- and voice-specific contributions coming later in the processing stream.

9:00

4aPP4. How does voice familiarity affect speech intelligibility? Emma Holmes (Univ. College London (UCL), Dept. of Speech Hearing and Phonetic Sci., Chandler House, 2 Wakefield St., London WC1N 1PF, United Kingdom, emma.holmes@ucl.ac.uk)

People often face the challenge of understanding speech when other sounds are present (speech-in-noise perception)—which involves a variety of cognitive processes, such as attention and prior knowledge. We have consistently found that familiarity with a person's voice improves the ability to understand speech-in-noise, using both naturally familiar (e.g., friends and partners) and lab-trained voices. In this talk, I will describe experiments in which we manipulated voice acoustics (such as fundamental frequency and formant spacing). For example, we have measured the smallest deviations in acoustics that listeners can discriminate for familiar and unfamiliar voices, and have examined how manipulations to voice acoustics affect voice recognition and speech intelligibility for familiar voices. This work has provided insights into the processes underlying the familiar-voice intelligibility benefit—for example, by contrasting explanations based on predictions of voice acoustics with those involving cognitive demands. I will discuss the implications of our findings for theories of speech perception, and the implications for populations who typically find speech perception particularly challenging (e.g., older adults).

9:20

4aPP5. Speech comprehension in the context of speaker changes: The importance of voice-feature continuity at the cocktail party. Jens Kreitewolf (McGill Univ., 2001 McGill College, Montreal, QC H3A 1G1, Canada, jens.kreitewolf@mcgill.ca)

In this talk, I will show how different groups of listeners use voice acoustics to enhance speech comprehension under adverse listening conditions, specifically when the auditory scene comprises a multitude of sounds heard at once. These "cocktail-party"-like situations pose a difficult conceptual problem: To comprehend target speech, listeners need to attend to the target voice while at the same time ignoring other irrelevant sounds. The cocktail-party problem is made considerably easier when all target sounds are spoken by the same talker. Previous work suggests that such benefits from voice continuity can be—in large part—attributed to two prominent voice features: Glottal-pulse rate (GPR) and vocal-tract length (VTL). GPR determines the fundamental frequency of a speech sound and is perceived as vocal pitch; VTL determines the spectral envelope of a speech sound and is perceived as an aspect of vocal timbre. Apart from being important voice identity cues, GPR and VTL have been shown to play a crucial role in cocktail-party listening. Here, I will present data from a series of experiments highlighting the importance of voice-feature continuity for speech comprehension at the cocktail party.

9:40–9:55 Break

9:55

4aPP6. Perception of dynamic pitch and prominence in speech. Hae-Sung Jeon (School of Psych. and Humanities, Univ. of Central Lancashire, University of Central Lancashire, Preston PR1 2HE, United Kingdom, hjeon1@uclan.ac.uk)

The perception of dynamic – constantly changing – pitch in speech has been extensively studied in psychoacoustics and linguistics. In psychoacoustic studies, listeners are usually presented with short stimuli such as vowels or syllables, and their ability to discriminate a pair of stimuli is assessed. On the other hand, linguistic studies concern intonation over an utterance. Intonation entails not only acoustic prominence realised by pitch, duration, and loudness, but also listeners' knowledge about the relative prominence between syllables or between words in their language. This paper discusses a series of speech perception experiments using both psychoacoustic and linguistic tasks. Participants judged either relative pitch height or prominence between two pitch peaks or valleys in an utterance. Native English speakers in different age and dialectal groups were tested. The results showed that, first, listeners' pitch height discrimination in the utterance context seems to be more accurate than previously reported. Second, there is a robust perceptual asymmetry between pitch peaks and valleys, the valleys posing significant challenges in perception. Third, listeners' perception of pitch height and prominence is disassociated. The findings taken together suggest an intricate interaction between the physical properties of the stimuli and listeners' top-down knowledge in the perception of speech intonation.

10:15

4aPP7. Sensitivity to speech-relevant features in hallucination-prone individuals. Julia Erb (Inst. for Systems and Robotics - Lisboa and Dept. of Bioengineering, Instituto Superior Técnico, Universidade de Lisboa, Portugal, Avenida Rovisco Pais, 1, Lisbon 1049-001, Portugal, erbjulia@gmail.com), Patrícia Figueiredo (Inst. for Systems and Robotics - Lisboa and Dept. of Bioengineering, Instituto Superior Técnico, Universidade de Lisboa, Portugal, Lisbon, Portugal), and Ana P. Pinheiro (Faculdade de Psicologia, Universidade de Lisboa, Lisbon, Portugal)

As hallucinations occur in the absence of an external stimulus, they constitute an intriguing model for how percepts are generated and for how perception can fail. This study explores whether hallucination proneness is linked to an altered perception of speech-related acoustic features. Involving 320 healthy adults with varying predispositions for hallucinations, participants evaluated ambiguous sound textures for their speech-likeness. Psychophysical reverse correlation revealed that higher hallucination proneness was associated with reduced weighting of speech-typical low-frequency acoustic energy. Temporal modulation discrimination capabilities were unrelated to hallucination proneness in a subset of 41 participants. Confidence judgments in individual trials were influenced by both acoustic evidence and individual hallucination proneness and schizotypy scores. Overall, these findings suggest that hallucination-prone individuals exhibit qualitative and quantitative changes in their perception of speech-relevant modulations, supporting the notion of altered perceptual priors and differential weighting of sensory evidence.

10:35

4aPP8. Enhancing sarcasm detection through multimodal data integration: A proposal for augmenting audio with text and emoticon. Xiyuan Gao, Shekhar Nayak (Campus Fryslân (Lang., Technol. and Culture), Univ. of Groningen, Leeuwarden, Netherlands), and Matt Coler (Campus Fryslân (Lang., Technol. and Culture), Univ. of Groningen, Wirdumerdijk 34, Leeuwarden 8911CE, Netherlands, m.coler@rug.nl)

Sarcasm detection presents unique challenges in speech technology, particularly for individuals with disorders that affect pitch perception or those lacking contextual auditory cues. While previous research [1, 2] has established the significance of pitch variation in sarcasm detection, these studies have primarily focused on singular modalities, often overlooking the potential synergies of integrating multimodal data. We propose an approach that synergizes auditory, textual, and emoticon data to enhance sarcasm detection. This involves augmenting sarcastic audio data with corresponding text using Automatic Speech Recognition (ASR), supplemented with emoticons based on emotion recognition and sentiment analysis. Emotional cues from multi-modal data are mapped to emoticons. Our methodology leverages the strengths of each modality: emotion recognition algorithms analyze the audio data for affective cues, while sentiment analysis processes the text generated from ASR. The integration of these modalities aims to compensate for limitations in pitch perception by providing complementary cues essential for accurate sarcasm interpretation. Our approach is expected to significantly improve sarcasm detection, especially for those with auditory processing challenges. This research highlights the potential of multimodal data fusion in enhancing the subtleties of speech perception and understanding, thus contributing to the advancement of speech technology applications.

10:55

4aPP9. Exploring auditory emotion perception in cochlear implant users: Investigating the interplay of speech processing and affective signals. Sebastien Paquette (Psych., Trent Univ., A-806 Foxe St., Peterborough, ON K9H6Y7, Canada, sebastienpaquette@trentu.ca)

Human emotions are intricately expressed through vocal sounds, encompassing affective prosody in speech and non-verbal cues such as screams and laughter. Recent evidence indicates that vocalizations take neurophysiological precedence over speech-embedded emotions and are generally easier to identify. However, Cochlear implant (CI) users still face challenges deciphering the subtle nuances in these primal signals. The implant's limited fidelity in transmitting acoustic information results in highly variable levels of emotion perception abilities among its users. Identifying the factors explaining this significant variability in abilities among CI users remains of great interest. Our recent investigations into CI users' abilities to perceive emotions and speaker sincerity have often incorporated diverse aspects of auditory proficiency, including pitch discrimination, music processing, and speech intelligibility. The combination of results from these different projects can help shed light on the intricate interplay between speech processing and emotional recognition in CI users. Surprisingly, even when presented with emotional musical stimuli, CI users' proficiency often leaned toward processes related to speech intelligibility, proposing common mechanisms underlying linguistic and affective processes in CI users that do not readily relate to musical skills or pitch sensitivity. Hence, maintaining a clinical focus on speech processing remains crucial, even when exploring affective skills in CI users.

Contributed Papers

11:15

4aPP10. Spectral resolution and voice cue weighting in adults with normal hearing. Victoria A. Sevich (Dept. of Speech and Hearing Sci., The Ohio State Univ., 1070 Carmack Rd., Columbus, OH 43210, sevich.1@osu.edu) and Terrin N. Tamati (Dept. of Otolaryngology—Head & Neck Surgery, Vanderbilt Univ. Medical Ctr., Nashville, TN)

Cochlear implant (CI) users use acoustic voice cues differently than normal hearing (NH) adults to identify a talker's gender. Specifically, whereas NH listeners weight a talker's fundamental frequency (f0) and resonance (operationalized as vocal tract length or VTL) equally, CI users rely almost exclusively on f0. CI users' abnormal cue weighting may partially arise due to degraded auditory information delivered by the CI. We hypothesized that altering the amount of spectral information in the signal impacts voice cue weighting in NH listeners. Thirty NH adults performed a gender identification task. Auditory stimuli were monosyllabic words synthesized to have one of five f0 values and one of five VTL values. Synthesized voices were processed using a 16-, 8-, and 4-channel noise-band vocoder. Perceptual weights for each voice cue were estimated as the coefficients for f0 and VTL in regression models, with higher coefficients corresponding to stronger perceptual weightings. Listeners relied more on f0 in conditions with less spectral resolution than in conditions with greater spectral resolution. Cue weights for VTL did not change across conditions. Results suggest that removing frequency information from the auditory signal can modify the extent to which listeners use f0 to identify the gender of a talker.

11:30

4aPP11. Acoustic divergence from the training sample determines talker identification accuracy for emotional voices. Lue Shen (Dept. of Speech, Lang., and Hearing Sci., Boston Univ., Boston University, Dept. of Speech Lang. and Hearing Sci., Boston, MA 02215, shenlue@bu.edu), Yuxuan Wang (Dept. of Biostatistics, Boston Univ., Boston, MA), Patrick Wong (Dept. of Linguist., The Chinese Univ. of Hong Kong, Hong Kong, Hong Kong), and Tyler K. Perrachione (Dept. of Speech, Lang., and Hearing Sci., Boston Univ., Boston, MA)

Different emotional states introduce substantial acoustic variations in talkers' voices. It remains unclear how within-talker variability across emotional states affects listeners' ability to maintain perceptual constancy during talker identification. Here, we investigated (1) how changes in talkers' emotional state affected talker identification accuracy, (2) how emotional state affected key features of voice acoustics, and (3) how emotion-related changes in these acoustic features affected listeners' talker identification performance. Forty-eight listeners learned to identify talkers from speech expressing one emotional state (neutral, fearful, or angry) and then attempted to generalize that knowledge to speech expressing another emotional state. Talker identification accuracy was significantly worse in untrained emotions. Changes in voice acoustics across emotions were characterized for mean F0, F0 variability, jitter, HNR, speaking rate, and mean F2. To determine how emotion-related acoustic changes affected talker identification, we modeled talker identification accuracy as a function of difference in these features between training and test stimuli. Accuracy decreased as acoustic differences increased, regardless of talkers' emotion. Thus, perceptual constancy depends on acoustic similarity to prior experience with a talker's voice. Larger acoustic deviations, like those introduced by changes in emotional state, are more likely to cause a listener to misidentify a talker.

Session 4aSA

Structural Acoustics and Vibration, Biomedical Acoustics, Musical Acoustics, Physical Acoustics, and Engineering Acoustics: Additive Manufacturing and Acoustics

Christina Naify, Cochair
Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Rd, Austin, TX 78758

Kathryn Matlack, Cochair
University of Illinois at Urbana-Champaign, 1206 W Green St, Urbana, IL 61801

Matthew Luu, Cochair
Penn State, 446 Bluecourse Dr (Apt907), State College, PA 16803

Thomas Bowling, Cochair
Naval Surface Warfare Center Carderock Division, Bethesda, MD 20817

Chair's Introduction—9:00

Invited Papers

9:05

4aSA1. Ultrasonic characterization of material anisotropy in additively manufactured AlSi10Mg. Joseph A. Turner (Mech. and Mater. Eng., Univ. of Nebraska-Lincoln, University of Nebraska-Lincoln, Lincoln, NE 68588, jaturner@unl.edu), Nathanial Matz, W. T. Brandl (Mech. and Mater. Eng., Univ. of Nebraska-Lincoln, Lincoln, NE), Pulkit Kumar, Ronald A. Roberts, and Peter C. Collins (Ctr. for Nondestruct. Eval., Iowa State Univ., Ames, IA)

Metal additive manufacturing (AM) based on laser powder bed fusion (LPBF) generates components by melting metal powder on a layer-by-layer basis. The melting and cooling process often generates samples with a preferred material symmetry aligned with the build direction. This anisotropy affects mechanical performance and can be challenging to characterize nondestructively. Here, LPBF was used to create samples of AlSi10Mg with specific geometries to affect the overall anisotropy. In addition, the hybrid AM process of interlayer milling was used to impact the microstructure and residual stress of some samples. Ultrasonic measurements were used to characterize the samples using both coherent wave and diffuse wave experiments to capture the anisotropic nature of the wave speed and scattering. The material symmetry and morphology of the grains affect the wave speed, attenuation, and backscatter with respect to direction. Furthermore, spatially resolved acoustic spectroscopy was used to provide insight regarding the localized wave speeds with respect to sample location and propagation direction. The experimental data were used collectively to quantify differences between the AM processes used to create the samples. Such information can be used to guide AM process parameters to optimize sample performance. Finally, prospects for characterization of residual stresses will be discussed.

9:25

4aSA2. Direct sound printing: Way of manipulating ultrasonic chemistry to print directly engineering structures and remotely inside body. Shervin Foroughi (Mech. Eng., Concordia Univ., Montreal, QC, Canada), Mohsen Habibi (Mech. Eng., Univ. of California Davis, Davis, CA), and Muthukumaran Packirisamy (Mech. Eng., Concordia Univ., EV4-145, 1455 de Maisonneuve Blvd W, Montreal, QC H3G 1M8, Canada, m.packirisamy@concordia.ca)

Direct sound printing (DSP) is a new class of additive manufacturing processes developed in our lab, in which chemical reactions during the 3D printing process are driven by sonochemical route using cavitation bubbles induced by focused ultrasound waves. This invited paper will present methods and possibilities of printing engineering structures with DSP. In addition, this talk will cover a new area called remote distance printing (RDP) and consequent applications. RDP is a new realm introduced by DSP method in which the printing location is not accessible by common energy sources like light or heat. In this situation, ultrasound could penetrate optically opaque materials and conduct printing without direct access to the printing location. This concept opens a wide variety of applications in engineering or medical fields. The focus of this paper is the application of DSP-RDP in biomedical application to print objects inside body without open surgery in a non-invasive manner. Ultrasound penetrates skin and tissues in DSP and is focused on the printing location inside body where the printing material is injected. This work explains DSP in detail and the interaction of the sound with the printing material and how the material is transformed from liquid to solid. The process is demonstrated using a test study conducted using tissue phantoms and also real porcine tissue. This work opens new applications to 3D print with ultrasound where no other 3D printing approaches can achieve.

9:45

4aSA3. Monitoring the build history of a wire-arc additively manufactured part using structural resonances. Karl A. Fisher (Mater. Eng., Lawrence Livermore Natl. Lab., 7000 E. Ave. Liveremore, CA 94551, fisher34@llnl.gov), John Elmer (Mater. Eng., Lawrence Livermore Natl. Lab., Livermore, CA), and James Candy (Computational Eng. Div., Lawrence Livermore National Security, Livermore, CA)

Wire arc additively manufactured parts can take a significant amount of time to fabricate from several hours to days. During this process, the part is exposed to a steady localized heating at the build zone, and rapid cooling away from the arc contact point. The heated zone is constantly moving across the part geometry during the build resulting in spatially localized heating and cooling throughout the part geometry. In this investigation, we utilize the broadband emission signal from the wire arc to measure the acoustic response of the part build. Experimental results are obtained using contact emission transducers mounted such that they are isolated from the direct heat to avoid damage. Transient acoustic signals are recorded throughout the entire build process and evaluated using Fourier analysis. Acoustic spectra are utilized to capture modal data which slowly track the parts construction. In a second effort the modal response of the part is modeled as a function of each build layer and the two modal structures are compared. The emphasis is to understand and track the modal response of the structure of the part as it is fabricated and develop a real time process monitoring capability.

10:05–10:20 Break

Contributed Papers

10:20

4aSA4. Preliminary investigation and screening study of relationships between acoustic emission signal features and powder blown laser-directed energy deposition parameters for *in situ* monitoring of metals additive manufacturing. Emmeline Evans (Mech. Eng., Georgia Inst. of Technol., 801 Ferst Dr., Atlanta, GA 30308, eevans70@gatech.edu), Erin Lanigan (NASA Marshall Space Flight Ctr., Huntsville, AL), and Aaron Stebner (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

Metal additive manufacturing (AM) processes have allowed for easier fabrication of parts made from alloys that are difficult to machine and typically yield less material waste than traditional manufacturing methods. Therefore, interest in metal AM for in-space components has grown in recent years. As this interest in metal AM as grown, so too has the need for real-time process monitoring for defect detection during printing. However, commonly used process monitoring methods, such as melt-pool imaging, are too data intensive for automated analysis to be performed and reported in real-time. Acoustic emission (AE) monitoring presents an alternative approach in which one-dimensional data can be captured continuously during printing and analyzed more quickly than image data. This work presents a methodology for and preliminary findings from a screening study that explores the effects of powder blown laser beam directed energy deposition (DED-LB) parameters on resulting AE that occur during printing. The primary factors considered in this study are print parameters that have been found in literature to predict part density, such as laser power and mass flow rate, and are studied here to determine their contributions to AE. This work is supported by the NASA Space Technology Graduate Research Opportunities program.

10:35

4aSA5. Internal damping measurements of additive manufactured metal beams. Peter K. Jensen (Dept. of Phys. and Astronomy, Brigham Young Univ., Brigham Young University, Provo, UT 84602, peterkj@byu.edu), Joshua T. Mills, and Micah Shepherd (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Modern additive manufacturing (AM) techniques have created an endless number of new design possibilities. Naturally, it is important to understand how the material properties, including internal damping, differ for AM structures. An experimental procedure has been developed to measure an upper bound for the internal damping of metal beams by minimizing the effects of energy dissipation at the supports and acoustic radiation into the surrounding air. In this talk, measurements for the loss factor of several common metals will be compared to equivalent samples constructed by powder bed fusion at varying angles. The results will also be compared to Zener's thermoelasticity model, developed for isotropic Euler beams in flexure. Reasons for deviation from theory which arise from the manufacturing technique will be explored.

10:50

4aSA6. Influence of ultrasonic parameters on microstructural refinement and defect elimination in ultrasound-assisted laser-based metal additive manufacturing. Lovejoy Mutswatiwa (Eng. Sci. and Mech., The Penn State Univ., 212 Earth And Eng. Sci, University Park, PA 16802, lpm5609@psu.edu), Judith A. Todd (Eng. Sci. and Mech., The Penn State Univ., State College, University Park, PA), Edward Reutzel (CIMP 3D, Penn State Appl. Res. Lab., University Park, PA), and Christopher M. Kube (Eng. Sci. and Mech., The Penn State Univ., University Park, PA)

Acoustic cavitation, streaming, and energy absorption during solidification in ultrasound-assisted direct energy deposition additive manufacturing (DED-AM) have been reported to drive microstructural refinement, defect elimination, and mechanical property improvement. However, the influence of individual ultrasonic parameters such as frequency, amplitude, intensity, and sonication duration remains unknown. This is mainly because of challenges in real-time quantification of the influence of ultrasound on the rapid solidification and microstructural development processes encountered in ultrasound-assisted AM. High temperature and opaque molten metal confined within micro-length scale melt pools further challenge the characterization of ultrasound's influence on microstructural development. Building upon our recent *in situ* observation of acoustic cavitation and streaming in sonicated laser-generated melt pools, this talk will highlight our efforts to correlate vibration frequency, amplitude, and intensity with grain size and texture of Al7075 alloy fabricated using ultrasound-assisted DED-AM. DED additively manufactured Al7075 is susceptible to solidification cracking. Therefore, this presentation will also showcase the influence of ultrasonic parameters on cracking suppression and defect elimination. In addition, the effect of sonication duration on microstructure will also be elaborated. Lastly, the presentation will showcase our future work toward upscaling ultrasound-assisted AM to large parts with complex geometries.

11:05

4aSA7. Sound absorption properties of additively manufactured porous materials with minimal surface pore geometries. Anthony Ciletti (Aerosp. Engineerign, Wichita State Univ., Wichita, KS), Martha C. Brown (NASA Langley Res. Ctr., Hampton, VA), and Bhisham Sharma (Mech. Eng. - Eng. Mech., Michigan Tech Univ., 1400 Townsend Dr., Houghton, MI 49931, bnsharma@mtu.edu)

This study explores the use of additive manufacturing, specifically stereolithography (SLA), to create porous acoustical materials with precise pore geometries for aircraft engine noise reduction. Unlike traditional methods like foaming and fiber-spinning, SLA allows for exact control over pore shapes. Three triply periodic minimal surface (TPMS) pore designs were investigated: split-P, lidinoid, and diamond. These structures were designed through implicit modeling and made from polymeric resin. Their sound absorption capabilities were tested under normal incidence in a two-microphone impedance tube to examine how porosity influences their

acoustic performance. Findings revealed unique sound absorption profiles for each geometry, which can be modified by adjusting porosity. Additionally, the static flow resistivity was measured using a raylometer, where results showed that flow resistance is a key factor in their absorption efficiency. Among the three samples, the split-P geometry shows superior sound absorption. This preliminary research highlights the potential of using tailored TPMS geometries in acoustic liners for effective noise control in aerospace applications.

11:20

4aSA8. Abstract withdrawn.

11:35

4aSA9. Effects of printer variation on vibration response of fused deposition modeling-fabricated elastic beams with symmetric and asymmetric resonators: A round robin study. Christina J. Naify (Appl. Res. Labs., The Univ. of Texas at Austin, 4555 Overlook Ave SW, Washington, DC 20375, christina.naify@gmail.com), Colby W. Cushing, Nathan P. Geib, Matthew Wash (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Jared Allison (Univ. of Texas at Austin, Austin, TX), Caleb F. Sieck (Code 7160, U.S. Naval Res. Lab., Washington, DC), Alec K. Ikei, Amelia Ware (Acoust., US Naval Res. Lab., Washington, DC), Thomas Bowling (Naval Surface Warfare Ctr. Carderock Div., Bethesda, MD), Benjamin S. Beck, Callie Zawaski (Eng. Acoust., Penn State Appl. Res. Lab, State College, PA), and Abigail T. Juhl (Air Force Res. Lab., Wright Patterson Air Force Base, OH)

Additive manufacturing has expanded rapidly as a production tool due to its ease of use for rapidly producing parts. The most common polymer-based additive approach is fused deposition modeling (FDM) which uses plastic filament to build parts additively, one line at a time, to build a 3D shape. FDM printers are ubiquitous in university, government and industry settings, making them ideal for mass-production of shared designs across institutions. While it is commonly known that FDM finished products have variations due to differences in printer model and slicing and extrusion settings, little has been done to quantify the effects of these variations on elasto-dynamic response. In this study, a multi-institutional round robin approach is used to quantify the printer-to-printer variations of a structure comprised of a thin beam with attached resonators. The parameters of the round robin study involve printing the same geometry, with the same base material, on whatever FDM-type printers are available at each of six contributing institutions. All samples were then sent to a common location and tested on the same apparatus to limit experimental variability. This presentation will discuss the design of the study, test and modeling efforts, and a summary of results.

THURSDAY MORNING, 16 MAY 2024 ROOM 203, 8:00 A.M. TO 9:25 A.M.

Session 4aSCa

Speech Communication: Language and Sports

Shiloh Drake, Cochair
Department of Linguistics, University of Oregon, 1290 University of Oregon, Eugene, OR 97403

Melissa Baese-Berk, Cochair
Linguistics, University of Chicago, 1115 E 58th St, Rosenwald Hall Room 203, Chicago, IL 60657

Chair's Introduction—8:00

Invited Papers

8:05

4aSCa1. "He speaks great English—For a guy from Moscow": Language ideologies in NHL media. Shiloh Drake (Dept. of Linguist., Univ. of Oregon, 1290 University of Oregon, Eugene, OR 97403, sdrake@uoregon.edu) and Melissa Baese-Berk (Linguist., Univ. of Chicago, Chicago, IL)

Ideologies about languages and countries are hard to shake, even in a multinational, multilingual setting like the National Hockey League (NHL) and the journalists who report on it. Despite its historical roots in Montréal and the dominance of Canadian and European players, the lingua franca of the NHL is English. In this work, we used qualitative analyses to examine players', journalists', and

coaches' attitudes toward languages other than English used on the ice. Across all groups, we found that Russian speakers were most likely to be assessed negatively, from being taciturn and unwilling to be interviewed (Frederickson, 2023) to being unlikely to speak good English (Keefe, 2023). Additionally, English-speaking players were more likely to associate positive sentiments with native North American English, Swedish, and Finnish speakers, but negative sentiments about Canadian French speakers and players from eastern European countries. Players and coaches also tended to be split on whether it was acceptable for other languages to be spoken in the locker room and on the ice. This work points to a fragmented in-group view of the acceptable language to use in professional hockey.

8:25

4aSCa2. Acoustics and ice hockey: The sociophonetic impact of Canadian English on American-Born players. Andrew R. Bray (Linguist., Univ. of Rochester, 513 Lattimore Hall, Rochester, NY 14626, andrew.bray@rochester.edu)

My research utilizes sociophonetic analysis to document the linguistic identity construction process that is ongoing in the sport of ice hockey. I argue that American-born players are constructing a hockey-based identity influenced by Canadian English (CE) due to the historical Canadian dominance of the sport. This identity incorporates Canadian Raising, FACE and GOAT monophthongization, both commonly attributed to CE and largely unexplainable based on players' regional dialects, and altered vowel production in hockey-specific terminology, most notably in the word *hockey* itself, unique to the hockey community. To document this variation, I analyze vowel formant values taken from sociolinguistic interviews with professional hockey players. I assess F1 and F2 values throughout vowel durations to establish if players are converging in production away from regional dialectal variants towards shared hockey-based variants. I argue these variants have gained indexical value linked to an emerging hockey-based identity that, although influenced by CE, is unique to the hockey community. In ongoing research, I aim to further document that this variation is most evident in hockey-specific terminology and that lexical diffusion occurs outwardly from these terms over time leading to players developing a more prevalent hockey-based identity as the sport gains more importance in their lives.

8:45

4aSCa3. There's no "I" in hockey: Identity work in hockey post-game interviews. Sarah Adams (Linguist., Univ. of Colorado Boulder, 433 Buchanan, University of Colorado Boulder, Boulder, CO 80310, saad1393@colorado.edu)

The community of practice represented by a professional sports team yields fascinating yet understudied sociolinguistic data on identity construction. Within the specific institutional context of hockey, the post-game interview between athlete and media personnel is a site for meaningful interactional analysis, where these interactions have tangible implications. Data are taken from the post-game interviews of the Colorado Avalanche professional men's ice hockey team during the 2020 postseason in Edmonton. Analysis of these interactions provides insight on the athlete's ability to use the interview to develop himself as a part of his team, or his "teamness." I identify the ways in which the team's culture is constructed and reinforced through specific linguistic practices. I further introduce the interviewer as a co-constructor who at once holds an agenda and makes space for the athlete to be agentive. The data illustrate the role of praise and blame, positioning evasion as an appropriate answer when the athletes avoid self-praise or blaming a teammate, as well as referent shifts and evaluation shifts, and passive constructions. I use research on turn design to analyze these practices and link the institutional role of the athlete to his appropriate performance of "teamness" through his use of language. For Speech Communication Technical Committee Best Student Paper Award.

9:05

4aSCa4. Sports and linguistics pedagogy. Robert Kennedy (Dept. of Linguist., Univ. of California, Santa Barbara, CA 93106-3100, rkennedy@linguistics.ucsb.edu)

The domain of organized sports offers a unique circumstance to explore a wide range of fundamental concepts in linguistics classrooms. We present an overview of a pedagogical model that engages students of linguistics using data from sports oriented settings, with frequent reliance on peer-reviewed literature. The course model incorporates three domains of language behavior: (a) lexis, and the interplay of semantic specialization and sociocultural indexation of specialized sport-oriented vocabulary; (b) sports announcer talk, a spoken broadcast register whose study enables a novel exploration of syntactic nuance; (c) participant interaction with each other and with the media, which demonstrates fundamentals of pragmatic inference in spoken communicative exchanges. Some course topics span more than one of these domains. Nickname creation, for example, is phonologically and phonetically driven, with gradient effects that adhere to constraints of brevity and clarity, yet linked to sociocultural indexicality. While no single textbook serves as an appropriate primary text for such a course, the beneficial outcomes for students are numerous. Aside from the primary goal of applying linguistic concepts to an engaging domain, students probe a variety of published works that examine sports/language interactions, demonstrating a broad sample of research methods and theories.

Session 4aSCb

Speech Communication: Speech Perception Poster Session II

Benjamin Munson, Chair
University of Minnesota, 115 Shevlin Hall, Minneapolis, MN MN

All posters will be on display and all authors will be at their posters from 10:00 a.m. to 12:00 noon.

Contributed Papers

4aSCb1. Differences in prominence-related acoustic measurements in L2 read speech as a function of L1 framing typology. Carissa A. Diantoro (Linguist., Univ. of Oregon, 1290 University of Oregon, Eugene, OR 97403, carissad@uoregon.edu), Hakyung Sung (Linguist., Univ. of Oregon, Eugene, OR), and Melissa M. Baese-Berk (Linguist., Univ. of Chicago, Chicago, IL)

Speech production in a non-native language is often influenced by their first language's (L1) phonology. One factor that might affect their production prosodically is typological differences in event construal, specifically motion events. Building on this concept, prior studies on gestures and spontaneous speech show that how speakers frame motion events in their L1 can transfer to their second language (L2) through grammatical patterns. However, no study has looked at how L1-L2 differences in framing motion events may affect speakers' L2 production prosodically. To explore whether those differences affect how L2 speakers prosodically produce motion events, this study collected read speech from two spoken corpora of L2 English learners. We chose L2 learners from different L1 backgrounds that differ in the way they grammatically construct motion events. English production from Korean and Turkish L1 speakers (which are typologically similar) were compared to that of L1 German speakers (which is typologically similar to English), along with productions by L1 English speakers. Duration and various acoustic measurements of pitch and intensity were extracted and analyzed. Results indicated some similarities among learners of the same typology along with differences between the L1 and L2 English speakers. Implications on conceptual transfer will be discussed.

4aSCb2. Distributional learning of non-native tone contrasts by older adults after training and overnight consolidation. Yin-To Chui (Div. of Humanities, The Hong Kong Univ. of Sci. and Technol., The Hong Kong University of Sci. and Technol., Clear Water Bay, Hong Kong, Hong Kong, ytchuiac@connect.ust.hk), Susu Lai, and Zhen Qin (Div. of Humanities, The Hong Kong Univ. of Sci. and Technol., Hong Kong, Hong Kong)

Despite decline in psychoacoustic and statistical learning (SL) abilities, older adults demonstrate remarkably intact perceptual learning in both L2 (tone-word learning) and L1 settings (perceptual adaptation to accented/noise-vocoded speech) but show limited transfer of learning to untrained stimuli. This study tests whether perceptual learning is maintained in an implicit statistical learning task where older adults learn L2 tonal contrasts through exposure to probability distributions of tonal tokens, which may pose higher requirements on both psychoacoustic and SL abilities, and whether sleep-dependent consolidation helps the generalization of perceptual knowledge. L1-Cantonese older adults learned to discriminate a perceptually difficult level-falling tone contrast following a pre-test, training, post-training overnight interval. Training stimuli were synthesized by interpolating naturally produced Mandarin level and high-falling tones into six equidistant steps. Participants either heard a bimodal (two-peak resembling level-falling categories) or unimodal distribution (single-peak) consisting of 256 tokens. ABX discrimination task was administered for testing, with tokens by two genders and on two pseudo-syllables to test generalization. Pilot data of 14 participants showed a trend of group effect with the bimodal group outperforming the unimodal group after training and sleep-dependent consolidation, showing that perceptual learning is maintained in a paradigm that relies heavily on psychoacoustic and SL abilities.

4aSCb3. Portuguese rhotic variation in the Brazilian cities of Salvador and São Paulo. Francis Jagiella (Lingusistics, Indiana Univ., 1020 E. Kirkwood Ave., Ballantine Hall 504, Bloomington, IN 47405, fjagiell@iu.edu)

Brazilian Portuguese has two rhotic phonemes: the alveolar flap /ɾ/ and the historically long version which previous publications call velar, uvular, or glottal fricatives, or alveolar trills and approximants. This variation occurs both within and across dialects. Deletion is also common, most notably in word-final position. For the current project, thirty-five participants from Salvador and ten participants from São Paulo were recorded reading predetermined stimuli of isolated words and sentences, creating 6,383 instances of the rhotic phoneme. Productions were classified as exhibiting deletion or for having voicing, frication, flapping, and place characteristics. The results indicate a range of surface forms of the phoneme more variable than previously cited, with palatal fricatives common in Salvador and several flap + fricative variants common in São Paulo, along with other less frequent forms. In Salvador, glottal fricatives predominate across the board with much higher rates of deletion. In São Paulo, glottal fricatives predominate in onset positions, but alveolar trills and approximants and flap + fricative variants predominate in coda position. While deletion is most common word-finally, it occurs in all environments where the phoneme is found.

4aSCb4. The production of tone and intonation in Mandarin-English bilingual children—A pilot study. Jie Yang (Commun. Disord., Texas State Univ., 200 Bobcat Way, Round Rock, TX 78665, j_y90@txstate.edu)

In addition to consonants and vowels, Mandarin Chinese uses two levels of prosodic contrast, tone at word level and intonation at utterance level, for linguistic message generation. When tone and intonation interact, Mandarin speakers use global and/or localized fundamental frequency (f0) adjustments to maintain linguistic intelligibility at both word- and utterance-level. Word-level tonal contrast does not exist in English. For English speakers, f0 changes at utterance level for intonation were not confined by word-level tonal requirements. The present study investigated how Mandarin-English bilingual Children manage these two levels of prosodic contrast within the respective phonology at different developmental stages. Six- and nine-year-old Mandarin-English bilingual children and young adults (seven per group) completed the speech production tasks. Stimuli were carrying sentences (statements and questions) ending with monosyllabic target words that are phonetically similar in Mandarin and English (e.g., [li] 丽, Lee). The carrying sentences were statements with falling intonation, questions with and without inversion in English, and questions with and without question particle (吗) in Mandarin. Average f0 and magnitude of f0 change were measured within words and over utterances and compared between languages and among age groups. Results indicated the influences of language and

developmental age. Work supported by Texas State University Research Enhancement Program.

4aSCb5. Stroop task response times: The bilingual advantage. Kathleen Siren (Speech-Language-Hearing Sci., Loyola Univ. Maryland, 4501 N. Charles St., Baltimore, MD 21210, ksiren@loyola.edu), Bridget Killmurray (Speech-Language-Hearing Sci., Loyola Univ. Maryland, Baltimore, MD), Sarah Bayer (UNC-Chapel Hill, Chapel Hill, NC), Sophia Launay-Fallasse, Meghan Stapp, and Tepanta Fossett (Speech-Language-Hearing Sci., Loyola Univ. Maryland, Baltimore, MD)

Bilingual individuals often must focus on one language while suppressing interference of another language and may therefore develop a specialized capacity to selectively attend to the language currently being used, a form of cognitive control. For this investigation, monolingual and bilingual individuals completed a 40-item modified Stroop task. In this task, participants were shown incongruent visual stimuli, with the names of various colors presented in a different ink color. Participants were instructed to respond verbally by naming the ink color instead of reading the written color word. Investigators scored verbal responses for accuracy and measured speed of response. Additionally, investigators measured duration of selected responses. Results demonstrate that monolingual and bilingual individuals were similarly accurate on the task. However, on average, bilingual speakers were faster at responding than monolingual speakers. Further, bilingual speakers were less variable in response time with responses that were shorter in duration than monolingual speakers. These results are discussed with reference to theories of differential brain development and cognitive function in individuals who simultaneously acquire more than one language.

4aSCb6. Age and category structure in phonetic category learning. Christopher C. Heffner (Communicative Disord. and Sci., Univ. at Buffalo, 122 Cary Hall, South Campus, Buffalo, NY 14214, ccheffne@buffalo.edu)

When learning new speech sound categories in a second language, listeners must decide which sounds belong to the same category and which belong to different categories. This learning process is contingent on perceptual systems that change across the lifespan, which may, in turn, make certain categories easier to learn depending on which systems are employed by learners. In the present study, learners across a variety of ages categorize speech sounds taken from German that vary in their category structure (i.e., the complexity of the categories being learned within the stimulus space). Participants aged 7 to 70 are recruited and run at community sites such as museums and libraries. The preliminary dataset indicates that the simpler category structure saw strong age effects (with poorer performance on the younger and older ends of the age range and better performance at intermediate ages), while differences across ages were much smaller for the more complex category structure. These findings suggest that learning simpler category structures may rely on age-dependent learning systems, while learning more complex categories may rely on systems that are less age-dependent.

4aSCb7. Relating assimilation data to identification data in second language learners. Kenneth J. de Jong (Dept. of Linguist., Indiana Univ., Dept. of Linguist., Ballantine Hall, Bloomington, IN 47405, kdejong@indiana.edu), Yu-Jung Lin (Foreign Lang., Literatures, and Cultures, College of the Holy Cross, Worcester, MA), Yen-Chen Hao (World Lang. and Cultures, Univ. of Tennessee, Knoxville, TN), and Hanyong Park (Linguist., Univ. of Wisconsin-Milwaukee, Milwaukee, WI)

Commonly cited models of bilingual phonology, e.g., the Speech Learning Model and the Perceptual Assimilation Model, project hypotheses about second language (L2) developmental patterns based on hypothetical assimilation patterns between categories in the learner's first language (L1) and L2, with the addition of a learning component which differentiates an L2 and L1 system. This paper relates two experiments with two participant groups, one native Korean, and one native Taiwan Mandarin. Both groups identify English obstruents /p b t d f v θ ð/ in four prosodic positions (onset, coda, and intervocalic pre-stress and post-stress) using their native non-Roman graphemes in the first experiment, and using English labels in the second. The assimilation results generate predictions for discrimination performance in the second experiment, assuming only the L1 categories. The two groups show large differences in assimilation pattern due largely to the non-sibilant fricative /f/ in Mandarin, in opposition to the Korean system which has only non-sibilant /h/. Analyses reveal a strong correlation between L2 discrimination patterns and predictions based solely on the L2-L1 assimilation results, suggesting that the perceptual system of L2 learners can be surprisingly well-captured as a single perceptual system with a bilingual array of segmental categories.

4aSCb8. The perception of (trans)masculine speech: Effects of stimulus acoustics and rater identity. Benjamin Munson (Speech-Language-Hearing Sci., Univ. of Minnesota, 115 Shevlin Hall, Minneapolis, MN MN, munso005@umn.edu) and Devin V. Dolquist (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN)

A large body of work has examined the acoustic characteristics of cisgender, heterosexual (cishet) men and women's speech, and the acoustic features that predict listeners' judgments of binary gender through speech. Considerably less work has examined the diverse ways that non-cishet men, non-cishet women, and people whose gender is neither exclusively male nor exclusively female express gender through speech. Moreover, there is relatively little work on how gender-diverse listener groups perceive gender through speech. This poster presents the results of a perception experiment designed to fill those gaps. Stimuli were multiple sentence productions of 20 masculine-presenting individuals, including cisgender men, transgender men, and transmasculine nonbinary people. These are described and analyzed acoustically in Dolquist (2023). Listeners were 88 cishet men, 62 cishet women, and 50 gender and sexuality diverse (GSD) individuals. Listeners identified gender identity, gender orientation (i.e., cisgender or transgender), and a variety of attributes for each stimulus. GSD listeners were more likely than cishet men and cishet women to identify voices as transgender or nonbinary, and were more likely to evaluate those voices favorably. Moreover, cishet men and women's judgments of gender were more strongly predicted by sex-dimorphic acoustic characteristics like f0 than were GSD listeners' judgments.

4aSCb9. Racialization and sentence intelligibility in older adults with hearing impairment. Benjamin Munson (Speech-Language-Hearing Sci., Univ. of Minnesota, 115 Shevlin Hall, Minneapolis, MN MN, munso005@umn.edu), Tatiana Lyons, Matthew B. Winn (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN), Peggy Nelson (Ctr for Applied/Translational Sensory Sci., Univ. of Minnesota, Minneapolis, MN), Enengy Schutt, and Alayo Tripp (Speech Lang. Hearing Sci., Univ. of Minnesota, Twin Cisties, Minneapolis, MN)

Previous studies have shown that the perception of a talker's racial identity affects the intelligibility of that person's speech to younger, normal-hearing listeners (Babel & Russell, 2015; McGowan, 2015; Tripp, Lyons, & Munson, 2022). These findings show that social perception influences intelligibility. These effects are thought to reflect varied experiences interacting with racially diverse individuals, attitudes and beliefs about different racial groups, or a combination of these factors. The current experiment examines whether the strength of the influence of perceived racial identity on intelligibility is similar across different age groups and levels of hearing acuity. Participants (n = 42, including 16 normal hearing [NH] younger adults, 11 NH older adults, and 15 older adults with hearing loss) were played sentences produced by four talkers who varied in racial identity, in audio-only and audiovisual conditions, in background noise and reported what they heard. Participants also completed a measure of attitudes toward the four talkers. Analyses are ongoing, and examine (1) whether the influence of presentation modality on intelligibility varies across the four talkers, (2) whether these talker differences in audiovisual benefit vary across the three listener groups, and (3) whether attitudes toward the talkers predict intelligibility. [Funding: NIH grant R21 DC018070]

4aSCb10. Preceding word information for predicting speech errors in English as foreign language speech. Ki Woong Moon (Linguist., Univ. of Arizona, 1200 E University Blvd, Douglass 318B, Tucson, AZ 85721, kiwoongmoon@arizona.edu)

Speech errors, including disfluency errors (e.g., filled pauses ("uh", "um"), repetition ("I me-mean right now.")), and mispronunciation of speech segments (e.g., "think" as /sɪŋk/) are natural occurrences in speech production and they can affect speech fluency and proficiency. Detecting these errors is important, especially in assessing second language (L2) learners. Non-native speakers often produce speech errors, even in read speech, due to increased cognitive load when simultaneously producing the current word and processing the upcoming word. By analyzing two L2 speech corpora having different types of errors (disfluency and mispronunciation), the study aims to assess the effectiveness of incorporating preceding word information in speech error prediction. The results indicate that longer syllable duration in the word preceding the mispronunciation error correlates with error production. The study extends its investigation to identify acoustic and lexical features influenced by the presence of speech errors. The study found that speech errors increase duration of speech. The study also found that words with more syllables may cause L2 speakers to produce speech errors.

4aSCb11. Quantitative assessment of English accent variation: Levenshtein distance and perceptual identification. Lian J. Arzbecker (Communicative Disord. and Sci., Univ. at Buffalo, 103D Cary Hall, Buffalo, NY 14207, arzbecker.1@osu.edu) and Ewa Jacewicz (Speech and Hearing Sci., The Ohio State Univ., Columbus, OH)

This is a perceptual study with a basis in acoustic phonetics. It investigates the extent to which accents of English can be quantified by the Levenshtein distance (LD), a string metric. LD defines string similarity by the minimum number of character edits required to transform one string into another. Narrowly transcribed speech samples of English provide the phonetic strings required for LD comparison. For the purposes of the current experiment, 24 speakers representing four English accent varieties are included: Midland American (control), British/Australian, Hindi-influenced, and Mandarin-influenced. High- and low-pass filters are used to augment or attenuate perceptual contribution of high- versus low-frequency information in the speech signal. Predictions stem from published correlations between LD and listeners' perceptional ratings of intelligibility and native-likeness. For a group of monolingual American English listeners, confusion is expected between Midland American and British/Australian accents due to similarly low LDs, further intensified by low-pass filtering. However, fewer confusions are predicted between Hindi- and Mandarin-influenced English, due to varying L1 influences and the absence/presence of tonal information. This research seeks to explore accent perception, investigating the relationship between manipulated aspects of the speech signal and listener identification.

4aSCb12. Beyond visual primes: Audiovisual integration of racial identity and sentence intelligibility. Eleanor Nickel (Sociology, Univ. of Minnesota, 909 Social Sci. Bldg., 267 19th Ave. S, Minneapolis, MN 55455, nicke288@umn.edu), Alayo Tripp (Linguist., Univ. of Florida, Minneapolis, MN), Enengy Schutt, and Benjamin Munson (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN)

There is no biological basis for racial categories. Yet, the presentation of racialized stimuli impacts psycholinguistic processing, suggesting that a talker's racial identity affects measures of intelligibility (Babel & Russell, 2015; McGowan, 2015). Previous work has used static visual primes, presenting images accompanied by acoustic sentence stimuli. Static visual primes may prime expectations of the person's speech, with unexpected pairings decreasing intelligibility. These findings show that social expectations affect speech intelligibility and have broad implications for real-world speech perception in high-stakes educational (Evans, Munson, & Edwards, 2017), social (Purnell, Baugh, & Idsardi, 1999), and legal (Rickford & King, 2016) contexts. Expanding our understanding of the psycholinguistic and psychophysical mechanisms behind these phenomena requires experimentation with dynamic, audiovisual speech rather than static visuals. The current experiment presents conflicting visual and auditory cues indexing the white and Black racial identities of the talkers. The talker who is seen (the 'visual' talker) and the racial identity of the talker who is heard (the 'audio' talker) are fully crossed. This design allows us to examine how intelligibility is influenced by visual racial identity, audio racial identity, and the audiovisual (mis)match between them. Data collection is currently in progress and includes groups of both white and Black listeners.

4aSCb13. Revisiting rapid accent adaptation with computational modeling. Samantha Chiu (Psychol. Brain Sci., Univ. of Iowa, 340 Iowa Ave. Iowa City, IA 52242, samantha-chiu@uiowa.edu), Leo Moore, and Ethan Kutlu (Dept. of Linguist., Univ. of Iowa, Iowa City, IA)

Studies on accent adaptation report that monolinguals can rapidly adapt to a novel foreign accent with short exposure or training (Baese-Berk *et al.*, 2013; Clarke & Garrett, 2004; Maye *et al.*, 2008). While most of these studies rely on self-reported monolinguals, there is great variability in how much language diversity monolinguals encounter regularly (Castro *et al.*, 2022). We return to this question of rapid accent adaptation in monolinguals through computational modeling where linguistic diversity is directly controlled. We simulate monolingual speech perception using PyTorch's Wav2-Vec2 model (Paszke *et al.*, 2019), pre-trained on the Librispeech corpus with American-accented English (Panayotov *et al.*, 2015). We replicate the experimental design of Baese-Berk *et al.* (2013) by fine-tuning this model on a single accent ($n = 150$ sentences across 5 talkers) and multiple accents ($n = 30$ sentences for each of 5 talkers). Preliminary results find that exposure to a single accent (68.6% correct) or multiple accents (69% correct) does not induce accent adaptation to a novel accent (no accent training = 69% correct). We predict that exposure to a single or multiple accents will increase accuracy but requires many additional hours of exposure. We discuss implications of our models against accent adaptation studies.

4aSCb14. Using computational models to study accent adaptation: A tutorial. Leo Moore (Linguist., Univ. of Iowa, 340 Iowa Ave. Iowa City, IA, leo-moore@uiowa.edu), Samantha Chiu (Psychol. Brain Sci., Univ. of Iowa, Iowa City, IA), and Ethan Kutlu (Linguist., Univ. of Iowa, Iowa City, IA)

One of the limitations in accent perception research is the difficulty of quantifying how much exposure is needed to observe adaptation to a novel accent. While numerous studies have investigated this issue (Baese-Berk *et al.*, 2013; Clarke & Garrett, 2004), it has been impossible to control for individual-level differences and solely focus on linguistic differences with human participants. However, computational models can be intentionally set in terms of these features to better understand the process of adaptation to novel accents. Here we discuss how machine learning models can be implemented using PyTorch, a framework in Python made by Meta. PyTorch allows for the easy creation of language learning models that can be used in speech-recognition experiments (Paszke *et al.*, 2019). Here we use the Wav2Vec2 model created by Meta, pre-trained on 960 hours of the Librispeech ASR corpus (Panayotov *et al.*, 2015). We discuss importing the model and fine-tuning it using samples of different accents. Next, we demonstrate building a decoder using a greedy algorithm to transform the model's classification into a transcript. Our goal is to compare simulations against data from human participants and tease apart the direct effects of individual variability on accent perception.

4aSCb15. Intelligibility of British and American English in different listening conditions. Elizabeth D. Young (Commun. Sci. and Disord., Univ. of Utah, 390 S 1530 E, Rm. 1218, Salt Lake City, UT 84112, liz.d.young@utah.edu), Kayleena Faulkner, Isabella McHugh, and Sarah H. Ferguson (Commun. Sci. and Disord., Univ. of Utah, Salt Lake City, UT)

Communication specialists frequently hear complaints from patients that unfamiliar accents are more difficult to understand, particularly in background noise. Previous work (Schilaty *et al.*, 2021) has indeed shown that when presenting American listeners with both American and British accents, intelligibility in noise is poorer for the unfamiliar accent across age and hearing status. However, Schilaty *et al.* (2021) examined relatively easy sentence materials (Basic English Lexicon [BEL] sentences; Calandruccio & Smiljanic, 2012) in only one type of noise. The current study expands on previous work by comparing the intelligibility of American and British accents for a more difficult sentence set (Harvard sentences; Rothauser *et al*., 1969) in young adult American listeners across two noise types (speech-shaped noise and 12-talker babble). Stimuli consisted of 180 Harvard sentences produced by two male talkers: one with an American (Northwestern) dialect and one with a Southern British dialect. Young adult listeners with normal hearing transcribed the sentences in quiet, in speech-shaped noise at -3 dB SNR, and in 12-talker babble at -3 dB SNR. Listeners also completed a dialect familiarity questionnaire. The results will shed additional light on the interactions between talker and listener dialect, dialect familiarity, and background noise type.

4aSCb16. Perceptual representation of speaker gender in Spanish-English bilingual listeners. Brandon Merritt (Speech, Lang., and Hearing Sci., The Univ. of Texas at El Paso, 734 S. Mesa Hills Dr., #72, El Paso, TX 79912, bmmerritt@utep.edu)

The perceptual representation of speaker gender in monolingual English listeners has been found to be gradient and 2-dimensional. However, both gender expression and listener attribution of a speaker's gender are known to vary by cultural and linguistic norms. Thus, listeners' attribution of a speaker's gender is expected to vary based on the cultural and linguistic practices of their community. This study examined the perceptual representation of speaker gender in Spanish-English bilingual listeners at the U.S./Mexico border of El Paso, TX. Twenty-four Spanish-English bilingual speakers of diverse gender identities (e.g., cisgender men, cisgender women, and transgender women) were audio recorded reading sentences from the English and Spanish versions of the Hearing in Noise Test. Nineteen cisgender Spanish-English bilingual listeners completed an auditory free classification paradigm, in which they classified speakers by perceived general similarity and gender identity in both Spanish and English conditions. Multidimensional scaling of listeners' classifications in each language revealed that listeners organized speakers in a more expansive perceptual space in English (3 dimensions) as compared to Spanish (2 dimensions). Dimension weightings indicated that, in Spanish, listeners placed more emphasis on Dimension 1 as compared to Dimension 2 when classifying speakers, while, in English, listeners equally weighted the 3 dimensions.

4aSCb17. Effects of quality of initial exposure stimuli on speech intelligibility. Seung-Eun Kim (Linguist., Northwestern Univ., 2016 Sheridan Rd., Evanston, IL 60208, seungeun.kim@northwestern.edu), Matthew Goldrick (Linguist., Northwestern Univ., Evanston, IL), and Ann Bradlow (Linguist., Northwestern Univ., Evanston, IL)

Listeners can use lexical information to drive adaptation to talker-specific speech characteristics with a small amount of exposure. This suggests that adaptation may be strongest when listeners are initially exposed to speech in a condition that facilitates word recognition. This was tested by examining the intelligibility of second-language (L2) speech-in-noise when stimuli were initially presented in Quiet versus low signal-to-noise ratio (SNR) conditions. Recordings of 10 L2 English talkers were tested. For each talker, 120 sentences were mixed with speech-shaped noise ranging from -4 dB to 8 dB in steps of 2 dB and in Quiet (15 sentences at each SNR). These stimuli were presented from lowest to highest SNR (forward; -4 dB to Quiet) in one group of L1 English listeners ($n = 10$/talker) and from highest to lowest SNR (reverse; Quiet to -4 dB) in the other ($n = 10$/talker). The results showed that early exposure to a clean signal led to better speech perception. At lower SNRs, word recognition accuracy in the reverse order was significantly higher than the forward order, resulting in a steeper slope of psychometric function of intelligibility in the forward versus reverse condition. The quality of initial speech input may influence talker adaptation, particularly in adverse listening conditions.

4aSCb18. Non-native clear speech increases intelligibility but not through improved word segmentation: Evidence from a visual-world eye-tracking study. Madeline Smith (Dept. of Linguist., Univ. of Texas at Austin, 305 E. 23rd St., Stop B5100, Austin, TX 78712, madelinesmith@utexas.edu), Zhe-chen Guo (Dept. of Commun. Sci. and Disord., Northwestern Univ., Evanston, IL), and Rajka Smiljanic (Dept. of Linguist., Univ. of Texas at Austin, Austin, TX)

Listener-directed hyperarticulated clear speech produced by native (L1) talkers improves word segmentation and reduces lexical competition. Less is known about whether non-native (L2) clear speech also confers such benefit. In a visual-world eye-tracking study, we investigated if L2 clear speech improves word segmentation and the time course of the benefit for native listeners. Forty L1 English participants heard sentences produced in conversational and clear styles by a highly intelligible L2 English / L1 Spanish speaker with a discernable non-native accent. Sentences contained a target word (e.g., *doll*) with which a corresponding competitor overlapped phonemically (e.g., *dolphin*), creating temporary ambiguity with the target and the following word's onset (e.g., *doll found*). Each recording was presented in quiet alongside pictures of the target, competitor, and two distractors. Participants were instructed to select the picture mentioned in the sentence they heard. No significant clear speech segmentation advantage was found; the proportion of looks to targets over competitors indicates similar time course of disambiguation in both conversational and clear speech. The results suggest that L2-accented clear speech with its deviations from the target-language-specific modifications and greater phonetic variability increases signal uncertainty resulting in no benefit for word segmentation even though word recognition was improved.

4aSCb19. Assessing retroflex and alveolar liquid perception and production in heritage Tamil speakers. Kamala Muthukumarasamy (Carleton Univ., 1125 Colonel By Dr., Ottawa, ON K1S 5B6, Canada, kamalamuthukumarasam@cmail.carleton.ca) and Chandan Narayan (York Unversity, Toronto, ON, Canada)

In this presentation, we explore factors affecting and the relationship between perception and production of the alveolar-retroflex liquid contrast ([l]-[èÙ]) in heritage Tamil speakers. In particular, we examine the connection between language and identity in the maintenance of this acoustically fragile contrast with similar spectral characteristics, shown through low perceptual salience. Tamil speakers ($n = 18$) completed an AX discrimination task with non-word Tamil VCVs was administered. D-prime, a bias free measure of perceptual distance, was computed from the discrimination data. Additionally, participants provided minimal pairs with the target consonants in an elicited production task. An F3-F2 (Hz) score was taken as a measure of productive salience, with alveolars having a larger difference than retroflexes. Quantitative results showed a high degree of variation in productive salience, as some speakers clearly produce the contrast while others did not. Perceptual distance was also variable, with some participants clearly showing categorical discrimination while others did not. Qualitative results revealed that a concrete ties to tangible culture and a strong linguistic identity can serve as an indicator of accuracy in perception and production. This research addresses whether acoustically fragile contrasts are realized in heritage Tamil and provides new insight into heritage phonological contrast retention and maintenance.

4aSCb20. Durational and spectral factors in judgements of American Raising. Elliott Moreton (Linguist., Univ. of North Carolina, Chapel Hill, Dept. of Linguist., CB #3155, Chapel Hill, NC 27599-3155, moreton@email.unc.edu), Jeff Lamontagne (French and Italian, Indiana Univ., Bloomington, IN), and Monica Nesbitt (Linguist., Indiana Univ., Bloomington, IN)

Canadian Raising and its relatives in the USA can respond to underlying voicing of flapped /t/ (Raised "writer" vs. un-Raised "rider") from the earliest stages (Fruehwald 2016). How so, if Raising is phonologized from a phonetic precursor sensitive only to phonetic features? Proposal (Bermúdez-Otero 2019): Raising responds to duration, not voicing: diphthongs are shortened before underlyingly voiceless sounds by a pre-existing lexical phonological rule of Pre-Voiceless Clipping, after which postlexical Raising transparently affects the shortened diphthongs. Experiment: Participants

(N = 141, US dialects) read wordlists with /ai/ and /ei/ in voiceless and voiced contexts, then sorted the words into groups judged to share a vowel (DiPaolo & Faber, 1990). Clipping and Raising were measured using duration, F1, and F2. Predictions: (1) across speakers, /ai/-Clipping and /ai/-Raising should be positively correlated. (2) /ai/-sorting should be better predicted by /ai/-Clipping than by /ai/-Raising, because lexical rules change phonemes, while postlexical ones are subphonemic. (3) /ei/-sorting should be positively correlated with /ai/-sorting, since Clipping affects all vocoids. Results: (1) /ai/-Clipping did *not* predict /ai/-Raising ($r = -0.082$); (2) /ai/-Clipping predicted word sorting marginally *worse* than /ai/-Raising (95% CI for $r_{clipping}$-$r_{raising} = (-0.50, 0.03)$). (3) /ei/ and /ai/ judgements *were* positively correlated ($r = 0.37$, 95% CI = (0.21, 0.51)).

4aSCb21. Perceptual adaptation to unfamiliar dialects in an eye-tracking task. Larisa Bryan (Ohio State Univ., 1712 Neil Ave., Columbus, OH 43210, bryan.368@osu.edu) and Cynthia G. Clopper (Ohio State Univ., Columbus, OH)

Variation in pronunciation of vowels across American English dialects often creates the strongest distinctions between dialects and leads to lexical ambiguity between minimal pairs within dialects. This lexical ambiguity is stronger for listeners with lifetime exposure to multiple dialects than to a single dialect. In the current study, we investigated the influence of brief, in-lab dialect exposure on lexical competition and perceptual adaptation. Midwestern American English participants were first presented with a familiarization short story told in one of two unfamiliar dialects (Southern American English or a novel accent) and then completed a visual-world eye-tracking task containing both acoustically ambiguous and non-ambiguous words from each dialect in a forced-choice task. The results showed evidence of greater lexical competition for target dialect minimal pairs following familiarization. This pattern of results is consistent with previous work suggesting greater lexical competition following lifetime exposure to dialect variation among geographically mobile listeners, suggesting similarities in the effects of brief in-lab and lifetime exposure to dialect variation on lexical competition and processing.

4aSCb22. How rhythmic cues influence perception of accent distance across L1 and L2 English speakers. Holly Lind-Combs (Speech and Hearing Sci., The Ohio State Univ., 1070 Carmack Rd., Columbus, OH 43210, lind-combs.1@osu.edu), Rachael F. Holt (Speech and Hearing Sci., The Ohio State Univ., Columbus, OH), Tessa Bent, and Malachi Henry (Speech, Lang. and Hearing Sci., Indiana Univ., Bloomington, IN)

Segmental and suprasegmental cues can both contribute to listeners' accent strength and distance judgments. This study examined the influence of a suprasegmental cue—rhythm—on listeners' judgments of accent distance. Rhythm was quantified using two measures of variability in vocalic interval duration: normalized pairwise variability index (nPVI) and VarcoV. Thirty adults ranked productions from female and male speakers from six L1 and six L2 English varieties (24 speakers) across six sentences based on their perceived distance from Midland American English (the local dialect). Linear mixed effect models predicted accent distance ratings from rhythm measures—specifically, differences in nPVI and VarcoV between the L1/L2 speakers and Midland speakers, with speaking rate and sentence length as covariates. Differences between L1/L2 speakers' and Midland speakers' nPVI and VarcoV scores did not significantly predict accent distance rankings. However, significant interactions between nPVI and certain accents (e.g., Irish, Southern American, Cantonese-accented English) suggest that rhythmic properties may significantly impact accent distance judgements for some accent varieties. These results provide further insight into how rhythmic cues contribute to accent perception. [Work supported by NSF grants 1941691 and 1941662].

4aSCb23. The contribution of suprasegmental cues to perceived L1 and L2 accent distance. Malachi Henry (Speech, Lang. and Hearing Sci., Indiana Univ., 1579 S Renwick Blvd, Bloomington, IN 47401, mjhenry5519@gmail.com), Tessa Bent (Speech, Lang. and Hearing Sci., Indiana Univ., Bloomington, IN), Rachael F. Holt, and Holly Lind-Combs (Speech and Hearing Sci., Ohio State Univ., Columbus, OH)

First (L1) and second language (L2) pronunciations that diverge more segmentally from the local accent are rated as more distant. However, fewer studies have addressed relations between suprasegmental features, such as intonation, and accent distance. We examined the relation between perceived accent distance and four suprasegmental measures: time normalized f0 Euclidean distance, mean F0, pitch range, and speaking rate. Thirty adults completed six ladder tasks in which listeners ranked talkers' perceived distance from the local accent. For each ladder, listeners heard one sentence produced by 24 talkers: one man and one woman from six L1 and six L2 English varieties. Each suprasegmental measure independently predicted distance rankings with a variety of significant interactions. Talkers with higher accent rankings had narrower pitch ranges, lower mean f0, faster speech, and greater f0 Euclidean distance. Furthermore, suprasegmental measures related to accent distance differentially based on syntactic structure. For example, as f0 Euclidean distance increased, accent rankings increased for interrogative sentences, but not for declarative sentences. These data support the need for future studies examining how suprasegmental features impact perception of L1 and L2 accents. [Work supported by NSF grants 1941691 and 1941662].

4aSCb24. Social network characteristics impact the recognition of degraded speech by adult cochlear implant users. Terrin N. Tamati (Dept. Otolaryngol., Vanderbilt Univ. Medical Ctr., 1608 Aschinger Blvd, Columbus, OH 43212, terrintamati@gmail.com), Emily M. Clausing (Ohio State Univ., Columbus, OH), and Victoria A. Sevich (Speech and Hearing Sci., The Ohio State Univ., Columbus, OH)

Social networks influence the quantity and quality of linguistic input experienced in everyday listening environments. In normal-hearing listeners, the variable linguistic input provided by larger and more diverse social networks has been shown to support speech processing. However, for adult cochlear implant (CI) users, limitations in the perception of linguistic and talker details may limit input variability benefits. The current study examined the effects of social network size and diversity on the recognition of spectrotemporally degraded speech by adult CI users. Twenty-six postlingually deafened adult CI users completed a detailed questionnaire about their regular communication partners. Social network size was calculated as the number of regular communication partners, and social network diversity was calculated as the degree of age, education, and accent heterogeneity among communication partners. Social network metrics were compared to vowel, word, and sentence recognition accuracy scores, also controlling for basic auditory ability. Results showed that social network age diversity was moderately to strongly correlated with word and sentence recognition accuracy. Social network size and other diversity metrics were not related to word or sentence recognition accuracy. These findings suggest that more diverse input from different age groups facilitates spoken word recognition in adult CI users.

4aSCb25. Neural oscillation to music processing in children with different language backgrounds. Maxfield Rodgers, Kristal Reyes, Faith Chai, Blessy Gill, Angela Cheng (St. John's Univ., Queens, NY), and Yan H. Yu (St. John's Univ., 4631 216 St., Bayside, NY 11361, yanhyu@gmail.com)

Musical experience and tonal language experience are both associated with structural and functional changes in the brain that underlie facilitated pitch perception. We investigated whether neuronal oscillations in response to music were correlated with musical expertise and whether they reflected the language backgrounds (tonal versus nontonal, monolingual versus bilingual). We measured music processing in bilingual children (5–10 years old) from Mandarin (a tone language) households and three groups of age-matched children from non-tone language households (Bilingual Spanish-English, monolingual mainstream American English (MAE), and African American English (AAE)). Event-related brain potentials (ERPs) were

recorded in an oddball paradigm with six types of music changes. We analyzed the phase locking and amplitude modulations of ongoing oscillations in the theta (4–8 Hz) alpha (8–14 Hz), beta- (14–30 Hz), and gamma- (30–80 Hz) bands to music changes. Preliminary results suggest that musical expertise was associated with strengthened phase locking of neural oscillations to most music features. Tonal language experience was also associated with more robust phase locking. Bilingual experience alone does not show any enhancement in neural oscillation. There were no clear advantages of music processing for the bilingual experience.

4aSCb26. Fundamental frequency patterns across gender, task, and language in Spanish-English bilinguals. June M. Contreras (Speech Lang. and Hearing Sci., Univ. of Texas at El Paso, 3532 Keltner, El Paso, TX 79904, jmcontreras2@miners.utep.edu) and Brandon Merritt (Rehabilitation Sci., The Univ. of Texas at El Paso, El Paso, TX)

Speaking fundamental frequency (SFF) patterns are known to vary by language and speaker gender. Relatively little data exist for SFF patterns among cisgender and transgender speakers, especially those who speak languages other than English. Further, speaking task (e.g., read versus spontaneous speech) may impact SFF patterns. This study examined SFF patterns in Spanish-English bilinguals of varying gender identities. Twenty-four speakers (8 cisgender men, 8 cisgender women, and 8 transgender women) recorded a read passage and spontaneous speech in Spanish and English. SFF measures of minimum, maximum, range, and median were found to be stable across speech tasks and languages. A significant effect of gender was found. Cisgender women on average produced the highest values of minimum, maximum, and median SFF and largest SFF range. Cisgender men produced the lowest values and smallest range. Transgender women produced SFF measures within an intermediate range of cisgender men and women. Compared to monolingual English speakers, Spanish-English bilingual cisgender women had a larger SFF mean for both Spanish and English, cisgender men had similar SFF means for both Spanish and English, and transgender women had a slightly larger SFF mean for both Spanish and English.

4aSCb27. Intelligibility, recall, and voice evaluation across accents. Line Lloy (Linguist., Univ. of BC, Vancouver, BC, Canada), Nikolai A. Schwarz-Acosta (Spanish and Portuguese, Univ. of California - Berkeley, Emeryville, CA), and Molly Babel (Linguist., Univ. of BC, 2613 West Mall, Vancouver, Vancouver, BC V6T 1Z4, Canada, molly.babel@ubc.ca)

Speech elicits variable responses from listeners. Voices can vary in their intelligibility, how well listeners can recall messages produced by the voice, and what kind of social evaluation it explicitly or implicitly evokes. These different responses may be due to individual-specific attributes within a voice or the accent it carries. For example, familiar or more standard language varieties may be more intelligible producing more easily recalled, and eliciting more positive social evaluations from listeners. The current study uses 35 English-speaking voices from 7 different language backgrounds that vary in familiarity and prestige to the listener ($n = 430$) population, which is a representative heterogeneous sampling from the local university community. Specific voices were chosen from a larger data set based on their acoustic similarity. Listeners either completed a speech transcription task (quantifying intelligibility) or a cloze task (quantifying recall) and all listeners provided an evaluation of the voices' likability and perceived comprehensibility. Bayesian data analysis is used to quantify and characterize the relationship between a voice's intelligibility and how well it is recalled, and whether this relationship is predicted by social evaluation and listener experience. These results have implications for theories of speech recognition and how listeners process accents.

4aSCb28. The Accent Atlas: Effects of long-term exposure to nonnative accents on adaptive speech perception. Yuting Gu (Univ. of California Irvine, Irvine, CA), Seth Cutler, Chigusa Kurumada (Univ. of Rochester, Rochester, NY), and Xin Xie (Univ. of California Irvine, SSPB 2223, University of California Irvine, Irvine, CA 92617, xxie14@uci.edu)

As mobility increases and virtual interactions grow, quickly adapting to diverse accents is essential for effective communication. While lab-based accent training promotes rapid adaptation to unfamiliar accents, whether prolonged environmental exposure to linguistic diversity yields similar benefits is unknown. We conducted a large-scale perceptual experiment involving 600 + participants across 15 U.S. states, utilizing a cross-modal word matching task to measure speech recognition in both non-native accented English and native English with background noise. Participants were recruited from linguistically diverse (LD) or linguistically homogeneous (LH) states. Linguistic diversity was defined based on the prevalence of non-English speaking residents (U.S. census data). Our findings were two-fold: firstly, consistent with prior studies, only exposure to non-native accented speech—but not exposure to speech in noise—improved recognition of novel talkers with the same accent; secondly, individuals from the LD states outperformed those from LH areas in initial accented speech perception. However, by the end of the experiment, benefits from in-lab accent exposure did not differ significantly between the two groups, suggesting that short-term training could mitigate long-term environmental influences. These results illuminate the interplay between daily linguistic environments and accent adaptation, enhancing our understanding of speech perception's adaptivity in a multilingual context.

4aSCb29. Perception of speaker identity for bilingual voices. Sylvia Cho (Linguist., Simon Fraser Univ., 8888 University Dr., Burnaby, BC V5A 1S6, Canada, sylvia_cho@sfu.ca)

Voice is often described as an "auditory face"; it provides important information concerning speaker identity (e.g., age, height, sex). The acoustic properties related to voice can also vary substantially within a speaker based on one's emotional, social, and linguistic states. Recent work suggests that biological components have the greatest impact in the acoustic variability found in voice, followed by language-specific factors and speaking style [Lee & Kreiman, J. Acoust. Soc. Am. 153, A295 (2023)]. The effects of such within- vs. between-speaker acoustic variability on the perception of speaker identity, however, have not been explored. The present study therefore examines the perception of speaker identity in bilingual voices. The prediction is that acoustic variability will also affect speaker identity perception: voices will be discriminated best for between-speaker samples, while within-speaker variability will not affect perception of speaker to the same extent. To test this prediction, listeners participated in a voice discrimination task using bilingual voice data produced by Korean heritage speakers across different languages (Korean, English) and speech styles (read, extemporaneous). The data will be analyzed to measure the effects of speaker, language, and speech style on voice discrimination. The results will be reported in relevance to the relationship between bilingualism and speech style on voice quality and speaker identity.

4aSCb30. Tongue bracing robustness under perturbation: Comparison of L1 versus L2 speech with and without bite-block. Abiodun Ibikunle (Linguist., Univ. of BC, Vancouver, BC, Canada), Marcell Maitinsky (Linguist., Univ. of BC, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, mlmtnsky@student.ubc.ca), Annabelle Purnomo, Chenxi Xu, Yadong Liu, and Bryan Gick (Linguist., Univ. of BC, Vancouver, BC, Canada)

Lateral tongue bracing is a universal speech posture observed across many languages [Gick *et al*., 2017, JSLHR 60; Liu *et al*., 2022, JIPA, Phonetica 79]. Previous work has found that under perturbation by 10 mm bite-blocks, there is less lateral contact during a speaker's L2 speech compared to their L1 speech [Bengtson *et al*., 2023, J. Acoust. Soc. Am. 154]. However, the research did not investigate lateral contact without bite-blocks and the effect of this perturbation on posture remains unclear. The present study addresses this by comparing a speaker's L1 and L2 speech both with and without bite-block perturbation, with the hypothesis that posture during speech movement is more robustly maintained under perturbation with more language experience. Participants will read two short texts in their L1 and L2 separately, both with and without 10 mm bite-blocks, and F1/F2 of cardinal vowels will be compared between bite-block conditions for each language. L2 proficiency will be collected for each participant and results will be reported with relevance to posture and speech movement coordination for L1 versus L2, with implications discussed for proficiency. [Work supported by NSERC].

4aSCb31. Individual differences in perception of Thai and Korean stops. Melissa Baese-Berk (Linguist., Univ. of Chicago, 1115 E 58th St., Rosenwald Hall Rm. 203, Chicago, IL 60657, mmbb@uchicago.edu) and Charlie Nagle (Univ. of Texas, Austin, TX)

Differentiating between pairs of unfamiliar speech sounds is often challenging for adult listeners. However, this difficulty varies depending on multiple factors. For example, it is clear that the relationship between a listener's first language and the target language can impact the ease or difficulty of differentiating pairs of sounds. However, individuals also vary widely in their ability to differentiate these novel sounds, even if they share a language background. In this study, we compare native English listeners' ability to differentiate between initial stop consonants in Korean and in Thai. In addition to comparing individual performance across these two target languages, we also assess performance as a function of various linguistic and cognitive measures including phonological short-term memory, language learning aptitude measures, and L1 production and perception patterns.

THURSDAY MORNING, 16 MAY 2024 ROOM 213, 8:00 A.M. TO 11:55 A.M.

Session 4aSP

Signal Processing in Acoustics, Acoustical Oceanography, Physical Acoustics, Computational Acoustics, and Underwater Acoustics: Bayesian and Machine Learning in Acoustics II

Yangfan Liu, Cochair
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Chair's Introduction—8:00

Invited Papers

8:05

4aSP1. Application of machine learning on complete vehicle data for interior vibration prediction based on transfer paths in narrowband spectrum. Marcus Maeder (School of Eng. and Design, Tech. Univ. of Munich, Garching near Munich, Germany, marcus.maeder@tum.de), Martin Eser, and Steffen Marburg (School of Eng. and Design, Tech. Univ. of Munich, Garching, Germany)

Interior noise and vibration are critical quality criteria for vehicles in the automotive sector. The latter is essential in driving comfort, where many development resources are used for the design and the evaluation. As a result, testing the entire vehicle is very important to assess vibration and ride comfort. The assembly of the entire vehicle is carried out virtually using transfer functions derived either from simulations or from measurements. Since simulations of a complete vehicle are still challenging,transfer functions are usually measured with considerable effort. Therefore, limiting the necessary measurements to specific components is desirable to be efficient and to save costs. In this work, the authors use an artificial neural network to predict the interior vibration profiles of a complete vehicle based on full test drives at different speeds and road conditions. Triaxial acceleration measurements serve as the database. In addition, a criterion is proposed to select essential sensors for the learning process. The results show that if the data is handled carefully, many sensors can be discarded, and the network can predict accurate acceleration spectra for virtual sensors at various test conditions.

8:25

4aSP2. Implementation of supervised machine learning classification for detection and severity determination of electrified power train noise and vibration fault diagnosis. Joohyun Lee (Ray W. Herrick Laboratories/School of Mech. Eng., Purdue Univ., 3977 State Rd. 38 E, Apt 207, Lafayette, IN 47905, lee2243@purdue.edu), Yangfan Liu, J. S. Bolton, and Patricia Davies (Ray W. Herrick Laboratories/School of Mech. Eng., Purdue Univ., West Lafayette, IN)

Vehicles equipped with electrified powertrains produce lower sound and vibration levels compared to those equipped with internal combustion engine powertrains. This makes noise and vibration (N&V) from other non-engine components more perceptible. Gear growl is one of the newly observed N&V that brings concerns by the passengers and manufacturers. The understanding of signal characteristics and the threshold for determining whether gear growl requires attention remains limited. To address this, supervised machine learning classification is employed. Root-mean-square (RMS) and spectral entropy values are sufficient for the classification of vibration data with test accuracy of 0.983. However, the acoustic signal required more features due to background noise, making data linearly inseparable. Features that describe the characteristics of acoustic data are studied, extracted, and selected. Utilizing a support vector machine (SVM) for classification, the study achieves an average test accuracy of 0.918. Further, a multi-class classification model is implemented based on preliminary subjective listening studies, classifying different severities of gear growl. Further listening studies are suggested for improving multi-class classification performance. Methods described in this study mainly focus on the analysis of gear growl, but they can be generalized for N&V signal-based fault diagnosis applications.

Contributed Paper

8:45

4aSP3. Detecting ringed seal vocalizations in multiple environments using deep learning. Karlee Zammit (Earth and Ocean Sci., Univ. of Victoria, 504 Ridgebank Crescent, Victoria, BC V8Z4X9, Canada, karlee.zammit@gmail.com), William D. Halliday (Wildlife Conservation Society Canada, Whitehorse, YT, Canada), Fabio Frazao (Comput. Sci., Dalhousie Univ., Halifax, NS, Canada), Sebastien Fabbro (Phys., Univ. of Victoria, V8Z4X9, BC, Canada), and Stan Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada)

Deep learning methods have recently been successfully applied to create a variety of automated acoustic detectors in the field of marine bioacoustics. Automated detectors are essential for analyzing large volumes of passive acoustic monitoring (PAM) data since manual analysis is prohibitively time-consuming and costly. PAM is the primary method for obtaining data on species which are endemic to remote regions, such as the Canadian Arctic. Arctic ringed seals are listed as a Species of Special Concern in Canada due to a loss of critical habitat caused by the effects of climate change. Here, ResNet, a convolutional neural network architecture, is trained on thousands of examples of ringed seal vocalizations recorded at various locations within the Canadian Arctic to create the first practical automated ringed seal detector. The network achieves a precision of 0.89, recall of 0.80, and F1 score of 0.85 when tested on 215 five-minute recordings from sites included in the training process. To improve the generalizability of the detector for new locations, fine-tuning is performed using a small subset of annotated data from new sites. The detector will be available as an open-source tool for researchers to use as the basis for further development of new automated detectors.

Invited Paper

9:00

4aSP4. Application of deep learning methods for obstacle classification in ultrasonic surround sensing. Jona Eisele (Chair of Vibroacoustics of Vehicles and Machines, Tech. Univ. of Munich, Boltzmannstr. 15, Garching near Munich 85748, Germany, jona.eisele@tum.de), André Gerlach (Bosch Res., Robert Bosch GmbH, Renningen, Germany), Marcus Maeder (Chair of Vibroacoustics of Vehicles and Machines, Tech. Univ. of Munich, Garching near Munich, Germany), Andreas Koch (Inst. of Appl. Artificial Intelligence, Stuttgart Media Univ., Stuttgart, Germany), and Steffen Marburg (Chair of Vibroacoustics of Vehicles and Machines, Tech. Univ. of Munich, Garching, Germany)

Ultrasonic sensors are widely used in driver assist systems for close-range surround sensing. Based on the pulse-echo method, the distance to obstacles is calculated, particularly in scenarios such as parking and maneuvering. For advanced ultrasonic perception systems, however, the classification of obstacles should be considered, too. For instance, it is relevant to distinguish between traversable and non-traversable objects, as well as to detect pedestrians. Classifying obstacles by ultrasonic echoes remains an area of ongoing research, and, in contrast to ultrasonic ranging, implementation in current systems is rudimentary at best. This work examines the application of deep learning methods for ultrasonic-based object classification in driver assist systems. Features based on raw time signals, amplitude- and phase-based echo features as well as time-frequency images are compared. The preprocessing and feature extraction methods are evaluated based on experimental data captured in low and high clutter environments. Promising classification results are achieved using convolutional neural networks, demonstrating a significant improvement over traditional methods. Finally, we discuss the benefits of employing small aperture ultrasonic sensor arrays to increase classification robustness using beamforming methods.

9:20

4aSP5. Direction of arrival estimation with convolutional neural networks and multiple signal classification. Christopher J. Bell (Univ. of Rhode Island, The Fascitelli Ctr. for Adv. Eng., Kingston, RI 02881, cj.bell91@gmail.com), Kaushallya Adhikari (Univ. of Rhode Island, Kingston, RI), and Lauren Freeman (NUWC Newport, Newport, RI)

Recently, there has been a proliferation of applied machine learning (ML) research, including the use of convolutional neural networks (CNNs) for direction of arrival (DoA) estimation. With the large increase of research in this area, it is important to balance the performance and computational costs of CNNs with classical methods of DoA estimation such as Multiple Signal Classification (MUSIC). We outline the performance of both methods of DoA estimation for single source and two source cases and compare them to the Cramer-Rao lower bound (CRLB). For each source case, a CNN was trained for a perfect uniform line array (ULA), a perturbed ULA, and a ULA with missing sensors. The three cases are interesting studies as classical methods for DoA estimation such as MUSIC assume perfect array conditions, whereas the CNNs assume nothing about the structure of the data. The training data for each network was created by leveraging a signal model to create synthetic data at different SNRs. The results for each network are then compared to the results for MUSIC using the same signal case and array condition. The results indicate that for the single source case the CNN only performs significantly better than MUSIC for the perturbed array case. For the two-source case, the CNNs significantly outperform MUSIC for all ULA conditions, however, when compared to the CRLB it is shown that the CNN typically produces a biased estimate.

9:35

4aSP6. Physics informed sound field interpolation using an acoustic sensor network. Rashen Fernando (Elec. and Comput. Eng., Univ. of Illinois at Chicago, 2955 South Emerald St., Unit 1F, Chicago, IL 60616, pferna20@uic.edu), Manan Mittal, Yongjie Zhuang, Ryan M. Corey, and Andrew C. Singer (Elec. and Comput. Eng., Stony Brook Univ., Stony Brook, NY)

Sound field estimation is the process of analyzing and characterizing the distribution of sound waves in a particular physical space. The applications of sound field estimation extend to various areas, including the visualization of acoustic fields, interpolation of room impulse responses, identification of sound sources, capturing sound fields for spatial audio, and spatial active noise control, among other potential uses. In a previous study, a directionally weighted kernel has been used to estimate the sound field, where the priori information of source directions is employed to improve the estimation accuracy. In another separate study, spherical harmonics have been used to represent the sound field. However, the order of spherical harmonic coefficients was limited due to the limited number of microphones. This research introduces a novel method for sound field estimation using multiple microphones to sample a source-free volume. A physics-informed neural network is used to predict the spherical harmonics coefficients and locations of unknown sources to estimate the sound field.

9:50–10:05 Break

10:05

4aSP7. Development of the 2nd Cadenza challenge for improving music listening for people with a hearing loss. Michael A. Akeroyd (School of Medicine, Univ. of Nottingham, Nottingham, United Kingdom), Scott Bannister (Univ. of Leeds, Leeds, United Kingdom), Jon P. Barker (Univ. of Sheffield, Sheffield, United Kingdom), Trevor J. Cox, Gerardo Roa, Bruno Fazenda (Univ. of Salford, Salford, United Kingdom), Jennifer L. Firth (Univ. of Nottingham, Nottingham, United Kingdom), Simone Graetzer (Univ. of Salford, Salford, United Kingdom), Alinka Greasley (Univ. of Leeds, Leeds, United Kingdom), Rebecca Vos (Univ. of Salford, Salford, United Kingdom), and William M. Whitmer (Hearing Sci. - Scottish Section, Level 3, New Lister Bldg., Glasgow Royal Infirmary, Glasgow G31 2ER, United Kingdom, bill.whitmer@nottingham.ac.uk)

The Cadenza project is an ongoing project that aims to improve music quality for those with a hearing loss. The project is running signal-processing and machine-learning challenges to address different listening issues and scenarios. During the first round, the challenge focused on non-causal music source separation to allow remixing for those with hearing loss. This fed into an ICASSP 2024 challenge, which had crosstalk from loudspeaker reproduction included. There are three potential arms to our upcoming 2024 challenge based on reported issues from hearing-impaired music listeners: (1) low-latency causal audio source separation, (2) lyric intelligibility enhancement without loss of timbre or instrumental balance, and (3) loudness/dynamic range control. Each of these potential challenges raise questions as to the appropriate reference signals, as well as the practicalities of deriving the appropriate signals for an open-source machine-learning challenge. [Work supported by UK EPSRC Grant No. EP/W019434/1]

10:20

4aSP8. Development of AcousticsAI, a large language model chatbot trained on peer-reviewed acoustics literature. Nathan Leo (Eng. Sci. and Mech., The Penn State Univ., 212 Earth and Eng. Sci. Bldg, University Park, PA 16802, nal5367@psu.edu) and Christopher M. Kube (Eng. Sci. and Mech., The Penn State Univ., University Park, PA)

Large Language Models (LLMs) like ChatGPT are altering how we interact and extract useful knowledge. However, the ability to extract domain specific knowledge about acoustics is still lacking. This presentation highlights our efforts in developing AcousticsAI, which is a LLM chatbot trained specifically with peer-reviewed acoustics literature. AcousticsAI is being developed as an advanced tool to aid acousticians at the cutting edge of acoustics research. Being research focused, AcousticsAI must be able to interface the many different data streams extracted from peer-reviewed literature. This is accomplished through AutoGen, a multi-agent conversation system, enhances Large Language Models (LLMs) workflows by enabling conversable, customizable agents that incorporate human inputs and tools. AutoGen helps reduce the response correlation space, ensuring consistent and accurate LLM outputs with citations. By deploying specialized, context-specific models within the AutoGen framework, the study demonstrates improved precision and reliability in semantic reasoning over commonly used AI chatbots.

Invited Paper

10:35

4aSP9. Speech detection models for effective communicable disease risk assessment in air travel environments. Tenon Charly Kone (Flight Res. Lab., National Res. Council Canada, 1200 Montreal Rd., Ottawa, ON, Ottawa, ON K1A 0R6, Canada, tenoncharly.kone@nrc-cnrc.gc.ca), Sebastian Ghinet, Sayed Ahmed Dana, and Anant Grewal (Flight Res. Lab., National Res. Council Canada, Ottawa, ON, Canada)

In environments characterized by elevated noise levels, such as airports or aircraft cabins, travelers often find themselves involuntarily speaking loudly and drawing closer to one another in an effort to enhance communication and speech intelligibility. Unfortunately, this unintentional behaviour increases the risk of respiratory particles dispersion, potentially carrying infectious agents like bacteria which makes the contagion control more challenging. The accurate characterization of the risk associated to speaking, in such a challenging noise environment with multiple overlapping speech sources, is therefore of outmost importance. Among the most advanced signal processing strategies that can be used to accurately determine who spoke when and with whom and for how long but most importantly how loudly, at one location, artificial intelligence-based speaker diarization approaches were considered and adapted for this task. This article details the implementation of speaker diarization algorithms, customized to extract speaker and speech parameters discreetly. Validation and preliminary study results are also provided. The algorithms calculate speech duration and sound pressure level for each sentence and speaker, aiding in assessing viral contaminant spread. The paper focuses on applying these algorithms in noisy environments, particularly in confined spaces with multiple speakers. The findings contribute to proactive measures for containing and managing communicable diseases in air travel settings.

Contributed Papers

10:55

4aSP10. Using convolutional neural networks to select the optimal wavelet for audio compression. Shaun Pies (Phys., Univ. of New Orleans, 2000 Lakeshore Dr., New Orleans, LA 70148, spies@uno.edu), Kendal Leftwich, and Juliette W. Ioup (Phys., Univ. of New Orleans, New Orleans, LA)

The vast amount of data generated and stored across domains is overwhelming. Noteworthy examples include the Event Horizon Telescope using 3.5 petabytes for the first black hole image, LIGO amassing 4.5 petabytes, and CERN with 487 petabytes. This research addresses the challenge of data storage and transmission using machine learning and wavelet analysis. The goal is to create an efficient data compression method that significantly reduces storage needs without compromising data integrity. Wavelets, mathematical tools which break down signals into coefficients, show promise for compression. Choosing the right wavelet is crucial; our research demonstrates that machine learning can effectively solve this problem. By training a convolutional neural network on spectrograms of audio files, we identify the optimal wavelet for data compression, achieving up to 50% compression with minimal loss.

11:10

4aSP11. Testing transfer learning for 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)

One difficulty in applying deep learning techniques to ocean acoustics is the spatially and temporally varying environmental properties. Another challenge is the lack of labeled data for training large networks. The overall goal of this work is to develop deep learning approaches that can be adaptable to different conditions. For example, a trained neural network will fail to generalize when the appropriate environmental variability is not included in the training data. To improve network performance, transfer learning can modify a pre-trained network to make predictions on data that was recorded under different conditions than the original dataset. In this work, a convolutional neural network was trained on acoustic data measured in a water tank, while the water was at room temperature, to predict source-receiver range. Transfer learning was used to update the pre-trained model with a smaller set of data measured at different water temperatures. The resulting model better generalizes to measurements at different temperatures. This approach illustrates how transfer learning can be used in ocean acoustics to improve generalizability in a specific area with less labeled data and lower computational cost. [Work supported by the Office of Naval Research, Grant N00014-22-12402.]

11:25

4aSP12. Machine learning for analysis of wind farm noise. Heather L. Lai (Eng. Programs, State Univ. of NY at New Paltz, 208 Eng. Innovation Hub, SUNY New Paltz, New Paltz, NY 12561, laih@newpaltz.edu), Anne C. Balant (Commun. Disord., State Univ. of NY at New Paltz, New Paltz, NY), and Chih-Yang Tsai (School of Business, State Univ. of NY at New Paltz, New Paltz, NY)

Two challenges associated with analyzing acoustical data from wind farms are: 1) separating turbine sounds from environmental sounds and 2) classifying acoustical samples into different types of wind turbine noise based on acoustic characteristics. Machine-learning methods for classifying general environmental sounds have been developed using large human classified databases (e.g., YAMNet), but only a few studies have targeted classification of wind farm noise (WFN) specifically. Techniques for classifying wind farm noise have focused on identification of amplitude modulation (AM) using both traditional methods such as low frequency peak prominence (IOA method) and machine learning methods using both targeted AM acoustics features and more general deep acoustic features. To address these two challenges, we are developing a multi-echelon machine learning framework to identify and classify noise from wind farms using publicly available windfarm data and open-source software. The first echelon provides an automated method for identifying WFN samples that are free of environmental sounds. The second echelon uses machine learning to classify these wind farm noises according to the degree of AM, prominent tones, and other factors that might contribute to the human response and are incorporated in the metrics used or potentially used to assess compliance with regulations.

11:40

4aSP13. Using deep learning for recreating binaural audio. Will Sloan (System and Comput. Eng., Carleton Univ., 1125 Colonel By Dr., Ottawa, ON K1S 5B6, Canada, willsloan@cmail.carleton.ca) and Amir Laghai (Dept. of Electronics, Carleton Univ., Ottawa, ON, Canada)

In recent years, there has been an increase in research around generating spatialized audio using a mono audio signal. Methods like using neural networks which combine image segmentation with object location for adding back the spatial qualities are often developed. Instead, our project focuses on taking arbitrary mono input sound and an input angle, and outputting spatial stereo audio of the input sound with the directionality of the angle. This is different from current implementations as it is a simpler approach to what spatial audio generation is, and it allows for the use in the model in new areas. Using a binaural microphone and a custom-made anechoic chamber, 120 hours of labelled binaural audio was recorded for use in our model. The audio consists of frequency sweeps, pink noise, and phonetically balanced speech. Our method is to predict a complex short-time Fourier transform mask which will contain the phase and amplitude within it. The model is an autoencoder based on the U-NET model, which is applied to the mono input before being compared to the labelled data for training. With this simpler approach, we hope to make spatial audio more accessible for a variety of applications.

Session 4aUW

Underwater Acoustics and Acoustical Oceanography: Scintillation in Acoustic Propagation

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

Dajun Tang, Cochair
Applied Physics Lab, University of Washington, 1013 NE 40th Street, Seattle, WA 98105

Chair's Introduction—8:00

Invited Papers

8:05

4aUW1. On the feasibility of using short-range, high-frequency transmissions to characterize the vertical-spectra of small-scale internal waves and turbulence in a bottom boundary layer. John A. Colosi, Timothy F. Duda (Woods Hole Oceanographic Inst., Woods Hole, MA), Matthew Alford, Ying-Tsong Lin, and Gunnar Voet (Marine Physical Lab., Scripps Inst. of Oceanogr., La Jolla, CA)

Using weak fluctuation theory, we examine the detectability of small-scale internal waves and turbulence via short-range, high-frequency transmissions to a vertical array. For internal waves the theory needs modification since the ranges of interest are shorter than the internal wave correlation lengths. The turbulence spectra are characterized by thermal dissipation, χ, kinetic energy dissipation, ε, and outer scale $L_{:0}$ (largest turbulent eddies) and regions of this space can be explored with different ranges and acoustic frequencies. Very little is known of the internal-wave structure when wave breaking starts. As an example, we consider an experimental arrangement atop the Atlantis 2 seamount where there is a strong, tidally driven boundary layer approximately 100-m thick.

8:25

4aUW2. Polynomial chaos expansions for modelling the statistics of acoustic propagation in random waveguides. Kevin D. LePage (NATO STO Ctr. for Maritime Res. and Experimentation, Viale San Bartolomeo 400, La Spezia, SP 19126, Italy, kevin.lepage@cmre.nato.int)

The use of Polynomial Chaos expansions to model the transfer of the statistical properties of the sound speed in ocean waveguides to those of acoustic waves passing through them is described. A perturbational framework approximating the interaction of acoustic normal modes with fluctuations around a background sound speed is used, with the fluctuations being represented vertically by empirical orthogonal functions and horizontally by correlation functions. The Polynomial Chaos expansions are derived to second order in the uncorrelated random variable representing the sound speed fluctuations for the generalized phase of the complex modal amplitudes, and to third order for the modal amplitudes themselves. The ability of the second order polynomial chaos expansion for the generalized model phase to accurately model acoustic propagation statistics is closely coupled to the accuracy of the adiabatic approximation in light scattering regimes. The weights of the Polynomial Chaos expansions are obtained using both direct (theoretical) and indirect (least-squares fit) methods. Divergence between these two sets of weights increases with combinations of increasing strength of sound speed fluctuations, increasing frequency, or increasing range, indicating where a higher order expansion may be necessary. Modelling results for the first and second moments of the acoustic intensity and the Scintillation Index in random waveguides are presented and compared to Monte Carlo results.

8:45

4aUW3. Probabilistic acoustic predictions leveraging NESBA and ARGO measurements. Bill Stevens (Portland State Univ., San Diego, CA) and Martin Siderius (Portland State Univ., 1900 SW 4th Ave., Ste. 160, Portland, OR 97201, siderius@pdx.edu)

Ocean acoustic propagation models used for sonar performance prediction often rely on global or regional ocean modeling and data assimilation, or monthly climatologic averages, for estimating the 4D ocean temperature and salinity (T/S) field. Ocean models rely heavily on satellite-based ocean surface measurements, e.g., sea surface temperature (SST) and height (SSH), plus typically vastly less resolved in space and time vertical water column measurements derived using ship-based and/or ARGO float instruments. Systems also exist that create 3D synthetic T/S fields from satellite SST/SSH measurements, e.g., the Modular Ocean Data Assimilation System (MODAS) and the more recent Improved Synthetic Ocean Profile (ISOP) system. It is unclear, however, how well these synthetic T/S fields capture significant ocean ducting conditions. Similarly, climatologic averages combine historical vertical T/S profile data in spatial cells by month; but the averaging process tends to blur out important features. This talk will address the use of 2021 New England Shelf Break Acoustics (NESBA) experiment and ARGO vertical profile data for: (1) testing the degree to which ocean models or climatology

capture important acoustic features under a range of conditions; and (2) providing more realistic sonar performance prediction means and uncertainties. [Work supported by the Office of Naval Research].

9:05

4aUW4. Abstract withdrawn.

9:25

4aUW5. Analysis of mid-frequency sound intensity fluctuations on the Washington shelf. Dajun Tang (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, dtang@uw.edu), Brian T. Hefner, John B. Mickett, Ramsey R. Harcourt, Guangyu Xu, Eric I. Thorsos, and Kumar R. Prakash (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

A joint Oceanography/Acoustics experiment was conducted 15 July–13 August 2022 in Washington's shallow coastal shelf waters to investigate mid-frequency sound intensity fluctuations and the oceanographic mechanisms driving them. Acoustic pulses centered at 3.5 kHz and 6 kHz were transmitted up-shelf from a stationary source and recorded by receivers on two moorings at 10 km and 20 km ranges. A ship-towed profiler, the Shallow Water Integrated Mapping System (SWIMS), repeatedly sampled the ocean between the source and receiver moorings, providing sound speed measurements as a function of range, depth, and time. In addition, five oceanographic moorings distributed along the acoustic path took additional ocean data over time. A roughly 15- dB ping-to-ping TL variability and scintillation index of near unity are found at the 20 km range. The intensity data show strong correlation between pings separated by a few minutes, beyond which it is uncorrelated. After smoothing, they also show trends at roughly 12-hour intervals. Assisted by environmental data and numerical modeling, the oceanographic mechanisms for these observed fluctuations are investigated. Implications for signal processing in the presence of such intensity fluctuations are also discussed. (Work supported by Office of Naval Research.)

9:45–10:00 Break

10:00

4aUW6. Measurements of mid-frequency acoustic signal variability in the Western Barents Sea. Altan Turgut (Naval Res. Lab., Acoust. Div., Washington, DC 20375, altan.turgut@nrl.navy.mil), Jeffrey Schindall (Naval Res. Lab., Washington, DC), Ewa Jarosz, and Ewa Wijesekera (Naval Res. Lab, Stennis Space Ctr., MS)

Two Vertical Line Arrays (VLAs) and one source moorings were deployed to study along- and cross-shelf acoustic propagation in the Western Barents Sea during October 12–19, 2022. An ITC 2010 source was used to transmit Linear Frequency-Modulated (LFM) acoustic signals covering a frequency band of 0.7–4.2 kHz. Several oceanographic moorings were also deployed to measure time series of ocean currents and sound-speed profiles. Analysis of the acoustic data showed strong effects of tidal currents and surface roughness on mid-frequency acoustic propagation. Along- and cross-shelf acoustic propagation data showed semidiurnal travel-time fluctuations due to tidal currents. In addition, a 15 dB increase in Transmission Loss was observed during the storm events due to rough surface scattering. Finally, an assessment of the Polar Front and its effects on acoustic propagation (both travel-time and amplitude) are studied. Several data and simulated examples are provided to demonstrate the 3D effects of a Polar front and nonlinear internal waves on mid-frequency acoustic propagation. [Work supported by the ONR.]

Contributed Papers

10:20

4aUW7. The Acoustic Laboratory for Marine Applications (ALMA) applied to fluctuating environment analysis. Samuel Pinson (École Navale, Lanveoc, France), Victor Quilfen (Shom, Brest, France), Florent Le Courtois (DGA TN, Ave. de la Tour Royale, Toulon 83000, France, florent.lecourtois@gmail.com), Gaultier Real (CMRE, La Spezia, Italy), and Dominique Fattaccioli (DGA TN, Toulon, France)

The Acoustic Laboratory for Marine Applications (ALMA) has been used to address problems in underwater acoustics, such as sound propagation in fluctuating environments. In this work, data from the ALMA-2016 at-sea campaign are used to analyze the ocean fluctuation's influence on sound propagation in a shallow-water waveguide. The experiment took place in November 2016 on the continental shelf of the eastern coast of the island of Corsica. A source and a receiver array were 9.3 km apart in a nearly constant water depth of 100 m. A thermistor chain was moored near the source to monitor sound speed fluctuations. The source emitted a variety of signals from which the chirp (1–13 kHz) is used to extract the waveguide eigenrays. To do so, a time-domain beamforming is performed on the match-filtered received signals with an automatic detection of local maxima in the time of arrival/direction of arrival (TOA/DOA) domain. A 2 min acquisition period of more than 13 h duration shows significant fluctuations in eigenray TOAs/DOAs. Qualitative comparisons with synthetic signals obtained from simulations permit reproduction of the observed eigenray fluctuations without including range dependence of the sound-speed profile. In addition, the joint analysis of the probability density function of the normalized acoustic intensity and of the thermistor chain data highlights the time dependence of the received signal characteristics.

10:35

4aUW8. Mid-frequency convergence zone propagation. Franklin H. Akins (Scripps, UCSD, 562 Arenas St., La Jolla, CA 92037, fakins@ucsd.edu), William Hodgkiss (Scripps, UCSD, San Diego, CA), and William Kuperman (Scripps, UCSD, La Jolla, CA)

Convergence zone propagation has a cycle distance of order 50 km with most of the path spent below the thermocline. Calculations using Garrett-Munk statistics and rays computed with a smoothed background sound speed profile over this single cycle predict propagation in the partially saturated regime and an integration time of 500 seconds (time spent in a single output bin of a discrete Fourier transform). Parabolic equation simulations with synthetic fine structure and ocean dynamics validate the ray-based calculations. Data of narrowband transmissions from 1.5 to 7.5 kHz from a shallow towed source (150 m) to a shallow drifting array (150 m) depth at a range of 59 km demonstrate smoothly varying phase in a beam permitting coherent integrations of 3.5 minutes. The shorter integration time in data is consistent with non-uniform motion estimated from GPS.

10:50

4aUW9. Issues in mid-frequency sound intensity fluctuation on the Washington Shelf. Brian T. Hefner (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, hefner@apl.washington.edu), Dajun Tang, John B. Mickett, Ramsey R. Harcourt, Guangyu Xu, Eric I. Thorsos, and Kumar Prakash (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Strong mid-frequency sound intensity fluctuations were found on the Washington shelf during an experiment in the summer of 2022. The Scintillation Index (SI) of integrated sound energy on single channels at 20 km range was near unity for both 3.5 kHz and 6.0 kHz signals. Accompanying oceanographic measurements from moorings and a towed profiling system revealed sound field variability at multiple time and spatial scales caused by linear and nonlinear internal waves, internal tides, and coastal trapped waves. While the 2022 dataset provides valuable information for addressing issues on intensity fluctuation, additional experimental data are needed to understand the impacts on active and passive mid-frequency sonar systems. To that end, a second Washington Shelf experiment is being planned for the summer of 2025. This talk will discuss the planned measurements and the range of issues to be addressed. These include the frequency-dependence of SI over 1–10 kHz, broadband fading, vertical and horizontal coherence, and the impacts of variability on propagation in subsurface ducts. These measurements are important for investigating fundamental oceanographic causes of sound fluctuation as well as for the design of appropriate signal processing methods. (Work supported by the Office of Naval Research.)

11:05–11:25 Panel Discussion

Session 4pAAa

Architectural Acoustics, Noise, Computational Acoustics, Signal Processing in Acoustics, and Structural Acoustics and Vibration: Artificial Intelligence and Machine Learning for Built Environments

Semiha Yilmazer, Cochair
Interior Architecture and Environmental Design, Bilkent University, Bilkent University, Faculty of Art, Design and Architecture, Department of Interior Architecture and Environmental Design, Ankara, 06800, Turkey

Dick Botteldooren, Cochair
Information Technology, Ghent University, Technologiepark 126, Technologiepark 126, Gent, 9052, Belgium

Chair's Introduction—1:00

Invited Papers

1:05

4pAAa1. Using large language models in the analysis of urban sound environments. Dick Botteldooren (Information Technol., Ghent Univ., Technologiepark 126, Technologiepark 126, Gent 9052, Belgium, dick.botteldooren@ugent.be), Tiebe Vercoutter, and Yuanbo Hou (Information Technol., Ghent Univ., Ghent, Belgium)

In recent years, Large Language Models such as Generative Pre-trained Transformers (GPT) have revolutionized Artificial Intelligence (AI). Such models are extremely successful in imitating general intuitive knowledge. Such knowledge is applied by humans in assessing the urban sound environment in different ways. Firstly, it helps to create expectations on the sound environment based on general knowledge of the place. Secondly, it allows to assess the plausibility and consistency of the verbal description of a sound environment. Hence, we propose to combine a GPT with a sound event and scene recognition AI to (1) contrast the recognized sounds with expectations based on geographical information on traffic infrastructure and points of interest near the measurement location; (2) create verbal soundscape annotations including perceived liveliness, calmness, etc. Prompt engineering, that is pre-conditioning and asking precise questions to the GPT, requires some domain knowledge and precise definition of the objective. Results show that labeling of sound events is improved, and in particular labels that would not be used by a human can be excluded by including contextual knowledge on the location with GPT. They also show that a plausible soundscape description is obtained.

1:30

4pAAa2. A non-linear model approach for predicting soundscape perception of study areas. Semiha Yilmazer (Faculty of Art, Design and Architecture, Dept. of Interior Architecture and Environ. Design, Bilkent Univ., Ankara 06800, Turkey, semiha@bilkent.edu.tr) and Zekiye Şahin (Interior Architecture and Environ. Design, Bilkent Univ., Ankara, Turkey)

A large majority of the studies use linear regression-based models for soundscape modeling due to their easy applicability. Only a few studies have chosen non-linear structures, such as neural networks. Moreover, students' perceptions of soundscape quality in study areas have yet to be explored. We aimed to predict soundscape perception of study areas by applying neural network models. We also compared our results with models applying linear approaches. Perceptual dimensions were obtained by applying the Principal Component Analysis. In this study, we used the data from a two-phase experiment to find Turkish soundscapes' affective quality attributes. The experiment was conducted in a well-insulated, quiet laboratory room at Bilkent University, and most of the participants were university students and Bilkent University faculty members.

1:55

4pAAa3. Mining user reviews, crowdsourced noise measurements, and metadata to inform design thresholds for acoustic conditions in restaurants. Samuel H. Underwood (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, 1110 S 67th St., Omaha, NE 68182-0816, samuelunderwood@huskers.unl.edu) and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, Omaha, NE)

High levels of noise are a well-known source of occupant discomfort in restaurants. While much previous work has focused on assessing subjective perceptions of restaurant soundscapes through survey instruments, less attention has been given to mining online user reviews for keywords related to psychoacoustic predictors. In this study, a dataset of online user reviews for over 2,500 restaurants, bars, and coffee shops in New York City has been obtained and compared against crowd-sourced measurements of noise levels within the same establishments. Analyses of review scores and the occurrence rate of acoustically relevant keywords in review text have been conducted using machine learning techniques to suggest design thresholds for restaurants across different categorizations. Results from this work can be used by consultants and owners to design restaurant soundscapes which may be more favorably reviewed.

2:20

4pAAa4. Toward audio-based sensing for pedestrian detection. Yiwei Ding (Georgia Inst. of Technol., 840 McMillan St. NW, Atlanta, GA 30332, yding402@gatech.edu), Chaeyeon Han, Pavan Seshadri (Georgia Inst. of Technol., Atlanta, GA), Bon Woo Koo (Toronto Metropolitan Univ., Toronto, ON, Canada), Noah Posner, Subhro Guhathakurta (Georgia Inst. of Technol., Atlanta, GA), and Alexander Lerch (Music, Georgia Inst. of Technol., Atlanta, GA)

The detection and counting of pedestrians plays a central role for the design of smart cities. Although the use of cameras for this task has been shown to have high accuracy, they come at a high cost and are susceptible to challenges such as poor lighting, fog, and obstructed views. Our study investigates audio-based pedestrian detection, combining potentially low cost sensors with advanced machine learning based audio analysis algorithms. With an audio sensor installed along the walkway, machine learning algorithms can tell from the audio whether there is a pedestrian or not, or how far the pedestrian is from the sensor. Results show the general feasibility of audio-based pedestrian detection but fall short of reaching the accuracy levels of video-based detection.

2:45–3:00 Break

Contributed Papers

3:00

4pAAa5. Toward smart acoustic spaces: embedded machine learning for sound event detection and classification in the built environment. Tre DiPassio (Elec. and Comput. Eng., Univ. of Rochester, 402 Comput. Studies Bldg., 160 Trustee Rd., Rochester, NY 14627, tredipassio@rochester.edu), Michael C. Heilemann, Benjamin R. Thompson, Jenna Rutowski, and Mark Bocko (Elec. and Comput. Eng., Univ. of Rochester, Rochester, NY)

Edge computing integrated into the built environment will enable the development of smart acoustic spaces in which the flexural vibrations of the walls and screens that define a space may be employed to generate and monitor sound and serve as unobtrusive and convenient audio interfaces. Integration of data processing and neural decision making on compact, low-power embedded hardware-software systems will make it possible to mediate real-time, personalized acoustic interactions of a user with their environment without the need to connect to the cloud, thus providing privacy and security. In this presentation we show how embedded machine learning (ML) models combined with vibroacoustic control and monitoring of elastic panels may be employed to perform various tasks such as speech recognition, sound source localization, or the detection of acoustic signatures of specific events such as a fall or other health emergencies. The selection of vibroacoustic features, the associated signal processing requirements, and the computational resources utilized by the ML models for various acoustic tasks will be discussed. We also highlight how the distributed modal response of extended flat panels can simplify the sensing and signal processing requirements in such systems.

3:15

4pAAa6. Extracting and classifying psychoacoustic features from online restaurant reviews. Martin J. Mann (School of Sci. and Eng., Baldwin Wallace Univ., 275 Eastland Rd., Berea, OH 44017, mmann18@bw.edu), Samuel H. Underwood, and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, Omaha, NE)

Recent work has shown that poor acoustic conditions persist in many restaurants. Owners who receive negative reviews of their establishment's soundscape may struggle to interpret the subjective customer responses into actionable corrective measures. Therefore, further work is needed to taxonomize the acoustically relevant keywords and phrases that occur in user reviews. In this study, an open-source database of restaurant, bar, and coffee shop reviews from across the United States has been obtained. Sentiment analysis and keyword count are used to extract positive, negative, and neutral subjective reviews and subjective features related to acoustics. The resulting subjective features are then categorized, weighed, and linked to objective acoustic parameters using machine learning techniques. Results from this study suggest that owners and consultants may be able to utilize online customer reviews to monitor acoustical comfort and ascertain the nature of an acoustical problem, not only in restaurants, but in any business sector where customers submit online user reviews.

3:30

4pAAa7. A general overview of methods for generating room impulse responses. Mihai-Vlad Baran (Music Res., McGill Univ., 1199 Rue Bishops, 807, Montreal, QC H3G 0A7, Canada, vlad.mihai.baran@gmail.com), Richard King, and Wieslaw Woszczyk (Music Res., McGill Univ., Montreal, QC, Canada)

The utilization of room impulse responses has proven valuable for both the acoustic assessment of indoor environments and music production. Various techniques have been devised over time to capture these responses. Although algorithmic solutions have been in existence since the 1970 s for generating synthetic reverberation in real time, they continue to be computationally demanding and in general lack the accuracy in comparison to measured authentic Room Impulse Responses (RIR). In recent times, machine learning has found application in diverse fields, including acoustics, leading to the development of techniques for generating RIRs. This paper provides a general overview, of approaches and methods for generating RIRs, categorized into algorithmic and machine learning techniques, with a particular emphasis on the latter. Discussion covers the acoustical attributes of rooms relevant to perceptual testing and methodologies for comparing RIRs. An examination of disparities between captured and generated RIRs is included to better delineate the key acoustic properties characterizing a room. The paper is designed to offer a foundational literature base for those interested in RIR generation for music production purposes, with future work considerations also explored.

3:45

4pAAa8. Robotization of "*in situ*" acoustic measurements. Gabriel Leroux (MJM Acoust. Consultants, Longueuil, QC, Canada), Émile Gagnon, Lionel Birglen (Dept. of Mech. Eng., Polytechnique Montréal, Montréal, QC, Canada), and Nicolas Leveque (MJM Acoust. Consultants, 753 Ste-Hélène, Longueuil, QC J4K 3R5, Canada, nleveque@mjm.qc.ca)

Currently, "*in situ*" acoustic measurements require a high level of human involvement. Measurements are carried out by technicians or engineers, and most of the time, two people are required. MJM Acoustical Consultants Inc. has explored the possibility of using an autonomous and semi-autonomous robotic solution to assist in the measurement process. Developed in partnership with university Polytechnique Montreal, the research project consists of developing a software platform and hardware for a mobile robot capable of moving around, locating itself and performing measurements using a Type I sound level meter. The prototype generates its own local area network (LAN), making it possible to send commands through high-density partitions such as brick walls or concrete floors remotely from a laptop. Localization system such as a laser remote sensing system allows the robot to navigate around obstacles. Its low noise level when stationary makes it ideally suited to conduct common standard tests such as impact noise isolation, airborne sound attenuation between rooms and background noise level measurements. Through tests carried out on field sites, the solution developed constitutes an effective instrumentation to make acoustic measurements sessions more efficient with a reduced level of human operation.

4:00–4:30

Session 4pAAb

Architectural Acoustics and Structural Acoustics and Vibration: Building Envelope Sound Isolation II

Joseph Keefe, Cochair
Ostergaard Acoustical Associates, 1460 US Highway 9 North, Ste. 209, Woodbridge, NJ 07095

Lucky S. Tsaih, Cochair
Dept. of Architecture, National Taiwan Univ. of Sci. and Tech., 43 Keelung Rd, Sec. 4, Taipei, 10607, Taiwan

Invited Papers

3:25

4pAAb1. Acoustically rated fenestration products. Casey Mahon (390 Industrial Blvd., Sauk Rapids, MN 56379, cmahon@stcloudwindow.com)

If noise is the enemy, then windows, doors, and other fenestration systems form a front-line of defense in preserving the peace. Starting with basic design elements and the material components of products manufactured to achieve an elevated level of attenuation, we will explore what it takes to reach even higher, especially as acoustic performance moves beyond simple punched openings to storefronts, curtain walls, and other specialty installations. We will see that certified test reports are something every bona fide product manufacturer should have for documentation. In summary, this presentation will explore a check list that acoustic professionals should consider for any project where noise attenuation is a primary performance objective.

3:45

4pAAb2. The effect of angle of incidence on exterior façade isolation. Wayland Dong (Paul S. Veneklasen Res. Foundation, 1711 Sixteenth St., Santa Monica, California, CA 90404, wdong@veneklasen.com) and John Lo Verde (Paul S. Veneklasen Res. Foundation, Santa Monica, CA)

A common acoustical task is the design and specification of the acoustical performance of the exterior façade system to protect building occupants from road and railroad noise. This involves calculating the outdoor-to-indoor noise reduction of the façade. The incident sound wave from a roadway will often approximate a plane wave from a specific angle of incidence, but the only data usually available is the diffuse field transmission loss of the assemblies as measured in the laboratory. While it is understood that there can be significant differences between the transmission loss at a given angle of incidence compared to the diffuse field value, a detailed and quantitative understanding of the effect is lacking. Measuring the transmission loss from plane waves at varying angles of incidence is difficult, and a combination of theoretical analysis and field measurements may allow a better understanding of the relationship and hence more accurate calculations of the interior noise level from transportation sources. The literature is reviewed, and data from field measurements is analyzed to begin to evaluate this effect.

4:05

4pAAb3. Case study of noise intrusion into glass facade residential building. Dana S. Hougland (Shen Milsom & Wilke, LLC, 1801 Wewatta, Fl. 11, Denver, CO 80202, dhougland@smwllc.com)

A combined residential and hotel building with a glass facade adjacent to a major transportation corridor provided challanges for the owner and designers in selecting an appropriate quantitative metric for evaluating the subjectively acceptable level of noise intrusion. Surveys at a series of buildings with similar proximities were evaluated to compare subjective impressions to quantitative projections.

4:25

4pAAb4. An overview of green roof acoustic performance. Shiang-I Juan (Taiwan Bldg. Technol. Ctr., National Taiwan Univ. of Sci. and Technol., 43 Keelung Rd. Sec. 4, TBTC, Taipei 106, Taiwan, S.Juan@mail.ntust.edu.tw) and Lucky S. Tsaih (Dept. of Architecture, National Taiwan Univ. of Sci. and Tech., Taipei, Taiwan)

This comprehensive review provides an in-depth examination of the acoustic performance of green roofs within urban and built environments over the past 15 years. Green roofs, characterized by their vegetative cover and substrate layers, offer a multifaceted solution to address challenges associated with urban noise pollution. The objective of this review is to synthesize existing literature, emphasizing key studies and methodologies employed to assess the noise reduction and sound isolation capabilities of green roofs. Factors influencing green roof acoustic performance, such as plant species selection in the plant layer, soil distribution, moisture content, and compaction level in the vegetation layer, as well as the overall roof design in terms of shapes and building configuration, are discussed in detail. Additionally, the impact of overall roof design, including shapes and building configuration, is explored. The relevant sound absorption coefficient and insertion loss values, drawn from the existing literature, will be reported. By systematically analyzing empirical findings, this overview offers valuable insights into the potential of green roofs as sustainable components for noise mitigation in building construction, contributing to the development of more resilient and harmonious urban landscapes.

Contributed Papers

4:45

4pAAb5. Developing and validating analytical tools to predict the aero-acoustic performance of bespoke noise mitigation solutions. Viken Koukounian (Parklane Mech. Acoust., 3-1050 Pachino Court, Burlington, ON L7L 6B9, Canada, viken@parklanemechanical.com)

The characterization of empirical knowledge from observations is, generally, referred to as the Scientific Method. Most engineering best practices rely on 'observations' (e.g., laboratory testing) to make informed decisions for future projects and/or applications. While this approach is still considered to be the 'state of the art' in most fields of acoustics, it is especially relevant in the studies of architectural acoustics and environmental noise. Since it is not possible to test every permutation of product (and/or its configurations) and environmental conditions, the results of only common geometries and parameters are reported in databases for use by professionals. However, the described framework is rigid and prohibits true optimization of a solution which should consider the solution's performance requirements, as well as the various constraints (site, project, multi-physics). A more sophisticated approach is possible, where analytical tools are developed to predict the multi-physics performance of products (such as silencers, plenum-silencers, louvers and enclosures). Numerical methods are relied on to validate these new tools against empirical data. The presented work demonstrates the optimization of a product's aero-acoustic performance according to multi-variable constraints.

5:00

4pAAb6. Residential outdoor-to-indoor acoustic conditions: Method and pilot data. Iara B. Cunha (National Res. Council Canada, 1200 Montreal Rd., M-27, Ottawa, ON K1A 0R6, Canada, iara.batistadacunha@nrc.ca), Jennifer A. Veitch, Ashley Nixon, Markus Mueller-Trapet, Sabrina Skoda, and Jeffrey Mahn (National Res. Council Canada, Ottawa, ON, Canada)

Health and well-being of people is directly influenced by the indoor environmental quality of the buildings they occupy. Noise exposure, for example, is not only a cause of annoyance but it is established as a risk factor for the development of cardiovascular diseases and it is a cause of sleep disturbance. As part of a broader research project focused on establishing guidance for suitable interior conditions for adults as they age, the National Research Council of Canada is developing a field study to measure, among other environmental conditions, outdoor-to-indoor noise conditions at homes. Objective parameters such as sound pressure levels will be monitored for a period of time and compared to residents' health and well-being assessments through sensors and surveys. The methodology and instrumentation used for the acoustic measurements will be presented as well as the early results from pilot tests. The future outcomes of this study will contribute to guidance for both new-build and retrofit scenarios of dwellings for aging in place.

5:15

4pAAb7. Effects of time varying aircraft noise on indoor speech interference. Adam Collins (400 – 2100 Derry Rd. West, Mississauga, ON L5N0B3, Canada, Adam.Collins@Stantec.com)

Aircraft noise can create noticeable speech interference inside buildings adjacent to airports. Outdoor aircraft take off noise measurements were completed and used to study time varying effects on indoor environments. Preliminary results suggest changing levels of speech interference over the duration of aircraft take off events.

Session 4pBAa

Biomedical Acoustics and Physical Acoustics: Droplets Strike Back II

Virginie Papadopoulou, Cochair
Biomedical Engineering, The University of North Carolina at Chapel Hill, 116 Manning Drive, 9004 Mary Ellen Jones Building, CB 7575, Chapel Hill, NC 27599-7575

Mario L. Fabiilli, Cochair
University of Michigan, 1301 Catherine Street, 3226A Med Sci I, Ann Arbor, MI 48109

Kevin J. Haworth, Cochair
Department of Internal Medicine, University of Cincinnati, 231 Albert Sabin Way, CVC 3939, Cincinnati, OH 45267

Invited Papers

1:00

4pBAa1. Enhancing cancer treatment through solid tumor fractionation with nanodroplet-mediated histotripsy and immunotherapy. Bar Glickstein (Bio Medical Eng., Tel Aviv Univ., Tel Aviv, Israel) and Tali Ilovitsh (Bio Medical Eng., Tel Aviv Univ., Ramat Aviv, Tel Aviv, Israel, Ilovitsh@tauex.tau.ac.il)

Checkpoint inhibition holds great promise in enhancing the body's immune response against cancer for durable and widespread anti-tumor effects. However, its efficacy in solid tumor therapy faces challenges such as limited immune cell infiltration and an immunosuppressive tumor microenvironment. This study introduces a cancer therapeutic platform, combining nanodroplet (ND)-mediated low-frequency histotripsy with anti-PD1 (aPD1) checkpoint inhibition. NDs, with their ability to penetrate capillaries and extravasate into tumor tissues, present a promising feature for noninvasive tumor treatments. The proposed method involves a two-step approach for low-energy ND-mediated histotripsy. First, NDs are activated volumetrically using a rotating imaging probe into cavitating gas bubbles. Subsequent low-frequency ultrasound implodes vaporized NDs, inducing tumor fractionation, tissue necrosis, and enhanced immune-cell infiltration, creating a T-cell–inflamed tumor. The goal is to amplify immunotherapy, advancing cancer treatment by targeting both the tumor and its microenvironment. Demonstrated in a breast cancer mouse model, the two-step approach showed significant lesions and tumor debulking. Combining aPD1 checkpoint inhibition further enhanced immune-cell infiltration and reduced tumor growth. Our findings demonstrate the potential of NDs with low-frequency ultrasound and immunotherapy for efficient noninvasive tumor treatment.

1:25

4pBAa2. Direct emulsification of liquid perfluorobutane: an improved method to formulate superheated perfluorocarbon nanodroplets. Adam Woodward, Robert Mattrey (Radiology, UTSouthwestern Medical Ctr., Dallas, TX), and Caroline de Gracia Lux (Radiology, UTSouthwestern Medical Ctr., 5323 Harry Hines, Dallas, TX 75390-8514, Caroline.Lux@UTSouthwestern.edu)

This talk will describe a simple and robust method to produce perfluorobutane (PFB) nanodroplets by direct sonication, and will compare the properties of the resultant emulsions with those obtained with the current advocated microbubble condensation method. We found that nearly 100% of particles produced by direct emulsification of PFB liquid at low temperature are liquid PFB-filled, whereas the emulsion produced by microbubble condensation if the microbubbles are not washed, contains 2000 times more non-PFB filled than PFB-filled particles. Consequently, when such suspensions are used to target receptors, the abundance of non-PFB particles will act as competitive inhibitors, and when used to extravasate to reach extravascular targets, the lower count of PFB-filled particles will require larger doses decreasing efficacy and increasing side effects. Adopting this direct formulation could be a game changer for all applications when experimental outcome is dependent on nanodroplet concentration, stability, purity and size. As a result, this method should accelerate the translation of these ultrasound activatable nanodroplets to both image and potentially treat diseased tissues.

1:50

4pBAa3. Ultrasound responsive multi-layered emulsions for drug delivery. Aaqib H. Khan (Chemical Eng., Indian Inst. of Technol. Gandhinagar, IIT Gandhinagar, Palaj Campus, Simkheda, Gandhinagar, Gujarat 382355, India, aaqib.khan@iitgn.ac.in), Sapna Bisht, Nishita Mistry, Karla Patricia Mercado-Shekhar (Biological Eng., Indian Inst. of Technol. Gandhinagar, Gandhinagar, Gujarat, India), and Sameer V. Dalvi (Chemical Eng., Indian Inst. of Technol. Gandhinagar, Gnadhinagar, Gujarat, India)

Vaporizable double emulsions, characterized by a central aqueous core, have demonstrated effectiveness in encapsulating hydrophilic drugs. This study aims to investigate the potential of incorporating an additional oil-layer in the double emulsions to encapsulate hydrophobic drugs. Vaporizable multi-layered emulsions were produced in three steps using perfluoropentane (PFP), phosphate-buffered saline (PBS), and sunflower oil. Curcumin, a natural anti-inflammatory drug, was dispersed in the oil phase. Krytox, polyglycerol polyricinoleate, and bovine serum albumin (BSA) were used as surfactants. PFP was sonicated in PBS (1:6) for 1 minute to create emulsion-1. Subsequently, emulsion-1 (1:4) was homogenized in oil to make emulsion-2. Emulsion-2 was homogenized in BSA (1:4) to yield emulsion-3 at 8000 rpm for 30 seconds. The vaporization pressure threshold was determined using 2 MHz focused ultrasound with a single-element transducer (f/# of 1.27, 0.5% duty cycle). B-mode imaging was conducted using a Verasonics Vantage 128 system with an L11-5v array to determine the droplet vaporization threshold, which was found to be 6.7 MPa. Curcumin-loading (0.87 ± 0.1 mg) was significantly higher in the multi-layered emulsions than in single-layered BSA-shelled microbubbles (0.019 ± 0.004 mg) ($p < 0.00001$), indicating that multi-layered emulsions exhibit higher drug loading capacity.

2:05

4pBAa4. The effect of decafluorobutane lipid nanoemulsion concentration on oxygen scavenging via acoustic droplet vaporization. Nour Al Rifai (Internal Medicine, Univ. of Cincinnati, 231 Albert Sabin way, Cardiovascular Ctr. 3950, Cincinnati, OH 45267-0586, alrifang@ucmail.uc.edu), Kateryna Stone (Internal Medicine, Univ. of Cincinnati, Cincinnati, OH), Diviyashree Kasiviswanathan (Medical Sci. Baccalaureate Program, Univ. of Cincinnati, Cincinnati, OH), and Kevin J. Haworth (Internal Medicine, Univ. of Cincinnati, Cincinnati, OH)

Acoustic droplet vaporization (ADV) is the phase transition of liquid droplets into gas microbubbles via ultrasound. Oxygen diffuses from the surrounding fluid into the microbubbles. This study determined the efficiency of oxygen scavenging using a lipid decafluorobutane (DFB) nanoemulsion. DFB nanoemulsions were prepared using high-shear pressure homogenization. Nanoemulsion size distributions and concentrations were measured ($n = 10$) using a Beckman Coulter Multisizer 4. Oxygen scavenging in 95% oxygenated water was measured in a flow phantom prior to introducing DFB nanoemulsion at concentrations of 0.05×10^{-4}, 0.25×10^{-4}, 0.5×10^{-4}, 2.5×10^{-4}, and 5×10^{-4} mL/mL and with ADV nucleated by an EkoSonic catheter driven with 47 W electrical power. The DFB nanoemulsion had a modal diameter of 920 ± 60 nm, polydispersity index (PDI) of 0.11 ± 0.03, and concentration of $3.11 \times 10–2 \pm 0.01 \times 10–2$ mL/mL. No statistical significant differences was detected in droplet size distribution metrics for at least 5 h at room temperature and 23 days at 4C. The baseline oxygen partial pressure (pO2) of the water was 553 ± 8 mmHg for all experiments. Peri-ADV pO2 dropped to 505 ± 16, 398 ± 44, 348 ± 11, 231 ± 27, 241 ± 7 mmHg for increasing concentration. A significant difference in pO2 was seen between thelowest and highest concentration ($p = 0.0097$). The computed ADV transition efficiency was highest at 0.25×10^{-4} mL/mL.

2:20

4pBAa5. Characterization of acoustic droplet vaporization in tissue-mimicking, fibrin-based hydrogels for microrheology applications. Anuj Kaushik (Radiology, Univ. of Michigan, Basic Radiological Sci., Medical Sci. I, A-wing, 6446B Ste. 1301 Catherine St., Ann Arbor, MI 48109-2026, ankaushi@umich.edu), Bachir A. Abeid, Jon B. Estrada (Dept. of Mech. Eng., Univ. of Michigan, Ann Arbor, MI), Mario L. Fabiilli, and Mitra Aliabouzar (Radiology, Univ. of Michigan, Ann Arbor, MI)

Ultrasound-induced vaporization of phase-shift droplets (PSDs) into microbubbles, termed acoustic droplet vaporization (ADV), has expanded biomedical applications. Here, we explored the potential of ADV as a high-resolution, on-demand microrheometer using theoretical and experimental methodologies. This approach could be used to characterize tissue elasticity at spatial resolutions unrealizable with conventional *in situ* techniques, thus assisting with identification of underlying pathologies. For theoretical studies, bubble dynamics were combined with appropriate material constitutive models accounting for viscoelasticity of the surrounding tissue. Experiments were conducted on submicron- and micron-sized, perfluoropentane PSDs embedded in fibrin-based hydrogels with varying shear elastic moduli (0.2 kPa–11 kPa) to mimic soft tissues. Experimental studies included optical high-speed imaging and acoustic interrogations to record radial dynamics and acoustic emissions of the ADV-bubbles, respectively. Radial dynamics and the acoustic emissions of microbubbles produced by ADV depended significantly on fibrin elasticity. For example, an increase in fibrin elastic modulus from ~0.8 kPa to ~6 kPa reduced the maximum expansion radius of the generated ADV-bubbles by 50%. This increase in elasticity significantly impacted both linear (e.g., fundamental) and nonlinear (e.g., subharmonic) acoustic responses of the ADV-bubbles, by up to 15 dB. These findings open doors to a novel ADV-assisted tissue characterization technique.

2:35–3:00 Break

Invited Papers

3:00

4pBAa6. Perfluorocarbon nanodroplets for ultrasound imaging and drug delivery. Geoffrey P. Luke (Thayer School of Eng., Dartmouth College, 15 Thayer Dr., Hanover, NH 03755, geoffrey.p.luke@dartmouth.edu)

Perfluorocarbon nanodroplets can be triggered to undergo a liquid-to-gas phase transition with an externally applied acoustic pulse. The resulting gaseous bubbles provide strong ultrasound contrast, enabling visualization of their distribution. This talk will focus on recent advances in imaging methods that enable multiplex and molecular imaging using the nanodroplets. In addition, a double-emulsion nanodroplet will be presented that simultaneously encapsulates hydrophobic and hydrophilic drugs. These dual-drug loaded nanodroplets can be used for on-demand delivery of combination chemotherapy and/or immunotherapy to a site of interest.

3:25

4pBAa7. Acoustic cluster therapy: An overview of research in cancer treatment. Jeffrey C. Bamber (Joint Dept. of Phys., Inst. of Cancer Res. and Royal Marsden NHS Foundation Trust, 15 Cotswold Rd., Sutton, London SM2 5NG, United Kingdom, jeff.bamber@icr.ac.uk), Nigel Bush (Joint Dept. of Phys., Inst. of Cancer Res. and Royal Marsden NHS Foundation Trust, London, United Kingdom), Udai Banerji (Drug Development Unit, Inst. of Cancer Res. and Royal Marsden NHS Foundation Trust, London, United Kingdom), Nina Tunariu (Radiology, Royal Marsden NHS Foundation Trust, London, United Kingdom), Mark O'Leary (Joint Dept. of Phys., Inst. of Cancer Res. and Royal Marsden NHS Foundation Trust, London, United Kingdom), Chanthirika Ragulan (Div. of Molecular Pathol., Inst. of Cancer Res., London, United Kingdom), Per Sontum (Oslo, Norway), Catharina de Lange Davies (NTNU, Trondheim, Norway), Melina Mühlenpfordt (Exact Therapeutics, Oslo, Norway), Erik Wennerberg (Div. of Radiotherapy and Imaging, Inst. of Cancer Res., London, United Kingdom), Anguraj Sadanandam (Div. of Molecular Pathol., Inst. of Cancer Res., London, United Kingdom), Alan Melcher (Div. of Radiotherapy and Imaging, Inst. of Cancer Res., London, United Kingdom), Amir Snapir, Andrew Healey, and Svein Kvåle (Exact Therapeutics, Oslo, Norway)

This presentation provides an overview of research on acoustic cluster therapy (ACT) in cancer treatment. ACT is a promising approach to enhancing the delivery of drugs to cancer cells that involves the use of intravenously injected clusters of microbubbles and microdroplets, which are activated by diagnostic ultrasound at the tumor site. Upon activation, the clusters expand to form bubbles of 20–40 μm diameter. When modulated by low-pressure (MI $\sim$ 0.2) low-frequency ($\sim$500 kHz) ultrasound while they sit in the tumor capillaries, these bubbles generate biomechanical effects that improve drug extravasation and interstitial permeation. They dissolve after 5–10 min. The technique has been used to temporarily open the blood brain barrier, improve the effectiveness of various chemotherapeutics in a range of tumor models, and provide ACT-imaging biomarkers that predict therapeutic outcome. It has exhibited no adverse effects to-date in a phase I/II clinical trial and improved clinical response of colorectal cancer liver metastases to chemotherapy. Ongoing research is investigating the potential synergistic effects of ACT with immunotherapy.

Contributed Papers

3:50

4pBAa8. Optimization of ultrasound contrast agent and treatment duration for drug delivery to methicillin-resistant Staphylococcus aureus diabetic wound biofilms in mice. Kelly VanTreeck (Biomedical Eng., UNC Chapel Hill, 116 Manning Dr., Chapel Hill, NC 27514, kevantre@unc.edu), Jamie Liu, Ashelyn Sidders, Kuan-Yi Lu, Amanda Velez, Phillip Durham, Duyen Bui, Michelle Angeles-Solano (Univ. of Chapel Hill, Chapel Hill, NC), Paul A. Dayton (Biomedical Eng., UNC Chapel Hill, Chapel Hill, NC), Sarah Rowe (Univ. of Chapel Hill, Chapel Hill, NC), and Virginie Papadopoulou (Biomedical Eng., UNC Chapel Hill, Chapel Hill, NC)

Chronic wounds are frequently infected with bacterial biofilms which pose a barrier to drug diffusion, drug uptake, and facilitate drug tolerance, thereby prolonging the healing process in chronic wounds and increasing the likelihood of relapse. A major contributor to relapse is persister cells, a subset resilient to high antibiotic doses. Previous work by our group showcased significantly improved gentamicin efficacy in a diabetic murine wound model using ultrasound, nanodroplets, and an anti-persister drug. Here, we aim to optimize nanodroplet formulation for persister cell targeting and treatment duration for clinical translation. In a methicillin-resistant Staphylococcus aureus (MRSA) diabetic wound model in SKH-1 hairless mice ($n = 5$), wounds were treated twice daily with gentamicin, ultrasound, and nanodroplets containing oxygen or palmitoleic acid (PA). These components aim to increase oxygenation and enhance persister uptake, respectively. Additionally, treatments with gentamicin, PA nanodroplets, and ultrasound for 2, 5, and 10 minutes were investigated ($n = 3$). Preliminary findings indicate PA most effectively potentiates antibiotic activity, 10-minute treatments improve efficacy by 1-log compared to 5-minutes, and 2-minute treatments achieve comparable efficacy to 5-minutes. These initial findings suggest a promising strategy for enhancing antibiotic efficacy and targeting persister cells in biofilms, with consideration for resource-efficient treatment times for clinical application.

4:05

4pBAa9. Localized compaction of collagen hydrogels using acoustic droplet vaporization promotes osteogenic differentiation of mesenchymal stem cells. Somnath Maji (Radiology, Univ. of Michigan, 1301 Catherine St., Med Sci. I, Rm. 6428, Ann Arbor, MI 48109, somnath.2812@gmail.com), Mitra Aliabouzar, Carole Quesada, Aidan Macpherson, Man Zhang (Radiology, Univ. of Michigan, Ann Arbor, MI), Brendon M. Baker, Renny Franceschi (Biological Chemistry, Univ. of Michigan, Ann Arbor, MI), and Mario L. Fabiilli (Univ. of Michigan, Ann Arbor, MI)

The lineage specification of mesenchymal stem cells (MSCs) is dependent on matrix stiffness, with osteogenesis promoted on or within stiff microenvironments. Unlike native extracellular matrix, conventional hydrogels have relatively static and uniform mechanical properties, thereby limiting the study of how spatiotemporally controlled mechanical cues impact MSC behavior. Here, we spatiotemporally control MSC differentiation in 3D hydrogels by actively modulating matrix stiffness using acoustic droplet vaporization (ADV). An acoustically responsive scaffold (ARS) was generated by co-encapsulating MSCs and perfluorohexane-based emulsion (13 μm) within type I collagen. ADV was used to generate bubbles within the ARS. Bubble diameter and the width of the compacted matrix region increased over time following ADV. Atomic force microscopy measurements demonstrated that the hydrogel region proximal to the bubble was significantly stiffer than the distal regions. Significantly greater levels of Runx2 and osteocalcin expression were observed in MSCs proximal to bubbles compared to distal on days 7 and 14. Higher alkaline phosphatase activity also validated the above findings. The significant upregulation of these

osteogenic biomarkers suggests ADV-induced, mechanical changes in ARSs were sufficient to enhance osteogenic differentiation of MSCs. This approach is a significant step towards controlling the 3D differentiation of MSCs in a spatially localized, non-invasive, and on-demand manner.

4:20

4pBAa10. Simultaneous plane-wave and ultra high-speed imaging of acoustic droplet vaporization. Laura Taylor (Imperial College London, London, United Kingdom), Qiang Wu (Univ. of Oxford, Oxford, United Kingdom), Kai Riemer (Imperial College London, London, United Kingdom), Luca Bau (Univ. of Oxford, Oxford, United Kingdom), Christopher Dunsby, Meng-Xing Tang (Imperial College London, London, United Kingdom), and Eleanor P. Stride (Univ. of Oxford, Inst. of Biomedical Eng., Oxford OX3 7DQ, United Kingdom, eleanor.stride@eng.ox.ac.uk)

Super-heated sub-micrometre perfluorocarbon droplets have been shown to offer advantages over microbubbles for ultrasound localization microscopy (ULM). These include better penetration of the microcirculation and enabling higher frame rates using methods such as acoustic wave sparsely activated localization microscopy (AWSALM). The variability of droplet behaviour has however been found to be substantially higher than that of microbubbles and this poses challenges for ULM image construction. The aim of this study was to use simultaneous high speed optical and acoustic imaging to capture the droplet vaporisation process and subsequent bubble dynamics. Droplets composed of octofluoropropane and decafluorobutane were injected into a polyethylene tube submerged in a 37 °C water bath and located at the focus of an ultrasound probe and microscope objective. Optical images were captured at 10 Mfps of the droplet response to ultrasound pulses of varying lengths and amplitude. The results indicate that the number of activated droplets is impacted by increasing either the ultrasound pressure or number of half-cycles as well as by the preceding activation ultrasound pulses; and that different droplet subpopulations can be activated by varying pulse parameters. These data indicate the potential for optimising droplet activation of ULM applications.

THURSDAY AFTERNOON, 16 MAY 2024 ROOM 212, 1:00 P.M. TO 5:05 P.M.

Session 4pBAb

Biomedical Acoustics, Structural Acoustics and Vibration, Computational Acoustics, Acoustical Oceanography, and Physical Acoustics: Wave Propagation in Complex Media: From Theory to Applications II

Pierre Bélanger, Cochair
1100 Rue Notre Dame O, Montréal, H3C1K3, Canada

Guillaume Haiat, Cochair
Multiscale modeling and simulation laboratory, CNRS, Laboratoire MSMS, Faculté des Sciences, UPEC, 61 avenue du gal de Gaulle, Creteil 94010, France

Contributed Papers

1:00

4pBAb1. Heterogeneous ultrasonic wave properties in leg cortical bones of Thoroughbreds. Shuta Kodama (Lab. of Ultrasonic Electronics, Faculty of Sci. and Eng., Doshisha Univ., 1-3 Miyakodani, Tatara, Kyotanabe, Kyoto 610-0321, Japan, ctwj0327@mail4.doshisha.ac.jp), Taisei Tsubata (Doshisha Univ., Kyotanabe City, Kyoto Prefecture, Japan), Norihisa Tamura, Hiroshi Mita (JRA Equine Res. Inst., Shimotsuke, Tochigi, Japan), Ko Chiba (Dept. of Orthopedic Surgery, Graduate School of Biomedical Sci., Nagasaki Univ., Sakamoto, Nagasaki, Japan), and Mami Matsukawa (Doshisha Univ., Kyotanabe, Kyoto, Japan)

Safe and inexpensive diagnosis of leg bone disease in the field is very important for Thoroughbreds and large animals, because large animals are difficult to transport, and prompt diagnosis is required. The quantitative ultrasound (QUS) is attracting attention for the evaluation of leg bones because it can give us information of elasticity, which is related to the bone quality. For the development of a QUS system of animal leg bones, understanding of characteristic ultrasonic wave properties (velocity and attenuation) is a key issue. Therefore, wave properties in Thoroughbreds' leg bones were obtained experimentally using an ultrasonic pulse technique. Precise measurements showed clear heterogeneity of ultrasonic wave properties due to the location. Opposed to the other large animals like bovines, high wave velocities were found in the medial and lateral parts, which possibly tells the effects of hard training in the fields. From a clinical perspective, the medial and lateral parts are less prone to diseases such as periostitis. The attenuation was greater in the posterior part, especially in the porous area near the trabecular bone. We also found that attenuation depended on bone matrix. Further investigation is necessary to discuss on the heterogeneity of wave properties in detail.

1:15

4pBAb2. Assessment of cortical bone phantom properties using ultrasonic guided waves transduced with a multi-element transducer. Aubin A. Chaboty (PULÉTS, École de Technologie Supérieure, 1100 Rue Notre Dame O, Montréal, QC H3C1K3, Canada, aubin.chaboty.1@ens.etsmtl.ca), Vu-Hieu Nguyen (Laboratoire Modélisation et Simulation Multiechelle, Univeristé Paris-Est Créteil, Créteil, France), Guillaume Haiat (Laboratoire Modélisation et Simulation Multiechelle, Ctr. National de la Recherche Scientifique, Créteil, France), and Pierre Bélanger (PULÉTS, École de Technologie Supérieure, Montréal, QC, Canada)

The past decade has seen extensive exploration of alternative methods for the early diagnosis of osteoporosis through the assessment of the the bone quality. Previous research on axial transmission of ultrasonic guided waves demonstrated their sensitivity to the intrinsic properties of elongated cortical bones. This study highlights the capacity of low-frequency guided waves to ascertain bone properties through the inversion of dispersion curves. The proposed inversion scheme relies on dispersion curves simulated using the semi-analytical iso-geometric analysis (SAIGA) method. The model incorporates the excitability of ultrasonic guided wave modes to ensure that the inversion uses both the dispersive trajectories and amplitudes of the modes. Two models were examined: (1) a cortical bone phantom plate covered with a soft tissue mimicking material and (2) a quasi-cylinder cortical bone phantom surrounded by a soft tissue mimicking material. A proprietary axial transmission multielement ultrasonic transducer, designed for exciting guided waves under 500 kHz, was employed to capture experimental dispersion curves on phantoms through the 2D-FFT. The mechanical properties of the bone phantoms were inferred by minimizing disparities between experimental and simulated dispersion curves. Inverse properties exhibited an error of less than 4% compared to reference values. The axial transmission probe demonstrated its proficiency in accurately measuring modes propagating inside the cortical layer.

1:30

4pBAb3. Using an instrumented hammer to detect the rupture of the pterygoid bone plate: an animal study. Manon Bas dit Nugues (MSME, CNRS UMR 8208, CNRS, 61 Ave. du Général de Gaulle, Créteil 94000, France, manon.bas-dit-nugues@u-pec.fr), Giuseppe Rosi (Université Paris-Est Créteil, CNRS, MSME ; Université Gustave Eiffel, MSME, Creteil, France), Charles-Henri Flouzat-Lachaniette (INSERM U955, IMRB Université Paris-Est ; APHP, Hôpital Henri-Mondor, Service de Chirurgie Orthopédique et Traumatologique, Créteil, France), Roman H. Khonsari (APHP, Hôpital Necker-Enfants Malades, Service de Chirurgie maxillo-faciale et chirurgie plastique, Laboratoire 'Forme et Croissance du Crâne', Paris, France), and Guillaume Haïat (MSME, CNRS UMR 8208, CNRS, Créteil, France)

Craniofacial osteotomies involving pterygomaxillary disjunction are common procedures in maxillofacial surgery to detach the pterygoid plates from the palatal bones. Surgeons still rely on their proprioception to determine when to stop impacting the osteotome, which is important to avoid complications such as dental damage and bleeding. Our group has developed a technique consisting in using an instrumented hammer that can provide information on the mechanical properties of the tissue located around the osteotome tip. The aim of this study is to determine whether a mallet instrumented with a force sensor can be used to predict the crossing of the osteotome through the pterygomaxillary plate. 31 osteotomies were carried out in 16 lamb skulls. For each impact, the force signal obtained was analyzed using a dedicated signal processing technique. A prediction algorithm based on an SVM classifier was applied to the database. We showed that the device can detect the crossing of the osteotome, sometimes before its occurrence. The prediction accuracy of the device was 94.7%. The method seemed to be sensitive to the thickness of the plate and to crack apparition and propagation. These results pave the way for the development of a peroperative decision support system in maxillofacial surgery.

Invited Paper

1:45

4pBAb4. Abstract withdrawn.

Contributed Papers

2:05

4pBAb5. k-Wave II: Introduction, progress, and road map. Bradley Treeby (Dept. of Medical Phys. and Biomedical Eng., University College London, London WC1E 6BT, United Kingdom, b.treeby@ucl.ac.uk), Antonio Stanziola, David Stansby, Devaraj Gopinathan (Medical Phys. and Biomedical Eng., Univ. College London, London, United Kingdom), Jiri Jaros (Brno Univ. of Technol., Brno, Czechia), and Ben Cox (Medical Phys. and Biomedical Eng., Univ. College London, London, United Kingdom)

k-Wave is an open-source toolbox for simulating the propagation of acoustic and ultrasound waves in complex and tissue-realistic media that is widely used across academia and industry. In this talk, we present an introduction to k-Wave II. k-Wave-II is a major re-write of the original k-Wave toolbox developed with the following aims: (1) re-engineering the code base to leverage object orientated programming, making it simpler to use for model inversions and coupled physics problems, (2) extending the algorithms to facilitate general boundary conditions on arbitrary surfaces, and to increase performance for narrow-band simulations, (3) improving the development and release process to incorporate good practice and advance long-term sustainability, and (4) improving training, user engagement, and support. We will introduce the new toolbox, discuss current progress, and a provide a road map for future development.

2:20

4pBAb6. Wavefield boundary interactions in a multi-domain environment—Experimental results. Michelle E. Swearingen (Construction Eng. Res. Lab., US Army ERDC, PO Box 9005, Champaign, IL 61826, michelle.e.swearingen@usace.army.mil), Oliver-Denzil Taylor (ECS Southeast, LLC, Baton Rouge, LA), Alanna Lester (Cold Regions Res. and Eng. Lab., US Army ERDC, Lyme, NH), Abigail Stehno (Coastal Hydraulics Lab., US Army ERDC, Vicksburg, MS), Michael J. White (Construction Eng. Res. Lab., US Army ERDC, Champaign, IL), Christa Woodley (Environ. Lab., US Army ERDC, Vicksburg, MS), and Mihan McKenna (US Army ERDC, Vicksburg, MS)

Acoustic and seismic signals may be generated and received in land, air, and water domains within a multi-domain environment such as a littoral (nearshore) zone. As the energy moves between domains, the waves undergo complex transformations at the media boundaries. Separate consideration of each physical domain (land, air, water) leads to an incomplete understanding of the full cross-domain wavefield, leading to significant challenges in interpreting signals and creation of accurate models that realistically couple multiple domains. To address this knowledge gap, a team at the US Army Engineer Research and Development Center constructed a simplified, uniform levee to enable the measurement of signals that had

passed through undisturbed boundaries. Sensor types were medium-specific, with microphones in the air, hydrophones in the water, and accelerometers in the soil. Using electric bridge wire detonators (EBW's), impulsive signals were generated in each medium type and measured on sensors in all three media. This presentation begins with an overview of the experimental design, discusses the medium-specific measurement results, and concludes with a discussion of joint multi-domain data analysis results. Approved for public release: distribution is unlimited.

2:35

4pBAb7. Rapid computation of steady state acoustic fields in heterogeneous and nonlinear media using the Acoustic Field Propagator. Ben Cox (Univ. College London, University College London, Gower St., London WC1E 6BT, United Kingdom, b.cox@ucl.ac.uk), Ratan Saha (Indian Inst. of Information Technol. Allahabad, Allahabad, India), Antonio Stanziola, and Bradley Treeby (Univ. College London, London, United Kingdom)

The Acoustic Field Propagator is a method for rapidly computing steady state complex acoustic fields in homogeneous media from spatially arbitrary single frequency sources. Here, an extension of the Acoustic Field Propagator to the computation of steady state fields in arbitrary heterogeneous, absorbing and nonlinear media, using nested iterations of the Convergent Born Series and a multi-frequency Helmholtz equation expansion, will be described.

2:50–3:05 Break

3:05

4pBAb8. A static memory method for modelling time-fractional power law absorption. Matthew J. King (Dept. of Medical Phys. and Biomedical Eng., Univ. College London, Malet Pl. Eng. Bldg., Gower St., London WC1E 6BT, United Kingdom, matthew.king@ucl.ac.uk), Timon S. Gutleb (Dept. of Mathematics, Univ. of BC, London, United Kingdom), Ben Cox (Dept. of Medical Phys. and Biomedical Eng., Univ. College London, London, United Kingdom), and Bradley Treeby (Univ. College London, London, United Kingdom)

The attenuation of ultrasound propagation through tissue is known to follow a frequency power-law in the time domain. This can be modelled as a loss operator in the equation of state within the Euler equations that takes the form of a fraction time derivative operator. This can be treated through a second order accurate transfer between the fractional time derivative and a fractional Laplacian spacial operator in order to avoid storage of the full time history. This is used for example by the k-Wave toolbox. As an alternate finite history methods have been suggested. Building on recent work, here, we re-write the time fractional derivative as a finite sum using a recursion relation. This allows the time fractional derivative to be directly computed with a static memory requirement. For homogeneous media, the advantage of this over the fractional Laplacian methods may not initially be obvious with a base higher memory requirement and computational cost with only a small increase in accuracy. However, upon introducing a heterogeneous medium with regions of different power law attenuation; the increased computational cost of the time fractional method is contained exclusively within the pre-computation, while the increased cost for the fractional Laplacian is applied to each time step.

Invited Papers

3:20

4pBAb9. Elastodynamic property closures of elastic waves in polycrystalline materials. Christopher M. Kube (Eng. Sci. and Mech., The Penn State Univ., 212 Earth and Eng. Sci. Bldg, University Park, PA 16802, kube@psu.edu)

In polycrystalline materials like many metals, grainy microstructures significantly influence elastodynamics. Bulk waves scattering at grain boundaries cause attenuation and speed variation in waves. This depends on grain characteristics, including local elasticity and spatial properties. These are modeled statistically to homogenize the microstructure or elastodynamic fields, within bounds. For example, the elasticity of a homogenized medium can't exceed that of individual grains. 'Property closure' is the range within which microstructures yield specific properties like Young's modulus. This presentation introduces property closures for modeling elastic wave properties, specifically attenuation and velocity, in metal alloys. The focus is on the modeling framework and integrating microstructure statistics. An excting prospect of this work is in linking measurable wave properties with other physical quantities dependent on microstructure.

3:40

4pBAb10. Ultrasonic scattering from polycrystals with transversely isotropic material texture. Nathanial Matz (Mech. and Mater. Eng., Univ. of Nebraska-Lincoln, Lincoln, NE), Waled Hassan (Rolls-Royce Corp., Indianapolis, IN), and Joseph A. Turner (Mech. and Mater. Eng., Univ. of Nebraska-Lincoln, University of Nebraska-Lincoln, Lincoln, NE 68588, jaturner@unl.edu)

During laser-based metal additive manufacturing (AM), the melted powder is subjected to rapid cooling and the resulting microstructures often have material texture. In many cases, the texture is uniaxial (i.e., transversely isotropic) such that one symmetry axis defines the material response. Therefore, ultrasonic inspection methods that exploit the scattering from the microstructure are complicated by the resulting texture which affects the coherent propagation and scattering. In this presentation, a generalized approach is described in which the covariance of the elastic modulus tensor for a uniaxial ensemble of cubic crystals is expressed in terms of a fundamental set of constants. These constants are determined for an arbitrary texture from synthetic polycrystals created using DREAM.3D. With this information, calculations for wave velocity, attenuation, and diffuse scattering can be made efficiently for any wave type and propagation direction relative to the material symmetry axis. Results are compared with analytical expressions for simplified cases and then more generalized textures are examined. Finally, prospects for characterization of components created using metal AM are discussed within the context of input data from electron backscatter diffraction measurements. These results are expected to provide insight regarding the inversion of measurement data for texture characterization.

Contributed Paper

4:00

4pBAb11. The effects of nonlinear approximation to mass operator on the wave propagation in polycrystalline materials. Marzieh Bahreman (212 Earth and Eng. Sci. Bldg., Penn State Univ., University Park, State College, PA 16802, mmb7533@psu.edu), Anubhav Roy, and Christopher M. Kube (212 Earth and Eng. Sci. Bldg., Penn State Univ., University Park, PA)

Existing theoretical models utilize the first or third-order smoothing approximations (FOSA and TOSA) to predict attenuation and wavespeed in polycrystalline materials. The FOSA-based model involves a Born approximation, limiting its application to the stochastic (low-frequency) scattering regime. This restriction is a significant factor contributing to its disagreement with FE-based models. The model based on TOSA was subsequently developed, incorporating two additional multiple scattering terms, and increasing the degree of heterogeneity. To extend into the high-frequency regime, the TOSA-based model was solved using numerical integrations. This presentation highlights the analytical solution of the Dyson equation based on the non-linear approximation, wherein the mass operator expression incorporates the mean dyadic Green's function. Evaluating this equation with consideration of higher-order correlations addresses potential truncation errors associated with FOSA and TOSA approximations and encompasses the full range of frequencies from Rayleigh to the geometric asymptote. This presentation will also demonstrate a comparison of the results obtained from the nonlinear approximation with those from FOSA and TOSA across a range of polycrystalline metals.

Invited Paper

4:15

4pBAb12. Characterization of complex stiffness and thickness of isotropic, viscoelastic plates using multi-modal Lamb waves. Clément Despres (Université de Sherbrooke, Sherbrooke, QC, Canada), Patrice Masson (Université de Sherbrooke, Sherbrooke, QC, Canada), Eric Ducasse (I2M, Université de Bordeaux, Bordeaux, France), Michel Castaings (I2M, Université de Bordeaux, Bordeaux, France), and Nicolas Quaegebeur (Université de Sherbrooke, Mech. Eng. Dept., 2500 blvd Universite, Sherbrooke, QC J1K 2R1, Canada, Nicolas.Quaegebeur@USherbrooke.ca)

This paper presents an approach exploiting the sensitivity of Lamb waves for characterizing the mechanical and geometrical properties of plates. The analytical sensitivity functions are first derived in the case of an isotropic plate and are integrated into an iterative inverse problem to optimize its mechanical and geometrical properties based on a zero-finding approach (Gauss-Newton algorithm for a multivariable problem). This method is validated numerically for a viscoelastic plate and shows high accuracy and low computational cost when compared to existing methods. The design af an experimental setup based on dedicated air-coupled transducers is proposed by optimizing the shape and size of the transducers in order to increase sensitivity of the measred signals with respect to the mechanical parameters of the structure of insterest. Experimental validation demonstrates the ability of the methodto assess simultaneously the viscoelastic properties and the thickness of isotropic plate-like structures.

Contributed Papers

4:35

4pBAb13. Shear wave propagation in a fiber-laden viscoelastic waveguide under prestress: Inverse modeling challenges. Lara Nammari (Biomedical Eng., Univ. of Illinois Chicago, 851 S. Morgan St., Rm. 218 SEO (MC 063), Chicago, IL 60607, lnamma2@uic.edu), Dieter Klatt, and Thomas Royston (Biomedical Eng., Univ. of Illinois Chicago, Chicago, IL)

The functional role of skeletal muscle and the hierarchal microstructure and arrangement of fibers within it results in anisotropy and inhomogeneity in both material properties and imposed stresses. Dynamic elastography reconstruction methods for estimating muscle tissue viscoelastic properties that are based on assumptions of homogeneity, isotropy and only bulk wave motion may produce inaccurate estimates. Biases may be introduced in reconstruction by homogenizing muscle with axially aligned fibers and approximating it as transversely isotropic. The significance of these biases, and their interplay with imposed stresses and confounding waveguide effects due to small cross-sectional dimensions, is quantified with a series of numerical finite element and experimental elastography studies on cylindrically-shaped fiber-laden muscle phantoms, with varying fiber dimensions. Specifically, numerical simulations of elastography are conducted on 4-, 60- and 113-fiber models with the aligned fiber cross-sectional area fixed (reduced fiber diameter as fiber count increases) and comprising approximately 40% of the cross-sectional area of the cylindrical phantom that is also undergoing tensile axial pre-loading. Fiber elastic moduli twice that of connective tissue are considered. Experimental studies of the 4-fiber model are used to validate the numerical model. [Funding support: NSF 1852691 and NIH AR071162]

4:50

4pBAb14. Rayleigh-Lamb wave propagation in a prestressed transversely isotropic viscoelastic waveguide: Inverse modeling challenges. Alexandra Vorobyeva (Biomedical Eng., Univ. of Illinois Chicago, Chicago, IL), Qifeng Wang (Northwestern Univ., Evanston, IL), Dieter Klatt (Biomedical Eng., Univ. of Illinois Chicago, Chicago, IL), Kenneth Shull (Northwestern Univ., Evanston, IL), Eric J. Perreault (Northwestern Univ., Evanston, IL), and Thomas Royston (Biomedical Eng., Univ. of Illinois Chicago, 851 S. Morgan St., Rm. 218 SEO (MC 063), Chicago, IL 60607, troyston@uic.edu)

The functional role and structure of skeletal muscle results in anisotropy in both material properties and imposed stresses, as well as waveguide effects. Dynamic elastography reconstruction methods for estimating muscle tissue viscoelastic properties that are rooted in assumptions of isotropy and bulk wave motion may produce inaccurate estimates. The superposition of axially-aligned orthotropy (transverse isotropy) in material properties and axially-aligned prestress conditions due to passive stretch or muscle activation makes it difficult to independently discern how much of the apparent anisotropy is due to the muscle material or the imposed stress field. Furthermore, this stress field may result in large strain conditions that require use of higher order terms in the stress-strain relationship. The significance of this confounding condition and strategies for decoupling material and stress-based anisotropy are investigated with a series of numerical finite element and experimental elastography studies using scanning laser Doppler vibrometry and magnetic resonance elastography. Shear and Rayleigh-Lamb wave motion is studied in a polymeric muscle phantom that is in the shape of a rectangular rod and has either isotropic or transversely isotropic material properties under zero stress conditions. [Funding support: NIH AR071162]

Session 4pEA

Engineering Acoustics: General Topics in Engineering Acoustics

Pranav Agrawal, Chair
Civil and Environmental Engineering, University of California, Los Angeles, 580 Portola Plaza, 5731 Boelter Hall, Los Angeles, CA 90095-1593

Contributed Papers

1:00

4pEA1. The feasibility of building an impedance tube to measure sound absorption coefficients of materials. Nicholas Zomparelli (Aercoustics Eng. Ltd., 1004 Middlegate Rd., Ste. 1100, Mississauga, ON L4Y 1M4, Canada, nicholasz@aercoustics.com)

This presentation focuses on the design and build of an impedance tube to measure sound absorption coefficients of materials. A practical approach was taken to develop an impedance tube from readily available 'off the shelf' materials, with the ultimate goal of determining feasibility and accuracy of such a build. The design theory from multiple sources will be discussed, as well as a detailed breakdown of the dimensions and materials used for the apparatus. Measurement methodologies, testing set-up and data processing techniques are also presented to contextualize the results. Finally, comparisons of the tube results are made to existing known sound absorption coefficient data of materials to ultimately assess the accuracy of the low-cost impedance tube and any deficiencies. Methods to improve the design of the tube are also explored, as well as future design improvements.

1:15

4pEA2. PCB-based miniature vibro-tactile display for the visually impaired. Maijie Xiang (Mech. Eng., Tufts Univ., 485 Foley St., unit 1706, Somerville, MA 02145, Maijie.Xiang@tufts.edu), Robert D. White (Mech. Eng., Tufts Univ., Medford, MA), and Jonathan Bernstein (Draper Lab, Cambridge, MA)

An estimated 2.2 billion people worldwide are visually impaired (WHO, 2019), yet current Braille readers are limited to one or two rows of text and cannot display images. Thus, a low-cost, electronically refreshable tactile display for accessing graphical and textual information is needed. We are developing such a display using a printed circuit board (PCB) as the substrate and bottom electrode array, and a metalized Kapton film as the vibrating membrane. Punched foam tape is used as a spacer to create a gap between the board and film so the pixels can be actuated electrostatically. The current prototype has 6 pixels at 3.5 mm spacing, Vibration amplitudes of 8 microns peak-to-peak were achieved in a prototype using a drive voltage of 600V at 200 Hz. This is sensible to the human touch. Further optimization requires matching the mechanical impedance of the finger, and maximizing vibration amplitude in the preferred sensible frequency range between 50 and 300 Hz. Finite element analysis (FEA), laser vibrometry, and nano-newton force-displacement measurements have been used to characterize the system. Future versions will increase amplitude, reduce drive voltages, and increase display resolution to larger arrays of more closely spaced pixels through further miniaturization.

1:30

4pEA3. Speakers—As a sensor for detecting acoustic loads with Artificial Intelligent (AI). Noori Kim (Purdue Univ., West Lafayette, IN), Hui Jun Kim (Dong-eui Univ., Bu, Korea (the Republic of)), Tobias S. Yoo (Purdue Univ., West Lafayette, IN), and Sung-Hee Kim (Dong-eui Univ., 176 Eomgwangro, Busanjin-gu, Busan, Korea (the Republic of), sh.kim@deu.ac.kr)

The potential of using speakers as a sensor to detect ear canal conditions was demonstrated previously. This research contains our ongoing and continuous effort to utilize a single speaker as a sensor by measuring electrical impedance varying acoustic loads. Electrical impedance data (magnitude and phase) from six different acoustic load conditions were collected as features for machine learning (ML) model training. To enhance the learning performance, the data were pre-processed and augmented with normalization and level-shifting techniques, respectively. The raw data were converted to images to optimize the learning performance to classify acoustic loads from the impedance measurement. Several forms of images were experimented such as magnitude only, overlapped magnitude and phase, and rectangular form. A total of 2100 data (350 each) were used with CNN-based State of the Art (SOTA) models such as AlexNet, ResNet, and DenseNet. Both binary and multiclass classifications were performed, showing 0.9716 average accuracy and 0.907 accuracy, respectively. This innovative single-speaker approach using impedance as ML features is poised to revolutionize traditional acoustic sensing research by harnessing the limitless power of AI.

1:45

4pEA4. Experimental verifications of perforation-modulated slow-wave phenomenon. Sihui Li (Mech. Eng., Hong Kong Polytechnic Univ. Shenzhen Res. Inst., Hong Kong), Xiang Yu (Mech. Eng., Hong Kong Polytechnic Univ. Shenzhen Res. Inst., Hong Kong), and Li Cheng (Mech. Eng., Hong Kong Polytechnic Univ. Shenzhen Res. Inst., Hung Hom, Hong Kong 852, Hong Kong, li.cheng@polyu.edu.hk)

Slow wave phenomenon has been theoretically and numerically predicted inside a in a retarding based on the so-called sonic black hole (SBH). A perforation-modulated SBH (PMSBH) retarding structure is then proposed, in which the two physical processes (sound velocity reduction and absorption) can be balanced to enhance the black hole effects by modulating the perforation parameters. This work presents an experimental effort to confirm the theoretically predicted slow wave phenomenon as well as the ABH-induced sound absorption inside a PMSBH. Alongside some theoretical development, 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.

2:00

4pEA5. A fluid filled MEMS hydrophone. Georgios Karamanis (Mech. Eng., Tufts, 574 Boston Ave. Rm. 318, Medford, MA 02155, georgios.karamanis@tufts.edu), Jonathan Anderson, Lalitha Parameswaran (Adv. Mater. and Microsystems, MIT Lincoln Lab., Lexington, MA), James Vlahakis (Mech. Eng., Tufts, Medford, MA), Livia Racz, Daniel Freeman (Adv. Mater. and Microsystems, MIT Lincoln Lab., Lexington, MA), and Robert D. White (Mech. Eng., Tufts Univ., Medford, MA)

MEMS hydrophones are of interest to provide a small form factor acoustic sensing capability in water. The majority of previous work on MEMS hydrophones (e.g., Bernstein 1997, Moon, 2010, Gu, 2018) use air backed diaphragms in order to provide increased sensitivity. However, this limits the operating depth of the device due to a reduction in the diaphragm burst pressure. In this work we investigate an architecture that has the potential to increase the operating depth by filling the backing cavity with a heavy fluid. We have designed architectures with multiple acoustic ports (Moon, 2010) and coupled piezoelectric diaphragms, in a bid to maintain sufficient sensitivity while incorporating the filling fluid. The sensing elements are Parylene coated, repackaged Vesper VM1000 aluminum nitride MEMS microphones. It was reported (Travaglione, 2018) that such a device can sense underwater sound, but characterization was limited. In our prototypes, the microphone die are co-packaged onto modified printed circuit boards which include the acoustic coupling elements and required preamplifiers. We report on computational predictions of sensitivity and resolution, and provide initial test results as part of our ongoing investigation. Support from OUSD (R&E) Innovation and Modernization Office.

2:15

4pEA6. The simulation and fabrication of the void head mass by an increase in the bandwidth ultrasonic transducers. Reza Rahnama (Mater. Sci. and Eng., Shiraz Univ., Iran Shiraz, Shiraz 7157743364, Iran (the Islamic Republic of), rezarahnamac@gmail.com)

To achieve the bandwidth of the frequency bandwidth, a new transducer of its main components is developed. The better performance of a transducer is proportional to its head weight. In this study, the internal part of the transducer head is hollow, which reduces the weight of the transducer head and is a factor in increasing the frequency bandwidth. In this paper, the finite element method (FEM) has been used to validate the results obtained from the effects of structural changes in the head applied to the transducer. In this paper, to achieve higher frequency bandwidth, the Tonpilz transducer structure was studied via analysis and computational results using the COMSOL Multiphysics software. The optimized transducer has higher frequency bandwidth, which confirms the effect of using the void of the transducer head mass.

2:30

4pEA7. Sound absorption performance of double layer sandwich structure containing micro-perforations and extended necks. Zacharie Laly (Dept. of Mech. Eng., Sherbrooke Univ., 2500, boul. de l'Université, Sherbrooke, QC J1K 2R1, Canada, zacharie.laly@usherbrooke.ca), Chris Mechefske (Dept. of Mech. and Mater. Eng., Queen's Univ., Kingston, ON, Canada), Sebastian Ghinet, Tenon Charly Kone (Aerosp., National Res. Council Canada, Ottawa, ON, Canada), Noureddine Atalla (Dept. of Mech. Eng., Sherbrooke Univ., Sherbrooke, QC, Canada), and Behnam Ashrafi (Aerosp. Manufacturing Technol. Ctr., National Res. Council Canada, Montreal, QC, Canada)

In this study, a double layer sandwich panels made of micro-perforated plate and perforated plate with extended necks is proposed and its acoustic attenuation is studied using the finite element method. The top panel of the structure is micro-perforated and is connected to a first honeycomb structure cells. The internal panel is perforated, and a neck is connected to each perforation and extends into each cell cavity of the second honeycomb structure. The proposed material design is made of 33 micro-perforations and 33 extended necks and has excellent mechanical stiffness due to the honeycomb structure performance. This study shows how to design the parameters of the necks and the micro-perforations to widen the sound absorption frequency band. With identical micro-perforation diameters and identical necks, the sound absorption coefficient presents two resonance peaks where the surface impedance is close to the impedance of air. The number of the sound absorption peaks increases when different micro-perforation diameters and different necks are used while the sound absorption of conventional double layer sandwich structure is zero over the entire frequency range. The proposed material design can be used to attenuate the noise in various engineering fields that require good bending stiffness.

2:45–3:00 Break

3:00

4pEA8. Characterizing the absolute sound levels of ambient noise in a water tank. Benjamin L. White (Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, bw392@byu.edu), Corey E. Dobbs, and Tracianne B. Neilsen (Phys. and Astronomy, Brigham Young Univ., Provo, UT)

The underwater acoustic measurement tank used at Brigham Young University is affected by several external sound sources which, although easily identified or hypothesized about on a superficial level, must be investigated in order to determine their specific effects on data collection. Therefore, our goal is to characterize the absolute sound levels of these external acoustic sources. The measurements are conducted using B&K and TC4038 hydrophones. Data are collected over frequency bands within the "flat" range of these hydrophones, facilitating the consistent identification of ambient sounds across several bands. Measurements are taken at various points of the tank in order to determine to what extent tank location affects the ambient sound. Data in Volts are converted to Pascals by applying sensitivity values. To evaluate the accuracy of levels obtained, the same method is applied to the signal from a known source and compared to the levels obtained via a deconvolution approach. The conclusions of this study are applicable in obtaining sound levels in our own tank and in other environments (e.g., aquariums). [Undergraduate Research supported by the College of Physical and Mathematical Sciences, Brigham Young University]

3:15

4pEA9. Comparative numerical investigation of flow-induced noise characteristics of high-speed trains using high-resolution compressible Large Eddy Simulation. Kwongi Lee (School of Mech. Eng., Pusan National Univ., 2, Busandaehak-ro 63 beon-gil, Busan, Korea (the Republic of), dlrnjsrl93@pusan.ac.kr), Ceholung Cheong (Ctr. for Adv. Refrigeration, Air-Condition and Energy, Busan, Korea (the Republic of)), and Jaehwan Kim (Hyundai Rotem, Uiwang-si, Korea (the Republic of))

The South Korean Ministry of Land, Infrastructure and Transport has developed a "Comprehensive Plan for 400km/h High-Speed Rail" aimed at enhancing the operational speed of the high-speed railways, currently running up to 300km/h. A short-term goal is to elevate the operating speed of high-speed trains to 370 km/h, up from 320 km/h. Because aerodynamic noise proportionally escalates with the 6th powers of the speed, aerodynamic noise becomes more significant at higher speeds. Consequently, there's a pressing need for design solutions that reduce aerodynamic noise in high-speed trains. This study involves an aeroacoustic analysis using real-scale models of the current model and the preliminary design targeting 370 kph operation. Each model's 8-car formation has been simplified to a 5-car setup. A challenge in predicting aerodynamic noise is the generation of detailed sound sources in the near field and precise noise propagation in the acoustic field. For that, a three-dimensional compressible Large Eddy Simulation technique is employed, utilizing high-resolution grids. This allows for concurrent computation of the external flow and acoustic fields for a real-scale, high-speed train in an open environment. The analysis comprehensively examines the aerodynamic and aeroacoustic properties of each train car, including the major contributors to aerodynamic noise in high-speed trains. The radiated noise is predicted using the Ffowcs Williams and Hawkings equation and is further examined in relation to vortex sound sources.

3:30

4pEA10. Modular design for optimized acoustic properties of multilayer system. Tao Yang (Tech. Univ. of Munich, Munich, Germany), Martin Eser (Tech. Univ. of Munich, Boltzmannstrasse 15, Munich 85748, Germany, m.eser@tum.de), Marcus Maeder (Tech. Univ. of Munich, Garching near Munich, Germany), and Steffen Marburg (School of Eng. and Design, Tech. Univ. of Munich, Garching, Germany)

It has been demonstrated that multilayer systems have the potential for excellent sound absorption, effectively attenuating sound across a wide range of frequencies and offering a versatile solution for addressing acoustic challenges in various environments. A key consideration is the design of a multi-layer structure and the parameters of each layer to achieve optimal sound absorption performance. Modular design is a crucial aspect of this work, involving the creation of smaller, units that can be customized and combined to achieve specific sound absorption properties. This study applies numerical optimization, inversion, and matrix methods to multilayer sound absorber systems, aiming to provide customized parameters for the modular design of fibrous materials with exceptional sound absorption properties. The effectiveness of the modular design will be validated and improved through impedance tube measurements of the manufactured multilayer systems, ensuring that the resulting sound absorbers meet the desired acoustic performance across diverse real-world scenarios.

3:45

4pEA11. Wind shadowing of a sonic anemometer in low Reynolds number flows. Julia Huckaby (Mech. Eng., Tufts Univ., 200 College Ave., Medford, MA 02155, julia.huckaby@tufts.edu), Robert D. White (Mech. Eng., Tufts Univ., Medford, MA), Ian Neeson (VN Instruments, Brockville, ON, Canada), Don Banfield, and Anthony Colaprete (NASA Ames Res. Ctr., Mountain View, CA)

Computational Fluid Dynamics (CFD) simulations have been carried out to better understand the wind shadowing effects on a sonic anemometer system. The anemometer will measure wind speed on the surface of Mars and the stratosphere of Earth.The wind velocity vector is measured through the difference in acoustic time of flight in 3 directions. However, the structure itself disrupts the flow and creates an internal wake. To understand this and generate correction factors, CFD simulations were run under Mars surface conditions while varying incident angle and flow speed. Spalart-Allmaras turbulence models were employed in both steady and unsteady Reynolds Averaged Navier Stokes (RANS) frameworks and compared to a laminar model. Due to the low Reynold's number (<150) we observe no shedding at speeds below 10 m/s. At speeds below 10 cm/s the flow diverts around the structure and sensitivity drops. At intermediate speeds near 5 m/s the wind shadow ratio (WSR) varies substantially with incident wind angle. Design changes to the structure allowed us to improve the worst WSR from 38% to 47%. We were also able to quantify correction factors. Validating comparisons to water tunnel and wind tunnel measurements will be presented.

4:00

4pEA12. Analysis of the linear regime of a power amplifier for underwater acoustics experiments. Natalie Bickmore (Phys. and Astron., Brigham Young Univ., N283 ESC, Provo, UT 84602, njbickmore@gmail.com), Tracianne B. Neilsen, and Corey E. Dobbs (Phys. and Astron., Brigham Young Univ., Provo, UT)

When transmitting sound in a laboratory water tank, a power amplifier is needed to boost the generated signal. If the amplitude of the generated signal is too high, the power amplifier introduces harmonics into the transmitted signal. Measurements were taken to determine the maximum input amplitude for which the TEGAM 2350 power amplifier, which provides x50 amplification, has a linear response. To evaluate when harmonics are generated, acoustical measurements were taken in our acrylic 1.22 m by 3.66 m tank filled with 0.5 m of water. Custom LabView software generated a chirp with a specified amplitude. This signal was sent through the power amplifier and an impedance matching transformer to a B&K 8103 transmitter. The sound was received on the B&K 8103 hydrophones. Both were attached to two different UR10e robots that could be positioned throughout the tank. The receiver hydrophone was moved to three different locations in the tank for a variety of chirp amplitudes to find the maximum amplitude without harmonic generation. Results of these measurements are shown for three frequency bands: 10–50 kHz, 50–100 kHz, and 100–150 kHz . [Undergraduate research supported by the College of Physical and Mathematical Sciences, Brigham Young University]

4:15

4pEA13. Design and construction of the Texas Christian University impedance tube. Claire Elrod (Mech. Eng., Texas Christian Univ., 28840 W Bowie St., Fort Worth, TX 76109, claire.elrod@tcu.edu) and Hubert S. Hall (Mech. Eng., Texas Christian Univ., Fort Worth, TX)

The two-microphone impedance tube test method is a well-established and widely used technique for determining the acoustic absorption coefficient and impedance ratio of materials. This method uses two closely spaced microphones to simultaneously measure the incident and reflected sound waves. A two-microphone impedance tube measurement system made of 6061-T6 Aluminum with a diameter of 3 inches, a 0.5 inch wall thickness, and microphones spaced 2.7 inches apart has been constructed for undergraduate research at Texas Christian University (TCU). These geometrical values suggest a usable frequency range of 50 Hz to 2637.77 Hz as referenced in ASTM Standard E1050-19. Validation of the system was achieved by taking measurements on Owen Corning Type 705 pressed fiberglass board with a 1-inch thickness and comparing them to absorption data provided by the manufacturer. Additional validation measurements were taken without a test sample in place. All validation tests suggest that the TCU impedance tube is an accurate measurement system.

4:30

4pEA14. Acoustic characteristics of OLED panel gaming monitor. Sungtae Lee (LG Display, 245 LG-ro, Paju-si, Gyunggi-do, Korea (the Republic of), owenlee@lgdisplay.com) and Hyungwoo Park (ICT, Dong-Seoul Univ., Seongnam-si, Gyeonggi-do, Korea (the Republic of))

Previous studies have presented that OLED panels can be fully utilized as high-quality audio devices. Through continuous research, methods for improving the sound quality of panel speakers and implementing multi-channel sound have been studied. OLED panel speakers have undergone continuous improvement, allowing this technology to be used in gaming monitors, and research is underway to replace existing desk soundbars or headphones. This OLED panel speaker technology improves picture and sound quality not only for games but also for various contents. In particular, information delivery methods that utilize direct sound coming through the screen, such as CSO(Cinematic Sound OLED), are sophisticated, accurate, and provide the best sense of realism. Sound generated close to the human auditory organ, such as headphones or earphones, impairs hearing health. The proposed method has the advantage of matching the focus of the screen and the sound, as well as the connection between the human auditory organ and the point of sound generation. It has the advantage of being able to maintain a certain distance. In particular, content such as games needs to have a fast response speed of the display, as well as the location and focus of the sound to be clearly defined. In this study, it is introduce the results of analyzing the acoustic characteristics of the OLED panel speaker using the proposed method and utilize this to contribute to commercialization.

4:45

4pEA15. Extending the spatial extent of steady state broadband noise signals using time reversal. Rylee Russell (Phys. & Astron., Brigham Young Univ., BYU, N283 ESC, Provo, UT 84602, rsrussell01@gmail.com) and Brian E. Anderson (Phys. & Astron., Brigham Young Univ., Provo, UT)

Exciting structures with broadband noise can be used to assess structural health by how it responds to the noise. Here time reversal (TR) is used to focus high-amplitude acoustic energy with the goal to use it to excite a structure. Equalization methods help achieve a desired spectral shape at the location of the TR focus. It is desired to increase the spatial extent of the focusing to excite larger regions. This study explores methods to expand the spatial extent first through the superposition of focusing to multiple locations and second by using a spatial inverse filter. These methods widen the spatial extent of the focusing but can come at the cost of the desired spectrum and/or overall level. The overall goal is to see if TR excitation yields higher amplitude equalized noise than the standard broadcast of it.

Session 4pED

Education in Acoustics, Biomedical Acoustics, Musical Acoustics, Physical Acoustics, and Structural Acoustics and Vibration: Teaching Acoustics With (or Without) Math

Kurt R. Hoffman, Cochair
Physics, Whitman College, 345 Boyer Ave, Hall of Science, Walla Walla, WA 99362

Jill A. Linz, Cochair
Physics, Skidmore College, 815 N Broadway, Saratoga Springs, NY 12866

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

Invited Papers

1:00

4pED1. Using musical acoustics to motivate math learning. Andrew A. Piacsek (Phys., Central Washington Univ., 400 E. University Way, Dept. of Phys., Ellensburg, WA 98926-7422, andy.piacsek@cwu.edu)

A course on musical acoustics, at the high school or introductory college level, can be an effective vehicle for improving student math skills. More importantly, it provides an opportunity to develop an appreciation for the utility of mathematics among skeptical students, including those from groups, such as women and minorities, who have historically been steered away from quantitative studies. Having an answer for the rhetorical "What is math good for?" can boost a student's motivation for learning math and open the door to consideration of STEM careers. This presentation gives an overview of how math skills are explicitly addressed throughout a 10-week (one quarter) general education course, "Physics of Musical Sound," taught at Central Washington University. Examples of specific in-class exercises will be described, as well as how math is incorporated into collaborative term projects. A discussion will be provided of what has worked, and not worked, throughout 25 years of teaching this course.

1:20

4pED2. Using music to motivate learning mathematics in acoustics courses. Gordon P. Ramsey (Phys. Dept., Loyola Univ. Chicago, Loyola University Chicago, Chicago, IL 60660, gramsey@luc.edu)

Mathematics often intimidates students, especially if they feel unprepared for the math level in the course. However, when they see how math and music are intimately related, their attitude often changes. Many fundamental aspects of music are described by math. These include note frequencies, chords and scales, consonance and dissonance, and instrument tuning. 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 designing instruments that play the music. Instruments must be designed to replicate the note frequencies, while their geometry allows them to be being playable by humans. Mathematics is at the heart of instrument design. I will suggest ways to present these concepts to students for their greater appreciation of how acoustics and math are related.

1:40

4pED3. Improving quantitative literacy through a general education acoustics course. Kurt R. Hoffman (Phys., Whitman College, 345 Boyer Ave. Hall of Sci., Walla Walla, WA 99362, hoffman@whitman.edu)

Most Colleges and Universities have some form of quantitative reasoning requirement for graduation. A narrow view of this requirement is that students need to do some mathematics in the course to demonstrate some facility at calculating quantities. Additionally, students often discuss their discomfort with mathematics when discussing their anxiety about registering in a non-majors physics course. In this presentation, I will discuss strategies used in a musical acoustics course for non-science majors to address proportionalities, graphical representations, and approximations to help students get past equations to think about the underlying physical relationships. Often, I find that students actually are much better at using mathematics than their self-assessments. By giving students different contexts for mathematical thinking, the quantitative elements make more sense to them and calculations become less intimidating. The examples I will discuss focus on using classroom-based experiments to identify relationships between dependent and independent variables on topics such as musical scales, simple oscillators, and fourier analysis.

2:00

4pED4. Building a mathematical model for a simple harmonic oscillator that uses educational methods found in both physics education research and in the language disciplines that make it accessible to undergraduate students in an introductory musical acoustics course. Jill A. Linz (Phys., Skidmore College, 815 N Broadway, Saratoga Springs, NY 12866, jlinz@skidmore.edu)

At the basis of any course in acoustics is the fundamental idea of the simple harmonic oscillator. The term alone is confusing to students with little to no background in physics or math. For courses in musical acoustics at the undergraduate level, this topic is often minimized due to the lack of preparation. This, in turn, results in a more superficial approach to the advanced topics. While deriving the mathematical treatment from first principles is out of reach to these students, approaching the math itself as a language where they are building a description of a simple mass and spring system in their new, yet somewhat familiar, language can be accomplished through pictures, graphs and hands on activities. Students begin to build a vocabulary of "words" that can be strung together in "sentences" that tell the story of how the motion of a mass on a spring is produced. Emphasis is placed on the analogous comparison of physical properties by relating variables such as amplitude and frequency to that of volume and pitch. This model can then be used as a building block to the understanding of how sound is produced, propagated and perceived.

2:20–2:35 Break

2:35

4pED5. Teaching acoustic and electromagnetic waves. David A. Brown (ECE/CIE, Univ. of Massachusetts Dartmouth, 151 Martine St., UMass Dartmouth, Fall River, MA 02723, dbAcousticsdb@gmail.com)

Students have observational experience with waves by speaking and hearing, seeing, and feeling but their use of mathematics may be delayed until introductory physics at middle school, high school, and/or college. Students following a university physics or engineering bachelor's degree are usually required to take a course on electromagnetic wave theory because of the foundational science and/or applications such as communications and observation. Regrettably, it is common that a mathematical description of various wave propagations may be acquired but without a thorough understanding of the underlying physics – especially in electromagnetics. Thus, we have tried to develop an introductory graduate level course that teaches electromagnetics and optic waves and its applications on the foundation of first understanding acoustics. This serves as a preparation for research or vocation. Topics include traveling waves, standing waves, wave impedance, radiation patterns, interference, interferometry, sonar and radar, imaging, waveguides in more. Examples of introducing the mathematics of solution first and then derivations of wave equation is presented. The goal is that mathematics should be a useful language to communicate physical information and that acoustic demonstrations should reinforce learning in multiple physical modalities.

2:55

4pED6. Learning to read an acoustics equation as a descriptive sentence. Daniel A. Russell (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg, University Park, PA 16802, dar119@psu.edu)

Teaching theoretical acoustics at the graduate level requires the use of mathematics, and often the math can be difficult enough that students get lost in the math and can't see the underlying physical concepts the equations are describing. What many students don't realize is that a mathematical equation is just a shorthand notation that condenses a long and complicated sentence into a few symbols. Being able to read and interpret an equation as a sentence (which is more than just reading the names of the symbols or variables) is a valuable skill that can help students grasp the underlying physical concepts. This talk will provide several examples from fundamental acoustics of how helping students learn to read and interpret mathematical equations as sentences can help them reveal and grasp the physical concepts the equations are describing.

Contributed Papers

3:15

4pED7. When is an approximate solution better than an "exact" solution? Steven L. Garrett (Salinas Union High School District, 1736 Lowell St., Seaside, CA 93955, sxg185@psu.edu)

The solution to any problem always has two parts: (1) The numerical "result," with units and possibly a related uncertainty, that is usually provided by a calculation (analytic or numerical) and (2) an intuitively satisfying explanation of that "result." The "exact" results will be presented for the frequency of the first standing-wave mode of a mass-loaded string (*i.e.*, pendulum) as well as the "exact" result for the frequency shift that occurs when the stiffness of a piano string is combined with the string's tension as the restoring forces. The modal frequency of the mass-loaded string will then be approximated by synchronizing the frequencies of a complete sinusoidal half-wavelength connected to a very short pendulum. The exact solution for the "stiff string" problem that was given by Morse in his second edition of *Vibration and Sound* will be shown to be incorrect. A simple approximation, using Rayleigh's method, will provide the correct result. Both approximate methods create an intuitively satisfying explanation for each result. These examples will also confirm John von Neumann's assertion (in 1947) that "Truth is much too complicated to allow anything but approximations."

3:30

4pED8. Early development of a set of interactive tools for undergraduate acoustics education. Noah J. Parker (Acoust., Penn State, 201 Appl. Sci. Bldg., University Park, PA 16802, np.acoustics@gmail.com) and Daniel A. Russell (Graduate Program in Acoust., Penn State Univ., University Park, PA)

Using interactive learning materials has been found to increase the efficiency of teaching math and physics concepts fundamental to acoustic phenomena; this presentation will discuss work done to explore their use in undergraduate acoustics courses. During the Spring 2024 semester at Penn State, three undergraduate courses in acoustics were offered, each aimed at different types of students at differing levels of mathematical rigor. These were: a general education elective for freshmen from non-STEM majors, a sophomore/junior level course required for communication science disorders majors, and a junior/senior elective for STEM majors. Observation of each class during the semester identified a variety of topics for which a web-based interactive tool could improve student understanding and engagement. This talk will discuss some of the topics identified as potentially benefiting from an online interactive tool, with a specific focus on one interactive webpage which may serve as an example template for the style and delivery of future developments. This specific interactive webpage

explains basic trigonometric functions in a manner approachable by students at any level of mathematical confidence, by illustrating geometric implications with interactive graphs, explaining with a conversational tone, and addressing the etymology of some of the mathematical terms.

3:45

4pED9. A tutorial and assignment on the analysis and syntheis of music to provide acoustic foundations for courses in the Psychology of Music and in Music Cognition. Annabel J. Cohen (Psych., Univ. of PE, Dept. of Psych., University of PE, 550 University Ave., Charlottetown, PE C1A 4P3, Canada, acohen@upei.ca)

Students in courses in the Psychology of Music and in Music Cognition benefit from an understanding of aspects of musical acoustics. A tutorial assignment has been developed to provide hands-on experience in synthesizing acoustic elements and analyzing real music using Audacity®, freely downloadable software. The tutorial emphasizes visualizing waveforms and spectrographs of target sounds. For the synthesis component, students generate sine tones related by small integers on separate tracks and combine the tracks to create complex waves. Students must answer questions regarding frequency, amplitude, pitch, loudness, and sound quality. They are also asked to synthesize a sequence of successive notes of the chromatic scale, so as to understand the significance of the semitone ratio $2^{1/12}$ (i.e., 1.05946), and they are asked to synthesize a sequence of successive notes from the diatonic scale. For the analysis component of the tutorial, students download a favourite piece of music, provide waveform and spectrographic representations, and answer questions about the representation of pitch, loudness, time and timbre. They also locate auditory repetitions and identify different sections of their piece. The students are asked to summarize what they have learned from the assignment. The assignment can be adapted for work in teams, and a detailed grading rubric facilitates student evaluation.

4:00

4pED10. Considerations for mathematical backgrounds and goals of music students in introductory acoustics courses. Daniel Choi (Newcomb & Boyd, 303 Peachtree Ctr. Ave. NE, Ste. 525, Atlanta, GA 30303, dchoi@newcomb-boyd.com)

Many music institutions include an introductory course in acoustics as prerequisites for certain classes and/or requirements for graduation of a degree program. With the topic involving science, there are mathematical concepts that are naturally introduced to the students. For instruction of music students this may result in challenges as the range of their mathematical background is generally larger. Aside from difficulties in the coursework, it is also common where students will not utilize the mathematical concepts from the course throughout their career. Therefore, it is crucial to survey the typical level of the students' interests and skills in mathematics to align it with the course objectives and determine the methods of instruction. This presentation will focus on modifying the level of mathematical instruction based on the goals of typical music students and the real-world applications of the learned content.

4:15

4pED11. Viewpoints on math in acoustics education and their reflections in the acoustics program's curriculum at Purdue University. Yangfan Liu (Ray W. Herrick Laboratories/School of Mech. Eng., Purdue Univ., Ray W. Herrick Labs., 177 S, Russell St., West Lafayette, IN 47907, yangfan@purdue.edu), Junfei Li, J. S. Bolton, and Patricia Davies (Ray W. Herrick Labs., Purdue Univ., West Lafayette, IN)

Faculties in the acoustics and noise control program at Purdue university will share their viewpoints on the roles of mathematics in acoustics education and math instruction practices in the acoustics curriculum at Purdue university. It is suggested to consider mathematical training from two prospective: (1) teaching math techniques and (2) cultivation of a mathematical mindset. Opinions will be shared on how each math training aspect benefits acoustics students with engineer careers and research careers as well as the long-term influences of math education on industry-academia collaboration. We will also briefly discuss how these views on math education are reflected in Purdue's current acoustics curriculum and could guide further improvements of the curriculum.

Session 4pID

Interdisciplinary and Student Council: Hot Topics in Acoustics

Ian C. Bacon, Cochair
Physics & Astronomy, Brigham Young University, 333 W 100 S, Provo, UT 84601

Chirag Gokani, Cochair
Walker Department of Mechanical Engineering, University of Texas at Austin, 10000 Burnet Road, Austin, TX 78758

Heui Young Park, Cochair
Pennsylvania State University, State College, PA 16801

Ann Holmes, Cochair
Psychological & Brain Sciences, University of Louisville, 2082 Douglass Blvd., Apt 5, Louisville, KY 40205

Chair's Introduction—1:00

Invited Papers

1:05

4pID1. Robustness of vortex wave based communications to operational variables. Mark Kelly (GA Tech, 313 Oakland St., Decatur, GA 30030, mkelly75@gatech.edu) and Chengzhi Shi (Univ. of Michigan, Atlanta, GA)

Orbital angular momentum (OAM) based acoustic communications are a promising method of increasing bandwidth in underwater acoustic communications networks as their unique phase patterns form an orthogonal basis set on which communications protocols may be based. The underwater acoustic environment, however, is highly complex and dynamic. These complexities make developing systems capable of long-range communications challenging. Methods for using BELLHOP's ray tracing software to simulate the time series of vortex-wave based communications signals are presented. Additionally, this study explores the robustness of the inner product deconvolution method of decoding OAM-based communications against various operating conditions, considering both environmental and platform variations. The effects on channel cross-talk are assessed against sound speed profile uncertainty, position errors, Doppler Effect, and turbulence in the water column. These analyses shed light on the operational implementation of OAM-based communications systems.

1:20

4pID2. AI-based headphones for augmenting human hearing. Bandhav Veluri, Malek Itani, Tuochao Chen (Univ. of Washington, Seattle, WA), and Shyamnath Gollakota (Univ. of Washington, Paul G. Allen School of CSE, Seattle, WA 98195, gshyam@cs.washington.edu)

Consider being able to listen to the birds chirping in a park without hearing the chatter from other hikers, or being able to block out traffic noise on a busy street while still being able to hear emergency sirens and car honks. We introduce semantic hearing, a novel capability for hearable devices that enables them to, in real-time, focus on, or ignore, specific sounds from real-world environments, while also preserving the spatial cues. Results show that our system can operate with 20 sound classes and that our transformer-based network has a runtime of 6.56 ms on a connected smartphone. In-the-wild evaluation with participants in previously unseen indoor and outdoor scenarios shows that our proof-of-concept system can extract the target sounds and generalize to preserve the spatial cues in its binaural output.

1:35

4pID3. Reciprocal dynamics in nonlinear systems. Andrus Giraldo (Korea Inst. for Adv. Study, Seoul, Korea (the Republic of)), Ali Kogani (Concordia Univ., Montreal, 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)

Nonreciprocal vibration transmission is an important problem from a fundamental perspective and because of the additional functionalities that it enables in mechanical or acoustic devices. A common realization of nonreciprocal dynamics relies on implementation of nonlinear internal forces within the system. In this talk, we identify and discuss different manifestations of nonreciprocity in nonlinear systems. As a necessary condition, nonreciprocal dynamics is realized in nonlinear systems with broken mirror symmetry. We show that a second symmetry-breaking parameter can counteract the original asymmetry and ultimately restore reciprocal dynamics in a system with broken mirror symmetry, even near the system resonances. Thus, breaking the mirror symmetry is a necessary but insufficient condition for realizing nonreciprocity in nonlinear systems. We also highlight the contribution of phase to nonreciprocal vibration transmission by discussing response regimes that are characterized by nonreciprocal phase shifts. Our findings showcase the potential of asymmetry to serve as an additional design parameter for devices that operate based on nonreciprocity.

1:50

4pID4. Hearing your voice in the crowd: How do bats echolocate in dense swarms? Laura Kloepper (Dept. of Biological Sci., Univ. of New Hampshire, 230 Spaulding Hall, Durham, NH 03824, laura.kloepper@unh.edu), Stephen P. Blackstock (Univ. of Texas at Austin, Austin, TX), Amaro Tuninetti (Dept. of Biological Sci., Univ. of New Hampshire, Providence, RI), Dieter Vanderelst (Univ. of Cincinnati, Cincinnati, OH), and Michael R. Haberman (The Univ. of Texas at Austin, Austin, TX)

In order to navigate their environment and find food, bats rely on comparing returning echoes to their broadcast signal. Through decades of research, we have a good understanding of how single bats accomplish this task, but we still don't know how bats in dense groups can echolocate without interference from others. A spectral Jamming Avoidance Response (JAR) has long been proposed as one strategy, in which bats flying together shift frequencies to avoid interference. Recent work, however, questions this strategy, and suggests bats may not use any JAR for sensing in groups. Bats have long served as bioinspiration for active sensing devices, so understanding strategies for collective sensing in bats has implications for applications of synthetic sonar. In this talk, I will review the "hot topic" of bat biosonar in swarms and show new data from our research that suggests spectro-temporal variation may play a role for sensing in even the largest, densest bat swarms.

THURSDAY AFTERNOON, 16 MAY 2024 — ROOM 205, 1:00 P.M. TO 5:00 P.M.

Session 4pNS

Noise, Physical Acoustics and Engineering Acoustics: Methods for Community Noise Testing and Analysis II

Alexandra Loubeau, Cochair
NASA Langley Research Center, MS 463, Hampton, VA 23681

Aaron B. Vaughn, Cochair
Structural Acoustics Branch, NASA Langley Research Center, 1 NASA Drive, Hampton, VA 23666

Duncan Halsead, Cochair
Aercoustics Engineering Ltd, 1004 Middlegate Rd, Mississauga, L4Y 0G1, Canada

Contributed Papers

1:00

4pNS1. Automotive tire facility noise characteristics. Peter VanDelden (RWDI, 600 Southgate Dr., Guelph, ON N1G 4P6, Canada, peter.vandelden@rwdi.com), Lorenzo Carboni, and Slavi Grozev (RWDI Windsor, Windsor, ON, Canada)

The environmental noise influence of facilities that change, replace or repair tires has notable acoustic characteristics, including the sound from air impact wrenches. Air impact wrenches are a ubiquitous tool at such facilities, where they are used to remove and replace the nuts that hold wheels on cars and trucks. The sound is commonly emitted through open doorways. Tire facility sound data was evaluated for impulsive characteristics using procedures described in the newest version of ISO 1996-3. The dynamic nature of air impact wrench sound also allowed the ISO 1996-3 approach to be evaluated.

1:15

4pNS2. Acoustic detection and classification of all-terrain vehicles and automobiles in pennsylvania state parks. Carter A. Paprocki (Acoust., Penn State, 201 Appl. Sci. Bldg., University Park, PA 16802, cap6296@psu.edu), Andrew Barnard (Acoust., Penn State, University Park, PA), and Peter Newman (Recreation, Park, and Tourism Management, Penn State, State College, PA)

The preservation of today's natural environment is more important than ever before; with the encroachment of civilization, protected areas are necessary. This protection must go beyond just saving wildlife but also protecting natural soundscapes for visitors to experience "dispersed low-density outdoor recreation." However, some visitors use the wilds for motorized recreation such as Off-Road Vehicles, All Terrain Vehicles, and Side-by-Side Vehicles producing elevated levels of noise which could impact the other

visitors. In order to determine how the protected areas were used, listening stations were placed at high-traffic areas to monitor the soundscape for one summer. This dataset was analyzed to understand the number and impact of anthropogenic noise sources. This research develops a training dataset that could be implemented with supervised machine-learning models to automate event detection and classification.

1:30

4pNS3. Sound levels of the UVU pedestrian bridge. Jacob Sampson (Phys., Utah Valley Univ., 800 W University Parkway, Orem, UT 84058, jacobbsampson@gmail.com), Isaac Setter (Phys., Utah Valley Univ., Orem, UT), Brian D. Patchett (Phys., Utah Valley Univ., Orem, UT), Abolfazl Amin, Joshua Goates (Utah Valley Univ., Orem, UT), and Bonnie Andersen (Phys., Utah Valley Univ., Orem, UT)

Utah Valley University provides a pedestrian bridge to travel from a train station to the school over Interstate-15. Due to traffic noise, people using the bridge could be exposed to sound levels that damage their hearing. Sound level measurements have been made at several locations on the bridge with two different instruments, the first being an Extech (Knoxville, USA) SL400 noise dosimeter and the other being a Larson Davis (Depew, USA) 831C sound level meter. Sound levels have ranged from 55 to 102 dB, depending on time of day and location on the bridge. In addition to our experimental results a COMSOL model was also developed to simulate the interactions of the freeway noise with the bridge, demonstrating how the geometry of the bridge affects the noise exposure of pedestrians. The Occupational Safety and Health Administration (OSHA) limit for non-occupational noise exposure of 100 dB is 15 minutes, suggesting that pedestrians that linger on the bridge could be exposed to damaging levels of sound. This research seeks to better understand and quantify the noise levels that pedestrians experience on the bridge.

1:45

4pNS4. Acoustic measurements of wind farm noise inside homes and comparison with the ISO 9613 standard. Umberto Berardi (Toronto Metropolitan Univ., 350 Victoria St., Toronto, ON M5B 2K3, Canada, uberardi@ryerson.ca), Gino Iannace, Amelia Trematerra, and Antonella Bevilacqua Bevilacqua (Università della Campania Luigi Vanvitelli, Parma, Italy)

To achieve the energy transition objectives, many wind farms are installed to produce electricity by exploiting wind energy. The production of electricity through wind farms is widespread in the world. One of the problems complained about by people who live near wind farms is the noise emitted by the rotation of the blades. This work reports acoustic measurements inside homes near wind farms. The acoustic measurements were performed for different wind speeds and directions. Furthermore, to verify the effectiveness of the simulation method of the propagation of the noise generated by wind farms, the measured values of the sound levels were compared with those obtained from the numerical simulations using the provisions of the ISO 9613 standard, and the wind turbines are considered point-like sound sources. The comparison between the measurements carried out in the field with the theoretical simulations has highlighted that the calculated levels are lower than the measured ones, this means that in the acoustic impact forecasting procedures, the noise emitted by the wind farms is underestimated and is one of the reasons why the Wind farms in operation are a source of complaints from the populations living nearby.

2:00

4pNS5. Empowering residents of communities near wind farms to record and document wind farm noise through the use of a phone app. Heather L. Lai (Eng. Programs, State Univ. of NY at New Paltz, 208 Eng. Innovation Hub, SUNY New Paltz, New Paltz, NY 12561, laih@newpaltz.edu), Anne C. Balant (Commun. Disord., State Univ. of NY at New Paltz, New Paltz, NY), and Kimberly A. Riegel (Phys., Farmingdale State College, Farmingdale, NY)

The impact of disturbances caused by wind farm noise (WFN) on residents can be difficult to predict. In talking with advocates for residents living near wind farms, it has become clear that some residents are negatively impacted by WFN, but the specific features that cause the greatest impacts are not well understood. A few studies have been published correlating in-home recordings of WFN with annoyance ratings by occupants in nearby residences, and some local municipalities have recorded SPLs at residences (such as an upstate NY town appointed volunteer noise officer who measures sound pressure levels on the property of concerned residents), but the vast majority of residents' experiences go undocumented. In this talk, we will present a plan for using a cell phone app, "Auditive," originally developed for collecting users' responses to urban noise to provide residents living near wind farms with the opportunity to document their experiences with WFN while making simultaneous recordings. These recordings in conjunction with the user ratings will allow identification of qualitative aspects of WFN that cause the most annoyance from a large sample of individuals and may also provide a sense of agency for residents who are most negatively affected.

Invited Papers

2:15

4pNS6. Improvements to a mobile app for community noise and assessment data collection. Kimberly A. Riegel (Phys., Farmingdale State College, 2350 Broadhollow Rd., Farmingdale, NY 10803, riegelk@farmingdale.edu) and Jody Resko (Social Sci., Queensborough Community College, Bayside, NY)

A mobile phone application was developed to collection health history, real time noise levels and participants' perceptions. The goal of this application is to collect data that can directly link community response, noise levels and community health from a large number of participants. This will help identify where noise is causing issues related to physical and mental health on a community scale. In order to identify these effects at a community level large numbers of responses are critical. Ease of use is essential to ensure that participants remain willing to participate in the study and continue to submit samples. This needs to be balanced with the usefulness of the data collected by the public. Previously, a beta version of the app was deployed to evaluate the useability and effectiveness of data collection. Several issues with the data collection were identified and improvements to the app were implemented. Evaluation of the data and data visualization options will be presented using the most recent version of the application.

2:35

4pNS7. Relationship between intra-community noise tolerance distributions and fitted dose-response funtions. Richard D. Horonjeff (48 Blueberry Ln., Peterborough, NH 03458, rhoronjeff@comcast.net)

Increasingly, researchers are employing fitted exposure-response functions to not only characterize the relationship between sound level and prevalence of annoyance but also to compare the relative noise tolerance of individual communities. The sound level at which 50% of the population is annoyed has been advanced as a community noise tolerance metric. This paper combines prior research and recent simulations to reframe the discussion in terms of plausible distributions of individual (personal) noise tolerances within a given community; the integrals of those (density) distributions become the (cumulative) observed exposure-response relationships. Various

tolerance distributions are developed, their resulting cumulative exposure-response relationships calculated, and the ability of commonly-employed functional forms to represent those cumulative relationships presented. The results of the current investigation strongly suggest that data sets which include annoyance fractions of 50% have the greatest chance of accurately predicting the 50%-annoyed sound level. The further the data set lies from the 50%-annoyed point the more dependent an extrapolated estimation becomes on the fitted functional form accurately representing the true cumulative relationship. Errors of estimation are shown for a number of tolerance distributions, functional forms and experimental data ranges. Finally, some remedial recommendations are offered.

2:55–3:10 Break

3:10

4pNS8. A unified framework for creating soundscape perception indices based on the SSID Protocol. Andrew Mitchell (Inst. for Environ. Design and Eng., Univ. College London (UCL), Central House, 14 Upper Woburn, London WC1H 0NN, United Kingdom, andrew.mitchell.18@ucl.ac.uk), Francesco Aletta, Tin Oberman, and Jian Kang (Inst. for Environ. Design and Eng., Univ. College London (UCL), London, United Kingdom)

The soundscape approach provides a basis for considering the holistic perception of sound environments, in context. While steady advancements have been made in methods for assessment and analysis, a gap exists for comparing soundscapes and quantifying improvements in the multi-dimensional perception of a soundscape. To this end, there is a need for the creation of single value indices to compare soundscape quality which incorporate context, aural diversity, and specific design goals. Just as a variety of decibel-based indices have been developed for various purposes (e.g., LAeq, LCeq, L90, Lden, etc.), the soundscape approach requires the ability to create novel indices for different uses, but which share a common language and understanding. We therefore propose a unified framework for creating both bespoke and archetypal single index measures of soundscape perception based on the soundscape circumplex model, allowing for new indices to be defined in the future. The implementation of this framework is demonstrated through the creation of a public space typology-based index using survey data collected under the SSID Protocol. Indices developed under this framework can enable a broader and more efficient application of the soundscape approach in design, planning, and regulation.

Contributed Papers

3:30

4pNS9. Investigation of whistle noise impacting a residential subdivision. Gregory Dennis (Valcoustics Canada Ltd., 41 Corianne Ave., Whitby, ON L1M2H9, Canada, greg.pwdennis@gmail.com)

There have been complaints from residences within a residential subdivision in New Tecumseth, Ontario, of train noise from the Canadian Pacific Railway (CPR) line that passes directly east of the development. The complaints relate to sound levels experienced within the dwellings during train pass-bys, most notably from the use of whistles at the grade level crossing southeast of the site. Sound level measurements were completed to quantify the indoor and outdoor sound levels at several dwellings closest to the rail line. The measured sound levels were compared to predictions made using the Ministry of the Environment, Conservation and Parks (MECP) noise model Sound from Trains Environmental Analysis Method (STEAM). This investigation discusses shortfalls in STEAM, which resulted in the sound levels (from whistle noise) being under predicted at the dwellings. The study also compares the sound levels predicted using the Federal Railway Administration (FRA) horn noise model to those measured on site and predicted using STEAM.

3:45

4pNS10. Revised comprehensive new definition of noise. Daniel Fink (The Quiet Coalition, The Quiet Coalition, P.O. Box 533, Lincoln, MA 01733, DJFink@thequietcoalition.org)

A revised comprehensive new definition of noise is proposed. Noise: a) For living things, noise is unwanted and/orharmfulnd. b) In engineering and electronics, noise is any unwanted disturbance within a useful frequency band, such as undesired lectric waves in a transmission channel or device. c) In scientific measurements, noise is erraic, intermittent, or statistically random oscillation. The revised comphrehensive new definition builds on the Acoustical Society of America/American National Standards institute definition to include technical considerations, and acknowledges the harmful effect of noise on plants. It updates the noise definition presented at the 2019 Acoustical Society of America winter meeting, *noise is unwanted and/or harmful sound*. Unlike the standard definition, *noise is unwanted sound*, that new definition emphasized that unwanted sound is harmful, able to cause adverse auditory and non-auditory health effects, and that wanted sound can also caus auditory damage. The Proceedings of Meetings on Acoustics article based on that presentation has been cited 37 times. The prevvious new definition opens the 2021 American Public Health Association policy statement, *Noise as a Public Health Hazard*, was adopted for use by the International Commission on Biological Effects of Noise (ICBEN) in 2023, and added to the ICBEN Constitution.

4:00

4pNS11. Reduction of noise impact from HVAC equipment that affects students in an Ecuadorian university. Felix Ramon Silva Tumbaco, David Andres Lince Correa, Galo Durazno (Escuela Superior Politecnica del Litoral, Guayaquil, Ecuador), and Carlos Yoong (Wood PLC, 2020 Winston Park Dr #700, Oakville, ON L6H6X7, Canada, carlos.yoong@woodplc.com)

The Faculty of Mechanical Engineering in the Coastal Polytechnic School located in Guayaquil, Ecuador currently presents an issue associated to the high noise levels from HVAC equipment located in close vicinity of classrooms. During the design and installation of the air conditioning system, noise control was never considered. This paper explores the different solution options that are pratical in an academic context while also taking into account local products, costs and environment. By taking into account the local guidelines, a group of students conducted the project from the field work to the reporting. These students then presented this study as their Capstone Project.

4:15

4pNS12. Sound and noise exposure dosimetry at large-scale music events and festivals. Wannes Van Ransbeeck (Ghent Univ., Ghent, Belgium), Nele De Poortere, Marcel Kok (dBControl, Zwaag, Netherlands), and Sarah Verhulst (Ghent Univ., Technologiepark 126, Zwijnaarde 9052, Belgium, s.verhulst@ugent.be)

Regulations governing sound exposure at amplified-music events aim to protect the audience from hearing damage. However, current level monitoring practices often rely on a single measurement position that is considered representative of individual exposure, overlooking crucial variation. This study addresses this lacking insight by analyzing individual dosimeter-based sound-level exposure at music events and its relation to existing guidelines. Dosimeters secured to 42 individual participants (19 females, aged 18–25) recorded octave bands and A/C-weighted sound levels at six large-scale music events (+20.000 visitors). Individual exposure durations varied

between 4.4 and 22 hours. Comparisons of exposure were made against local, German and WHO regulations. At four out of six events, individual doses surpassed the WHO's 100 dBA recommendation, with most subjects also exceeding the ISO1999 occupational limit and the WHO's 100 dBA-for-4-hours threshold (16Pa2h). Equivalent exposures ranged between [85.2,104.5] dBA (LA,eq) and [97.1,119.6] dBC (LC,eq) among participants. Additionally, LC,peak values fluctuated between [133.6,143.5] dBC. These findings highlight a discrepancy between the fixed-location noise exposure monitoring, as per country-specific legislation, and the actual individual exposure experienced by event attendees. Our results can inspire safe-listening guidelines for such music events, or help in deciding which single position measure is most representative. Work supported by UGent BOF-IOP EarDiMon and dBControl

4:30–5:00 Panel Discussion

THURSDAY AFTERNOON, 16 MAY 2024 — ROOM 206, 1:00 P.M. TO 5:25 P.M.

Session 4pPA

Physical Acoustics, Engineering Acoustics and Structural Acoustics and Vibration: Novel Methods and Applications in Nondestructive Evaluation

Luz D. Sotelo, Cochair
Purdue University, 2550 Northwestern Ave, 1900D, West Lafayette, IN 47906

Matthew D. Guild, Cochair
Naval Research Lab, 4555 Overlook Ave SW, Acoustics Division, Code 7160, Washington, D.C. 20375

Joseph A. Turner, Cochair
Mechanical and Materials Engineering, University of Nebraska-Lincoln, University of Nebraska-Lincoln, Lincoln, NE 68588

Chair's Introduction—1:00

Contributed Papers

1:05

4pPA1. Measurement of acoustic nonlinearity and anelasticity in metals with phase-sensitive nonlinear reverberation spectroscopy and noncontacting electromagnetic-acoustic transduction. Ward Johnson (National Inst. of Standards and Technol., 325 Broadway, MS 647, Boulder, CO 80305, wjohnson@boulder.nist.gov)

Nonlinear reverberation spectroscopy (NRS), in each of its various implementations, focuses on characterization of resonant acoustic nonlinearity and anelasticity through measurements of time-dependent changes in resonant frequency and vibrational amplitude during free decay after acoustic excitation. A general advantage of NRS over stepped-frequency nonlinear resonance techniques is the relatively short duration of free decay and associated small temperature drift during data acquisition. A recent innovation in NRS employs noncontacting electromagnetic-acoustic transduction and phase-sensitive superheterodyne reception to provide additional advantages of eliminating contributions to nonlinearity and loss from contacting transduction, simplifying signal analysis, and improving signal-to-noise ratios. Measurements of resonant axial-shear modes in a 7075 Al cylinder demonstrate precision of fractional frequency shifts on the order of 0.1 ppm, exceeding by two orders of magnitude the greatest reported precision of measurements achieved with nonlinear resonant ultrasound spectroscopy (NRUS). The high precision and resolution of the technique enable sensing of differences in nonlinearity and anelasticity associated with residual porosity at industrially relevant levels of less than half a percent in commercially pure additively manufactured aluminum.

1:25

4pPA2. Developments in an acoustic resonance test for the detection of manufacturing anomalies in hydroelectric generator stator windings. Kevin Venne (Hydro-Québec, 1800 Bd Lionel-Boulet, Varennes, QC J3X 1S1, Canada, venne.kevin2@hydroquebec.com), Mathieu Kirouac, Hélène Provencher, Mélanie Lévesque, and Mathieu Soares (Hydro-Québec, Varennes, QC, Canada)

To meet the ever-increasing demand for electricity, Hydro-Québec (HQ) is seeking to simultaneously increase the power of its generating stations while improving its service quality. Thus, the company has tasked its

research institute to investigate innovate methods to meet the aforementioned goals. Of interest in the current study is the development of an acoustic resonance test (ART) to improve the quality control (QC) in the manufacturing of hydroelectric generator stator windings. Since HQ and its suppliers are investigating new fabrication methods for stator windings to meet the required timelines and increased power requirements, QC is required to ensure the service quality of its new hydroelectric generators. Typical manufacturing anomalies found in stator windings are delamination and air pockets between insulation layers. Such anomalies can result in an acceleration in the degradation of the winding insulation, which reduces the service quality of hydroelectric generators. To benchmark the ART method, the results of the suspected locations of the anomalies along the stator windings were compared with an acoustic camera and the locations were dissected and inspected under microscope for validation. Ten different stator windings were tested and two metrics (variations in both force and frequency responses) were found to indicate the location of delamination sites in the stator windings and corroborated with the result of both the acoustic camera and the dissections.

Invited Paper

1:40

4pPA3. Acoustoelastic characterization of aluminum plates using zero group velocity Lamb modes. Rosa E. Morales (Lawrence Livermore National Lab., 7000 East Ave. Livermore, CA 94550, morales31@llnl.gov), Niket Pathak (Mech. Eng., Univ. of Colorado, Boulder, Boulder, CO), Jordan Lum (Lawrence Livermore National Lab., Livermore, CA), Christopher M. Kube (Eng. Sci. and Mech., The Penn State Univ., University Park, PA), Todd Murray (Mech. Eng., Univ. of Colorado, Boulder, Boulder, CO), and David M. Stobbe (Lawrence Livermore National Lab., Livermore, CA)

Acoustoelasticity, a characteristic of material anharmonicity, gives rise to a link between wave propagation velocity and the stress state in materials. Ultrasonic techniques to monitor this coupling, particularly with high sensitivity and in a noncontact manner, can have widespread application both in the quantification of applied and residual stress and in the characterization of nonlinear material behavior through measurement of higher order elastic constants. Here, we use a laser ultrasonic technique to excite and detect zero group velocity (ZGV) Lamb wave resonances in aluminum plates under uniaxial loading. A laser line source is used to excite these resonances at different orientations with respect to the applied load and the signals are detected using an interferometer. The effects of stress and source orientation on ZGV resonance frequencies are validated using the theory of acoustoelastic Lamb wave propagation. In addition, a model-based inversion technique is used to extract Murnaghan's third-order elastic constants from measurements of the stress dependence of the first two ZGV modes generated parallel and perpendicular to the applied load. Laser generation and detection of ZGV resonances is shown to be an effective and powerful approach for the noncontact and nondestructive acoustoelastic characterization of elastic waveguides.

Contributed Papers

2:00

4pPA4. Ellipsometry study of surface acoustic waves for viscoelastic material characterization: Estimation of complex Lamé coefficients versus the frequency. Aziz Bouzzit (CY Cergy Paris, 5 mail Gay Lussac, Neuville sur Oise, Ile de France 95000, France, aziz.bouzzit@cyu.fr), Loïc Martinez, Andres Arciniegas mosquera, Stéphane Serfaty, and Nicolas Wilkie-chancellier (CY Cergy Paris, Neuville sur Oise, Ile de France, France)

Adequate regarding material characterization, surface acoustic waves (SAWs) have a unique elliptic polarization that can be distinguished by two parameters. The first parameter, the H/V value, is the ratio between the in-plane and out-of-plane components of the particle's motion while it undergoes the surface wave. The second parameter is the orientation angle θ of the elliptic motion, defined as the angle between the major axis of the ellipse and the horizontal axis. The present paper proposes a method for estimating the viscoelastic isotropic material properties from these two parameters. A complete characterization is done by identifying the variation of the complex Lamé coefficients versus the frequency. This is achieved by employing the Quaternion Fourier Transform (QFT) on the quantitative measurement of the polarization. The propagative characteristics of the SAW, represented by the complex wavenumber, are extracted using the Prony algorithm. The inverse problem is based on the theoretical models of the propagation of SAW on viscoelastic materials. The proposed method is implemented and tested on space-time signals extracted from numerical simulation and experimental setup.

2:15

4pPA5. Buried object localization by spectral analysis of surface wave reflections. David Baumann (Elec. & Comput. Eng., Lake Superior State Univ., Sault Ste Marie, MI), Emily Hagelthorn, Andrew Heiny (Mech. Eng., Lake Superior State Univ., Sault Ste Marie, MI), Robert Hildebrand (Mech. Eng., Lake Superior State Univ., 650 W Easterday Ave. Sault Ste Marie, MI 49783, rhildebrand@lssu.edu), Morgan Kelly, Haluk Kucuk (Mech. Eng., Lake Superior State Univ., Sault Ste Marie, MI), Dustin Mangone, Omar Nobani (Elec. & Comput. Eng., Lake Superior State Univ., Sault Ste Marie, MI), and Masoud Zarepoor (Mech. Eng., Lake Superior State Univ., Sault Ste Marie, MI)

A proposed method of buried object localization, based upon spectral analysis of surface wave reflections, is investigated numerically. This arises from a hypothesis to the effect that reflectivity would be maximized at some intermediate wavelength, shorter ones associated with disturbances too shallow to substantially excite the object as a secondary (reflecting) source, and longer ones with disturbances involving such volumes of earth in motion that the object become insignificant as a reflector. Discernment of that intermediate wavelength of maximum reflectivity might, thereby, provide an index of the object's depth. Horizontal distance from echo return time, and azimuth from phase array techniques, could complete a localization methodology. Presented, accordingly, are 2D FEA simulations, in which narrow-banded surface wave trains are excited by a point source on the surface of an elastic medium, a void provided at some horizontal distance and depth, the strength of reflected surface motions thereafter examined in relation to frequency. These bear out the hypothesis, finding maximal reflectivity for a surface wavelength about two-thirds of the object depth. Proposed is that this could serve as the basis for a horizontal stand-off detection method, especially for applications like landmines, for which it may be undesirable to scan from above.

Invited Papers

2:30

4pPA6. Development of a scanning acoustic microscopy-based structural prior for microtexture region characterization. Laura Homa (Univ. of Dayton Res. Inst., Dayton, OH), Tyler Lesthaeghe (Univ. of Dayton Res. Inst., Dayton, OH), Matthew Cherry (Air Force Res. Lab., Fairborn, OH), Erik Blasch (Air Force Res. Lab., 875 N. Randolph, Ste. 325, Arlington, VA 22203, erik.blasch.1@us.af.mil), and John Wertz (Air Force Res. Lab., Beavercreek, OH)

Nondestructive evaluation (NDE) plays a crucial role in ensuring aircraft availability. The current NDE paradigm often relies on mono-modal testing and signal-over-threshold criteria to provide robust defect or damage detection, not characterization. One example is found in the risk-based management of surface-breaking cracks in metal, where cracks of a given size can be detected by eddy current testing (ECT) with a calculable probability. Yet, there are cases where detection proves insufficient. Consider the case of microtexture regions (MTR) found in certain titanium alloys, which can increase the risk of cold dwell fatigue failure when found above a certain size and in specific orientations relative to the surrounding material. At present, the size and orientation of MTR cannot be characterized using only one NDE modality. In this work, a data fusion-based solution to MTR characterization is developed. First, two inspection methods—scanning acoustic microscopy (SAM) and ECT—are selected, where each method is individually capable of only partial characterization. Then, matching component analysis is used to develop a surrogate forward model relating MTR orientation to ECT output. This data is then inverted using boundaries provided from the SAM data as a structural prior.

2:50–3:10 Break

Contributed Papers

3:10

4pPA7. A novel ultrasonic technique for the inspection of a plate heat exchanger. Pooja Dubey (George W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., 2 Rue Marconi, Georgia Tech-Europe, Metz 57070, France, pooja.dubey@gatech.edu) and Nico Declercq (George W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., Metz, France)

Biofilm formation in process industries, water treatment devices, and drinking water pipe networks pose a significant risk to public health and cause a variety of operational issues. One of the major challenges is the inspection of biofilms in heat-exchanging devices such as plate heat exchangers. A plate heat exchanger (PHE) is integral to food industries, water treatment plants, and others. Conventional techniques for cleaning biofilm formation in these plates (foulant) include chemical cleaning, steam, and hydro-blasting. However, they are inefficient and labor-intensive because of limitations such as increased handling risk, over-cleaning, and corrosion. Thus, developing novel techniques for real-time monitoring of biofilm growth on these devices is critical for efficient working. Previous studies were restricted to the development of ultrasound-assisted heat exchangers to reduce the deposits in these plates. However, only a few studies have investigated ultrasound as a probable monitoring tool. Thus, this research aims to explore the nonlinear ultrasonic parameters using the second harmonic generation technique as a real-time tool for monitoring biofilms in PHE. The proposed research will help design more effective ultrasonic-assisted plate heat exchangers to achieve maximum heat transfer efficiency.

3:25

4pPA8. Thickness characterization of test specimens using frequency-modulated ultrasonic signal generated via. fluidic oscillator. Viswa R. Sunkavalli (Zerstörungsfreie Prüfmethoden für das Bauwesen, Bundesanstalt für Materialforschung und -prüfung, Unter den Eichen 87, Berlin, Berlin 12205, Germany, viswa.sunkavalli@bam.de), Christoph Strangfeld, and Stefan Maack (Zerstörungsfreie Prüfmethoden für das Bauwesen, Bundesanstalt für Materialforschung und -prüfung, Berlin, Germany)

Traditional ultrasonic non-destructive testing methods in civil engineering require the use of coupling agents, leading to prolonged and labor-intensive measurement procedures along with the risk of surface damage. To alleviate these concerns, air-coupled ultrasonic actuation has been introduced as an effective alternative. However, the power loss due to impedance mismatchs at the interfaces remains an important limitation. To mitigate the power losses at the interfaces while characterizing the specimen thickness, we employ the fluidic oscillator as an ultrasonic source, wherein air acts as both the operating and coupling medium. The fluidic oscillator generates signals through self-sustained oscillations of the exiting air jet under continuous pressurized air supply, and therefore, eliminates the need for intricate design and manufacturing processes. Our preliminary investigations highlighted the dependence of the dominant spectral characteristics of a given fluidic oscillator on the mass flow rate of input air. Leveraging this observation, frequency-modulated chirp signals are produced by rapidly varying the flow rate, enhancing the signal-to-noise ratio for a reliable assessment of material characteristics. To demonstrate the applicability of the air-coupled, frequency-modulated ultrasonic signal generated using fluidic oscillators, in this study, an exploratory thickness characterization of concrete and polymer specimens using both laser vibrometer and piezoelectric sensors as receivers was performed.

3:40

4pPA9. Ultrasonic scattering behavior in austenitic NiTi with varying grain size and precipitate characteristics. Olivia J. Cook (Eng. Sci. and Mech., Penn State Univ., 212 Earth Eng. Sci., University Park, PA 16801, ojc3@psu.edu), Foster K. Feni, Mique A. Gonzales, Reginald F. Hamilton, and Andrea P. Arguelles (Eng. Sci. and Mech., Penn State Univ., State College, PA)

Shape memory alloys like NiTi exhibit high strain recovery due to a reversible martensitic transformation when cooled or stressed, enabling actuation and biomedical applications. The characteristics of the stress-induced martensitic transformation in Ni-rich NiTi depend on the heat treatment. For example, aging induces the precipitation of Ni_4Ti_3 and lowers the critical stress needed for martensitic transformation. However, comprehensive characterization of these nanoscale precipitates is challenging as only small areas can be inspected efficiently. Ultrasonic immersion testing may assist in developing linkages between structure and transformation behavior as ultrasonic scattering is sensitive to the microstructure and easily mapped on the bulk scale. To this end, the ability of scattering behavior to discern between different precipitation characteristics must be probed to establish the sensitivity of ultrasonic parameters to changes in precipitation. This work explores the dependence of wave speed, attenuation, and backscattered energy on varying annealing and aging heat treatments of Ni-rich NiTi. The annealing treatments result in a gradient of grain sizes, while the aging treatments result in nucleation and growth of precipitate phases with increasing hold time. Variations in the ultrasonic parameters elucidate the sensitivity to changes in the microstructure, which are more quantitatively explored through existing analytical scattering models.

3:55

4pPA10. Ultrasonic evaluation of additively manufactured components with embedded structures. Harshith Kumar Adepu (Mech. Eng., Purdue Univ., 500 Central Dr., B36, West Lafayette, IN 47906, adepu@purdue.edu), Meher Mirza, Jacey Birkenmeyer, and Luz D. Sotelo (Mech. Eng., Purdue Univ., West Lafayette, IN)

In recent years, the use of ultrasonic nondestructive evaluation (NDE) for *in situ* and *ex situ* assessment of Additive Manufacturing (AM) parts has garnered interest due to its proficiency in defect detection and material characterization. This study focuses on evaluating the ability to resolve internal structures created by AM and hybrid AM using ultrasonic NDE as well as assessing the impact of microstructure heterogeneity. For this purpose, first the parameter space for laser powder bed fusion (LPBF) of SS316L is mapped in a Lumex Avance 25 system. Based on this parameter mapping, optimal process parameters are identified and sets of process parameters are selected to vary the thermal history of the samples, while minimizing the influence of porosity. Two groups of samples are printed with these parameter sets: a solid group of samples, and a group with internal structures, such that each internal structure sample has an equivalent solid sample. Measurements of ultrasonic velocity, attenuation, and backscatter amplitude are collected and compared against the intended geometry to assess the impact of non-optimal processing on the ability to resolve internal structures with ultrasound. The viability and limitations of using ultrasound to assess internal features created with AM are discussed.

4:10

4pPA11. Quantitative ultrasonic characterization of Ti-6Al-4V components with surface roughness. Sydney N. Assalita (Eng. Sci. and Mech., Penn State Univ., 212 Earth and Eng. Sci. Bldg., University Park, PA 16802, sna5228@psu.edu), Olivia J. Cook, and Andrea P. Arguelles (Eng. Sci. and Mech., Penn State Univ., State College, PA)

Ultrasonic immersion testing enables nondestructive characterization of material microstructure through quantitative linkages between wave propagation parameters and features of interest. However, accurate measurements of metrics such as wave speed and attenuation generally require smooth test samples to ensure the reflected wave packets are not distorted. Thus, rough surfaces are mechanically polished before ultrasonic testing, which compromises the nondestructive nature of the measurement, distorting the sample geometry and imparting local damage. This challenge is particularly salient with the advent of manufacturing processes, such as metal additive manufacturing, that result in rough samples that must be characterized for heterogeneities in their as-manufactured state. This work examines the influence of surface roughness on longitudinal wave speed and attenuation in wrought and additively manufactured Ti-6Al-4V specimens. Samples are measured in an ultrasonic immersion setup before and after mechanical polishing using varying transducer frequencies and focal profiles. The resulting data informs the error generated in ultrasonic measurements as a function of roughness parameters, thereby informing the sample preparation needed for desired precision levels. Lastly, new measurement protocols are explored to correct for the presence of surface irregularities in a point-by-point fashion. This work was supported by the Robert W. Young Award for Undergraduate Student Research.

4:25

4pPA12. The impact of the summer undergraduate research or internship experience in acoustics program on my knowledge and passion for material science and acoustics. Aja Leatherwood (case western reserve Univ., 10900 Euclid Ave., Cleveland, OH 44106, axl862@case.edu), Haley N. Jones, and Andrea P. Arguelles (Mater. Sci. and Eng., Penn State Univ., State College, PA)

Throughout an enriching 10-week experience with the Acoustical Society of America's Summer Undergraduate Research Experience in Acoustics program (SUREIA), I had the privilege of contributing to the innovative research at Penn State's Arguelles Lab, focusing on ultrasound as a tool for characterizing material properties and defects. Despite my background in communication science, I immersed myself in material science through the lab. I collaborated with graduate students to learn immersion ultrasound testing on bulk ceramic materials and processed data using MATLAB. At the end of the opportunity, I presented my findings at Penn State University and the SURIEA Program Open House. Beyond the academic pursuits, the true richness of this experience lies in the meaningful connections cultivated within my SUREIA cohort and lab members. These connections have not only enhanced my understanding of material science but have also provided me with a network of like-minded individuals whom I can collaborate with in the future. Additionally, the Arguelles Lab has provided me with invaluable mentorship and guidance, enabling me to enhance my research skills and broaden my knowledge in the unfamiliar yet fascinating field of material science.

4:40

4pPA13. Abstract withdrawn.

4:55

4pPA14. Topological acoustic sensing using the geometric phase. Keith Runge (New Frontiers of Sound (NewFoS) Ctr., Univ. of Arizona, 1235 E. James E. Rogers Way, Dept. of Mater. Sci. and Eng. University of Arizona, Tucson, AZ 85721, krunge@arizona.edu) and Pierre A. Deymier (New Frontiers of Sound (NewFoS) Ctr., Univ. of Arizona, Tucson, AZ)

We introduce a method, topological acoustic sensing, which exploits changes in the geometric phase of acoustic waves to sense defects in some structure or environment. This method is illustrated in two cases of perturbations taking the form of (1) a mass defect located on an array of coupled acoustic waveguides, and (2) a small subwavelength object on a flat surface submerged under water. We represent the state of the acoustic field in the unperturbed and perturbed cases as multidimensional vectors. The change in geometric phase is obtained by calculating the angle between those vectors. This angle represents a rotation of the state vector of the wave due to scattering by the perturbation. By exploiting sharp topological features spanned by the acoustic field multidimensional state vector, we show that this geometric phase sensing modality can have higher sensitivity than magnitude-based sensing approaches.

5:10

4pPA15. Long-term acoustic emission monitoring of a new alkali-activated material for sealing structures in nuclear waste repositories. Anna M. Sklodowska (8.2 Non-destructive Testing Methods for Civil Eng., Bundesanstalt für Materialforschung und -prüfung (BAM), Unter den Eichen 87, Berlin 12205, Germany, anna.sklodowska@bam.de), Vera Lay (8.2 Non-destructive Testing Methods for Civil Eng., Bundesanstalt für Materialforschung und -prüfung (BAM), Berlin, Germany), Franziska Baensch (Deutsches Institut für Normung (DIN), Berlin, Germany), Ernst Niederleithinger (8.2 Non-destructive Testing Methods for Civil Eng., Bundesanstalt für Materialforschung und -prüfung (BAM), Berlin, Germany), and Hans-Carsten Kühne (7.4 Technol. of Construction Mater., Bundesanstalt für Materialforschung und -prüfung (BAM), Berlin, Germany)

The crucial part of nuclear waste storage is the construction of sealing structures made of reliable, well-understood, and safe materials. Within the SealWasteSafe project, we compared the performance of an innovative alkali-activated material (AAM) and standard salt concrete (SC), as potential materials for sealing structures for nuclear waste repositories. Two 340-liter-cubic specimens were studied for up to $\sim$250 days by a multisensory monitoring setup. Specifically, the long-term acoustic emission monitoring aimed to analyze the development of microstructural changes within materials. The monitoring analysis showed fewer acoustic emission events in AAM compared to SC in the first 61 days. After approximately two months of monitoring, the number of AE events in AAM significantly exceeded the number of events in SC. The analysis showed, however, that the increased AE activity was mainly caused by the surface effects of the AAM material and not by the formation of cracks within the material. This contribution presents the use of acoustic emission analysis, both in the time and frequency domains, for monitoring and characterization of materials with potential use as engineering barriers for nuclear waste repositories.

Session 4pPP

Psychological and Physiological Acoustics: Psychological and Physiological Acoustics Best Student Poster Award Session

Z. Ellen Peng, Cochair
Boys Town National Research Hospital, 555 North 30th Street, Omaha, NE 68131

Daniel R. Guest, Cochair
Department of Biomedical Engineering, University of Rochester, 601 Elmwood Ave, MC 5-6483, Rochester, NY 14620

All posters will be on display from 1:00 p.m. to 5:00 p.m. Authors of odd-numbered abstracts will be at their posters from 1:00 p.m. to 3:00 p.m. and authors of even-numbered abstracts will be at their posters from 3:00 p.m. to 5:00 p.m.

Contributed Papers

4pPP1. An exploratory investigation of acoustic features underlying arousal and valence perception of vocalizations from non-speaking individuals. Simon M. Radhakrishnan (Elec. Eng. & Comput. Sci., Massachusetts Inst. of Technol., 77 Massachusetts Ave., Cambridge, MA 02139, rdkn@mit.edu), Amanda M. O'Brien (Speech and Hearing Bioscience and Technol., Harvard Univ., Cambridge, MA), Thomas Quatieri (MIT Lincoln Lab., Massachusetts Inst. of Technol., Lexington, MA), and Kristina T. Johnson (Elec. & Comput. Eng., Northeastern Univ., Boston, MA)

Emotion perception of vocalizations, especially from individuals with no or few spoken words, remains an underexplored topic in acoustical research. The aim of this exploratory study was to identify acoustic features within non-speech vocalizations correlated with perceived arousal and valence. 364 vocalizations were selected from the open-access ReCANVo dataset, comprising non-speech communicative sounds by non-speaking individuals with autism and neurodevelopmental disorders. 108 listeners independently rated each vocalization for arousal and valence on a 5-point Likert scale (78624 total ratings). We then used random forest and elastic net regression techniques to determine correlations between acoustic-based features and perceived affect. Using 4 published (e.g., GeMAPs) and 2 custom feature sets (dim 12, 28) encompassing duration, intensity, pitch, formant, and spectral centroid features, we evaluated each model using R-squared and mean-squared error. Our 12-feature custom elastic net model achieved the highest performance for predicting arousal (R-sq = 0.763, MSE = 0.075), with intensity features weighted highest, comparable to other sets. Our 28-feature random forest model, which included vocal quality features, significantly outperformed other models in valence prediction (R-sq = 0.491; MSE = 0.139, $p < 0.001$). While further validation on a held-out test set is essential, these exploratory results expand our understanding of affective acoustic features of non-speech sounds from individuals with neurodevelopmental disorders.

4pPP2. Effects of musical training experience and sentence difficulty on noisy speech recognition in second language learners. Shuhang Chen (Appl. Psych. Programme, Dept. of Life Sci., BNU-HKBU United Int. College, q030016008@mail.uic.edu.cn, Zhuhai, Guangdong 519000, China, q030016008@mail.uic.edu.cn), Siheng Li (Appl. Psych. Programme, Dept. of Life Sci., BNU-HKBU United Int. College, Guangzhou, China), and Yu Li (Appl. Psych. Programme, Dept. of Life Sci., BNU-HKBU United Int. College, Zhuhai, Guangdong, China)

Correctly recognizing second-language speech in a noisy environment is a difficult task for second-language learners. Early works have indicated the beneficial effects of musical training in native language acquisition and development. However, it remains unknown how musical training experiences influence second-language speech-in-noise (SIN) recognition in second-language learners. To examine this question, we recruited 45 right-handed young adults who speak English as their second language and showed different musical training experiences over the past ten years, and employed an English SIN test in which two factors were manipulated, signal-to-noise ratio level (SNR; quiet, + 5 dB, −5 dB) and sentence difficulty (easy, hard). The percentage of words correctly identified from sentences was used in the data analyses. The results revealed a significant interaction between musical training experience, SNR level, and sentence difficulty. Further analyses revealed that compared with the short musical training group, the long musical training group exhibited larger differences between easy and hard sentences in the quiet condition. Overall, these results provide new evidence for the benefits of music training experiences for SIN recognition in the context of second language learning. [Work supported by the Humanities and Social Sciences Foundation of Ministry of Education of China 20YJCZH079.]

4pPP3. Adults show own-age advantage in inter-talker similarity ratings. Yu Li (Dept. of Life Sci., BNU-HKBU United Int. College, 2000 Jintong Rd., Tangjiawan, Zhuhai, Guangdong 519087, China, yuli@uic.edu.cn), Tongyu Qiu, Junze Li (Dept. of Life Sci., BNU-HKBU United Int. College, Zhuhai, Guangdong, China), Sabrina Yanan Jiang (Macau Univ. of Sci. and Technol., Macao, China), and Linjun Zhang (Peking Univ., Beijing, China)

Own-age advantage in talker identification has been reported in adults (Cooper *et al.*, 2020; Creel & Jimenez, 2012). Whether the own-age advantage can also be observed in an inter-talker similarity rating task is still an open question. In the current study, we first reversed sentences (completely unintelligible but have most sound structures preserved) recorded from 8 adults and 8 children. Then we asked adults to rate the inter-talker similarity of pairs of these reversed sentences by using a visual analog scale (0-10 with 0 denoting "totally different" and 10 "totally same; see Fleming *et al.*, 2014). For each participant, we calculated the average inter-talker similarity of pairs of reversed sentences separately for child and adult talkers. A lower inter-talker similarity reflects higher sensitivity to the differences in voices between talkers. The results showed that the inter-talker similarity was significantly higher for child talkers than adult talkers, suggesting that adults are more sensitive to the differences in voices between adult talkers. The finding demonstrates the own-age advantage in adults in a similarity rating task. The current study extended our existing knowledge of talker processing. [Work supported by the Humanities and Social Sciences Foundation of Ministry of Education of China 20YJCZH079.]

4pPP4. Neurophysiological biomarkers of speech-in-competition performance under different conditions of attentional load. Erick Correa-Medina (Lab. of Cognit. and Clinical Neurophysiology, National Inst. of Neurology and Neurosurgery, Circuito Ciudad Universitaria Avenida, C.U., Mexico City 04510, Mexico, neurick.psi@gmail.com), Rodolfo Solís-Vivanco (Lab. of Cognit. and Clinical Neurophysiology, National Inst. of Neurology and Neurosurgery, Mexico, Mexico), Tess Koerner, Frederick J. Gallun (Oregon Hearing Res. Ctr., Oregon Health and Sci. Univ., Portland, OR), Aaron Seitz (Ctr. for Cognit. and Brain Health, Northeastern Univ., Riverside, CA), and Sebastian Lelo de Larrea-Mancera (Ctr. for Cognit. and Brain Health, Northeastern Univ., Boston, MA)

A body of research has identified a complex integration of perceptual and cognitive processes that can modulate comprehension of speech-in-competition (SIC). However, it is not well understood how the neurophysiological correlates associated with SIC performance vary under different conditions of attentional load. Here, we analyzed the neurophysiological correlates of a SIC task using variations in masker type (spatialization, gender, language), target-to-masker ratios (TMRs), and presence of a simultaneous working memory (WM) task to manipulate attentional load during SIC performance. We used a counter-balanced block design to record EEG in each condition in 15 undergraduate young adults without hearing difficulties. SIC was assessed using the spatial release from masking task in the PART digital application. Behavioral results indicated significant differences in accuracy depending on TMR across types of maskers. We did not observe behavioral differences in SIC performance associated with the inclusion of a parallel WM task. Quantitative EEG showed a series of potential EEG biomarkers that can complement behavioral measurement in discriminating different attentional load conditions of SIC performance. These post hoc observations are discussed in relation to the extant literature and will be further investigated in future work.

4pPP5. Enhanced adult voice recognition ability in children. Tongyu Qiu (Life Sci., BNU-HKBU United Int. College, Zhuhai, Guangdong, China, 2642082558@qq.com) and Yu Li (Life Sci., BNU-HKBU United Int. College, Zhuhai, Guangdong, China)

Language acquisition during infancy and childhood is much influenced by various factors including social interactions with others. Previous works have indicated that children likely benefit more from interactions with people showing advanced language skills such as adults rather than peers who show less or inferior language skills. These benefits could be reflected by better recognition of adult voices containing rich linguistic information and conveying advanced language skills. We therefore hypothesized that children recognize adult voices better compared to child (peer) voices. To test this hypothesis, we used a voice recognition task to assess children's ability to recognize adult voices and child voices in a group of 8–9 years old children. The results showed children exhibited superior recognition for adult compared to child voices, which is in line with our hypothesis. The finding demonstrates the importance of interacting with adults who show advanced language skills in language acquisition from the perspective of voice recognition. We discussed the result by introducing the own-age advantage in voice recognition observed in adults (Cooper *et al.*, 2020) and provided implications for better facilitating child language acquisition. [Work supported by Guangdong Basic and Applied Basic Research Foundation 2022A1515110748 and UICstartup R72021207.]

4pPP6. Hearing of warning sounds under the settings of railway running noise considering age-related hearing loss. Kei Hoshino (Ergonomics Lab., Railway Tech. Res. Inst., hoshino.kei.10@rtri.or.jp, k.hoshino@ruri.waseda.jp, Tokyo 1858540, Japan, k.hoshino@ruri.waseda.jp), Ayako Suzuki (Ergonomics Lab., Railway Tech. Res. Inst., Tokyo, Japan), and Yasuhiro Oikawa (Intermedia Art and Sci., Waseda Univ., Tokyo, Japan)

Railroad drivers may become drowsy because of their irregular work schedules, such as late nights and early mornings. Therefore, in our research, we developed a system that detects driver drowsiness using images based on facial expressions and issues a warning sound to assist drivers during driving. However, it is necessary to function warning sounds even in loud running noise areas or various running noise qualities to make the system more practical. Moreover, it is necessary to consider the effect of a driver's age-related hearing loss on the setting of the warning sound volume in noisy environments, even for those who meet the hearing criteria as drivers. This is because the number of elderly drivers in their 60 s will increase in the future. Therefore, we investigated the appropriate volume of the warning sound under various running noise types and levels. Then, we compared it between groups of people in their early 20 s and 60 s regarding their hearing ability and the way they hear warning sounds under running noise. The results provide new insight into the relation between hearing ability and the ability to hear warning sounds under driving noise.

4pPP7. Localizing sounds revolving at very high velocities: An auditory wagon-wheel effect. Noa Kemp (Dept. of Physiol./School of Information Studies, McGill Univ./Ctr. for Interdisciplinary Res. in Music Media and Technol., 3460 Mc Tavish St., Montreal, QC H3A 0E6, Canada, noa.kemp@mail.mcgill.ca), Ulysse Lefeuvre, Cynthia Tarlao, and Catherine Guastavino (School of Information Studies, McGill Univ./Ctr. for Interdisciplinary Res. in Music Media and Technol., Montreal, QC, Canada)

Localizing sound sources as they move around us is a critical function of the auditory system. Yet most research focuses on static sound sources or sources moving at slow velocities. The present work explores circular trajectories at very high velocities well above the velocity at which we lose the sense of direction ($\sim$2.5 rot/s with white noise). As the number of rotations per second approaches the fundamental frequency of the spinning sound, a sense of direction re-emerges. This creates what has been described informally as the auditory equivalent to the wagon-wheel effect: the sound appears to move in one direction when the velocity is below the fundamental frequency, and it appears to move in the opposite direction when the velocity is above the fundamental frequency. We report on two experiments testing this effect with a 200-Hz complex sound using adaptive VBAP spatialization on a 16-loudspeaker array. Experiment 1 ($N = 15$) confirmed that participants perceived opposite directions when the velocity was below or above the fundamental frequency. Experiment 2 ($N = 7$) explores the relationship between this effect at very high velocities and the ability to track sound at slower velocities. Preliminary findings suggest some overlap between localization processes at slow and high velocities.

4pPP8. Effect of subsequent context on the real-time interpretation of ambiguous target words in spectrally degraded speech. Anna R. Tinnemore (Neurosci. and Cognit. Sci. Program, Univ. of Maryland, 0100 LeFrak Hall, College Park, MD 20742, annat@umd.edu), Sandra Gordon-Salant, and Matthew J. Goupell (Hearing and Speech Sci., Univ. of Maryland, College Park, MD)

Subsequent context in a sentence can be used to revise the interpretation of a word that was not clearly heard, a frequent occurrence in degraded speech. One strategy for avoiding the cognitive cost associated with revising an initial interpretation is to delay commitment to an interpretation until the end of the sentence, sometimes called a "wait-and-see" strategy. Listening with a significant hearing loss (i.e., constantly hearing degraded speech), may prompt a listener to use this strategy even in relatively easy listening conditions. This study used an eye-tracker to record gaze patterns during a phoneme classification task of ambiguous target words in two levels of background noise ($+5$ and -5 dB SNR). Participants either used cochlear implants (CIs) or were presented 8-channel noise vocoded speech. The target words were at the beginning of sentences; the subsequent context cued one target word interpretation (based on phoneme classification) or the other. We hypothesized that listeners who use CIs will demonstrate a delaying strategy at both levels of background noise, while listeners who have acoustic hearing will only demonstrate a delaying strategy at the highest level of background noise. These findings have implications for rehabilitative strategies to improve communication outcomes in adults with CIs.

4pPP9. Profile analysis in the inferior colliculus: physiology and modeling studies. Swapna Agarwalla (Biomedical Eng. Dept., Univ. of Rochester, University of Rochester, Rochester, NY 14642, swapwiz16@gmail.com), Daniel R. Guest, and Laurel H. Carney (Biomedical Eng., Univ. of Rochester, Rochester, NY)

Profile-analysis is a strategy for assessing sensitivity to differences in spectral shape [Green, Oxford University Press, 1988]. Despite extensive behavioral data, the neuronal mechanisms underlying profile-analysis remain elusive. To bridge this gap, extracellular recordings were made in awake rabbit inferior colliculus (IC) and simulated using a computation model. The standard stimulus was a log-spaced, n-component, zero-phase, equal-amplitude complex tone, presented diotically. The central component was incremented in the target stimulus. The discharge rate as a function of characteristic frequency (CF) was inferred by shifting the stimulus spectrum past the CF of each neuron. When the increment frequency was near CF, IC neurons had rates that decreased as the increment increased, contradicting a simple energy-based code. In some cases, the differences between rates for target and standard were largest for stimuli with component spacing that yielded the lowest thresholds in listeners. Responses of IC neurons that were excited by amplitude-modulated stimuli were consistent with model predictions for profile-analysis stimuli [Maxwell *et al.*, *JASA* **147**, 3523 (2020), Guest *et al.*, b*ioRxiv* (2023)]; however, the models did not explain responses of other types of IC neurons. Potential factors contributing to this discrepancy, such as off-CF inhibition and chirp selectivity, will be further explored. Support: NIHR01-DC010813

4pPP10. Neural mechanisms of spatial auditory attention with magnified interaural level difference cues. Benjamin N. Richardson (Neurosci. Inst., Carnegie Mellon Univ., 5702 Darlington Rd., Apartment 3, Pittsburgh, PA 15217, bnrichar@andrew.cmu.edu), Jana Kainerstorfer (Neurosci. Inst., Carnegie Mellon Univ., Pittsburgh, PA), Barbara Shinn-Cunningham (Carnegie Mellon Univ., Pittsburgh, PA), and Christopher A. Brown (Commun. Sci. and Disord., Univ. of Pittsburgh, Pittsburgh, PA)

Bilateral cochlear implant users struggle in spatial release from masking (SRM) tasks, likely due to restricted access to interaural time difference (ITD) cues. Instead, they must rely on interaural level difference (ILD) cues; however, our previous behavioral experiments suggest that magnification of ILDs can facilitate SRM. Here, we probed the neural mechanisms underlying the benefit of magnified ILDs. We tested 18 normal-hearing subjects in an anechoic chamber. Listeners heard target and masker sequences of object and color words from opposite (left and right) quarterfields and were asked to detect color words in the target stream. Both streams were spatialized using either 50 μS ITD (ITD50), 500 μS ITD (ITD500), a broadband 10 dB ILD (ILD10), or the largest naturally occurring frequency-specific ILD (70 degrees; ILD70n). We recorded task-elicited hemodynamic responses in dorsolateral prefrontal cortex (DLPFC) using functional Near-Infrared Spectroscopy. Subjects performed best in the ITD500 and ILD10 conditions. Hemodynamic response magnitudes were smaller for ITD50 than for all other conditions, consistent with frontal activity increasing when perceptual segregation is possible and spatial attention can be deployed successfully. These data show that magnified ILD cues enhance SRM and that the benefit ILDs confer arises because listeners can engage cognitive attentional processes.

4pPP11. Performance on a spatially selective auditory attention listening task for sources in the horizontal or coronal planes. Grace E. Otto (Neurosci. Graduate Program, Western Univ., London, ON, Canada) and Ewan A. Macpherson (National Ctr. for Audiol., Western Univ., 1201 Western Rd., EC 2262, London, ON N6G 1H1, Canada, ewan.macpherson@nca.uwo.ca)

Listeners can use spatially selective auditory attention (SSAA) to focus on one talker in a complex acoustic scene. SSAA has been primarily investigated with targets and distractors arrayed in the frontal horizontal plane. In this study we compared normally hearing human listeners' performance on static and dynamic SSAA tasks in a frontal target/distractor configuration to performance with sources arrayed in either the rear horizontal plane, the overhead coronal plane, or the "underhead" coronal plane, all of which provide similar binaural difference cues with which to differentiate target and distractor locations. To achieve the coronal plane configurations, the listener's head was tipped forward to align the normally vertical axis with the horizontal plane. In the SSAA task, listeners attempted to report a 4-digit sequence of spoken digits from the target location while ignoring two simultaneous equal-intensity sequences spoken by the same talker presented from flanking loudspeakers separated by $\pm 22.5^\circ$ from the target. Listeners either held their head still (static conditions) with the front (horizontal configurations) or top (coronal configurations) of the head oriented toward 0 azimuth, or were oscillated passively at $\sim$0.14 Hz with an amplitude of $\pm$45 degrees about the vertical axis (dynamic conditions).

4pPP12. Exploring the asymmetry in perception and production of mandarin tones: effects of high variability phonetic training with visual animation on native Thai speakers. Yilin Xiang (School of Foreign Studies, Xi'an Jiaotong Univ., No. 28 Xianning Rd. (W), Xi'an, Shaanxi, 710049, P. R. China, xiangyl@stu.xjtu.edu.cn), Bing Cheng (School of Foreign Studies, Xi'an Jiaotong Univ., Xi'an, Shaanxi, China), and Xiaojuan Zhang (Xi'an Jiaotong Univ., Xi'an, Shaanxi, China)

This study investigated the impact of a web-based high variability phonetic training (HVPT) program integrated with pitch contour visualization on Mandarin tone acquisition by native Thai speakers. Through a pre-test and post-test design, involving tasks of word identification and recordings, we examined the progress in both perception and production of four Mandarin tones in 20 participants. Our findings reveal a notable asymmetry in progress: While production accuracy improved significantly across most tones, as evidenced by higher ratings from Mandarin native speakers, perceptual accuracy did not show a parallel enhancement, except for Tone 1. Further analysis confirmed significant variability in training effectiveness across different tones. These findings underscore the pivotal role of perception-oriented training in enhancing tonal production for native speakers of tonal languages. Moreover, the differential improvements across four tones highlight the need for tailored training approaches in both perception and production. Key words: High variability phonetic training (HVPT), pitch contour visualization, Mandarin tone acquisition, perception and production asymmetry.

4pPP13. Auditory brightness in room reverberation. Chad Bullard (Indiana Univ., 2631 East Discovery Parkway, Bloomington, IN 47408, charbull@iu.edu) and Jennifer Lentz (Indiana Univ., Bloomington, IN)

This work examines the impact of room reverberation on the auditory perceptual brightness of sounds. We hypothesized that room size would impact the perception of brightness, with smaller rooms (less reverberant) leading to sounds being perceived as more bright than larger rooms. We generated single-note instrument sounds from five different instruments using Logic Pro which were reverberated in three different-sized simulated rooms. Reverberant stimuli were presented in sequential room-size pairs (small-medium, small-large, medium-large) in a randomized order. Participants (15 musicians) were asked to rate the brightness of the second sound as compared with the first using a Likert-style scale, and the stimuli were blocked by instrument such that in each block a participant would only hear room-size comparisons of one instrument. Results were analyzed using an ordered-probit regression of the brightness ratings against the spectral centroid of the stimuli. We found no clear effect of room size. However, brightness ratings were weakly correlated with the average spectral centroid as shown in previous literature, and a stronger correlation was found with the steady-state portion of the sound. This suggests room size has little effect on brightness. Future work should examine the effect of different segments (e.g., attack, steady-state) on perceived brightness.

4pPP14. Human perception of impact sounds suggests auditory intuitive physics. Vinayak Agarwal (Mech. Eng., MIT, 77 Massachusetts Ave., 46-4065, Cambridge, MA 02139, vinayaka@mit.edu), James Traer (Psychol. and Brain Sci., Univ. of Iowa, Iowa City, IA), and Joshua H. McDermott (MIT, Cambridge, MA)

Upon hearing objects collide, humans can estimate many of the underlying physical attributes, such as the objects' material and mass. Although the physics of sound generation are well established, the inverse problem that listeners must solve – of inferring physical parameters from sound – remains poorly understood. In this work, we show that humans leverage an understanding of acoustical physics to constrain their perceptual inferences, allowing them to disambiguate multiple object properties from a single impact sound. We derived a linear generative model of impact sounds, combining theoretical acoustics with empirically measured statistics of object resonances. We used an analysis-by-synthesis algorithm to infer mode parameters from recorded object impulse responses. We then fit distributions to these parameters, from which object impulse responses could be sampled. Perceptual experiments demonstrated that humans could judge material and mass from sound alone, even when both of the underlying objects varied. However, performance with synthetic sounds was impaired if the simulated physical regularities were altered to be unnatural. The results suggest that listeners use internal physical models to separate the acoustic contributions of the objects that interact to create sound.

4pPP15. Investigating the dynamic relationship between mandarin tone perception and production: A study on native thai speakers through perceptual training. Xujia Li (English Dept. & Lang. and Cognit. Neurosci. Lab, Xi'an Jiaotong Univ., No. 28 Xianning Rd. (W), Xi'an, Shaanxi 710049, China, lixujia@stu.xjtu.edu.cn), Kangzhi Liao, and Bing Cheng (English Dept. & Lang. and Cognit. Neurosci. Lab, Xi'an Jiaotong Univ., Xi'an, Shaanxi, China)

This study aims to explore the correlation between perception and production of four Mandarin tones among Thai learners of Chinese before and after one-week intensive perceptual training. Existing research has provided inconsistent findings regarding this correlation, and limited investigations have examined changes during the learning process. The experiment involved twenty Thai speakers who completed a self-developed perceptual test program comprising 120 natural speech stimuli (30 monosyllabic words for each tone). The participants' recordings of these 120 words were presented to four native Chinese speakers in random order, who were required to identify the tone for each word. Before training, a strong and positive correlation between Mandarin tone perception and production was found ($r = 0.89$, $p < 0.001$). However, this correlation significantly weakened after the training, indicating a complex relationship between perception and production in speech learning for second language learners. Further analysis revealed that the perceptual training led to imbalanced development in perception and production for the four tones. Specifically, Tone 2 and Tone 3 did not exhibit significant correlation after training, as production accuracy improved much faster than perception. These findings also have implications for instructional approaches in teaching Mandarin tones. Keywords: Mandarin tone, speech learning, perception-production links

4pPP16. Abstract withdrawn.

4pPP17. Contributions of auditory processing and cognition to the development of frequency discrimination performance during adolescence. Jordin T. Benedict (Speech Pathol. and Audiol., Kent State Univ., 1325 Theatre Dr., Kent, OH 44240, jbened11@kent.edu), Serena A. Sereki, Bruna S. Mussoi, and Julia J. Huyck (Speech Pathol. and Audiol., Kent State Univ., Kent, OH)

Performance on auditory perceptual tasks develops throughout adolescence, likely due to the development of auditory and cognitive regions of the brain. We investigated the contributions of auditory processing and cognition to frequency discrimination. Frequency discrimination thresholds were measured in three conditions using a series of 3AFC tasks. For two conditions, each stimulus was two 15-ms tone pips separated by 100 ms. These stimuli were used both with a 3-down, 1-up procedure (threshold = 79.4%) with the method of constant stimuli (threshold = 66.7%). The third condition used a 130-ms stimulus and a 3-down, 1-up procedure. The auditory electrophysiological acoustic change complex was measured passively using an 800-ms tone that changed from 1000 Hz to a lower frequency. Cognitive testing was also administered. Preliminary results suggest thresholds correlate between conditions and performance improves with longer stimuli. Adolescents (10-17 years) performed worse than young adults (18-23) on only the adaptive-tracking conditions. Cognition contributed more to these conditions than the condition using the method of constant stimuli, suggesting that age differences on adaptive conditions are caused by cognitive development. Adolescents had larger acoustic change complex responses to the frequency contrasts. Behavioral data did not correlate with the electrophysiological data. [Funded by NIDCD].

4pPP18. Musical timbre with varied amplitude envelope improves efficacy in auditory alarms. Andres E. Elizondo Lopez (Psych., Neurosci. & Behaviour, McMaster Univ., 1280 Main St. West, Hamilton, ON L8S 4M2, Canada, elizonda@mcmaster.ca), Joseph J. Schlesinger (Vanderbilt Univ., Nashville, TN), and Michael Schutz (Psych., Neurosci. & Behaviour, McMaster Univ., Hamilton, ON, Canada)

Current alarm standards used in safety critical environments (e.g., medical alarms used in hospitals) suffer from a myriad of complications with detectability, annoyance, and alarm fatigue affecting the wellbeing of patients and staff. To a large extent, these are based on the same simplistic, temporally invariant tones. Here we explore how insights from the acoustic properties of the musical triangle can aid in detection, reducing overall levels hence reducing annoyance ratings. Two tones are used, (1) a standard tone similar to those used in current medical devices, and (2) a tone synthesized based on the spectral-temporal structure of a concert triangle. We conducted a detection experiment where participants indicated if the auditory stimulus is heard when presented a range of signal-to-noise (SNR) ratios and a two-alternative force choice task to measure annoyance ratings. Although reductions in SNRs reduced detectability for the standard tone, similar deductions had no meaningful effect on detectability of tones modeled off the musical triangle. Crucially, we identified a number of triangle inspired tones which are both less annoying and more detectable than standard tones. This suggests that these more complex sounds can reduce annoyance without harming detection, offering useful insight to medical device sound design.

4pPP19. Characterizing the primary resonator in the cochlea. Wei-Ching Lin (Dept. of Mech. Eng., Univ. of Rochester, Mech. Eng., Rochester, NY 14627, wlin28@UR.Rochester.edu), Anes Macic, and Jong-Hoon Nam (Mech. Eng., Univ. of Rochester, Rochester, NY)

The cochlear traveling waves are explained by a bank of independent resonators coupled longitudinally by lymphatic fluids. Many cochlear models require at least two resonators to account for observed responses. To investigate the resonators in the cochlea, we used high-resolution optical coherence tomography to measure 2-D vibration patterns of the organ of Corti in acutely excised cochleae from young Mongolian gerbils. The excised tissues were acoustically stimulated. The transverse and radial vibrations of the basilar membrane (BM) and the tectorial membrane (TM) were obtained over their radial span. The BM vibrated from the primary to a higher mode transversely as the stimulating frequency increased. The higher-order mode appeared near the best frequency (BF) of the measured location. Meanwhile, the TM showed no sign of a mode transition up to 1 octave above the BF in radial or transverse vibrating patterns. Within the physiological frequency range, the BM exhibits the characteristic behavior of a resonator. In contrast, the TM does not. Our results suggest that the TM as the second resonator is not the universal mechanism across the entire cochlea.

4pPP20. Acoustic indicators of voice quality in the context of social support. Luisa E. Hernández Melo (Integrated Program in Neurosci., McGill Univ., 3495 University St., Montreal, QC H3A2A8, Canada, luisa.hernandezmelo@mail.mcgill.ca) and Marc D. Pell (School of Commun. Sci. and Disord., McGill Univ., Montreal, QC, Canada)

Is social support communicated throught the subtle, yet powerful, acoustic variations in speech? This study attempted to answer this question by testing whether acoustic parameters vary when expressing social support. Participants underwent an experiment in which they watched video testimonials of a woman describing either a neutral subject or a sensitive, emotionally-charged experience. After this, participants provided voice messages to the person appearing in the testimony. Employing the openSMILE toolkit, we extracted from these speech responses the Geneva Minimalistic Acoustic Parameter Set (GeMAPS), a set of emotion-related acoustic features. Our investigation reveals an acoustic profile characteristic of supportive speech, distinguished by changes in the Alpha ratio, spectral slope, and the Hammarberg index — parameters representing the high-frequency content and spectral balance. These acoustic differences not only help to differentiate supportive utterances but also characterize its voice quality, thereby enhancing the emotional richness of this affective stance. Our research findings have potential applications in therapeutic and communication settings and open avenues for further exploration in speech science.

4pPP21. The contributions of acoustic and non-acoustic factors to spatial release from masking. Morgan Barkhouse (Speech-Lang. Pathol. and Audiol., Towson Univ., Towson University, Towson, MD 21252, mbarkh1@students.towson.edu), Sadie O'Neill, Chhayakanta Patro, and Nirmal Kumar Srinivasan (Speech-Lang. Pathol. and Audiol., Towson Univ., Towson, MD)

The ability to identify speech at worse signal-to-noise ratios when the maskers are spatially separated compared to when they are colocated is called spatial release from masking (SRM) and is thought to be a combination of monaural and binaural advantages arising from spatially separating the target from the maskers. Also, a host of other cognitive skills including attention, working memory, and executive function that support listeners' ability to segregate, track and attend to a "target" signal while tuning out other unwanted signals contributes to understanding speech better in complex environments. Previous research with young normal hearing listeners from our lab indicated that SRM was predicted by an individual's ability to use binaural cues (ITD thresholds), switch attention between two streams of information (trail making test) and ability to suppress irrelevant information (flanker task). Here, we present data from older listeners with varying hearing abilities on acoustic (envITD sensitivity) and non-acoustic tasks measuring attention (auditory and visual single and dual task), processing (trail making task), executive control (flanker task) and speech in noise tests (spatial release from masking using coordinate Response Measure sentences). The relationship between these acoustic and non-acoustic factors to speech understanding in complex listening environments will be discussed.

4pPP22. Spatial release from masking with simulated hybrid cochlear implant speech. Bailey Borkowski (Speech Lang. Pathol. and Audiol., Towson Univ., 8000 York Rd., Towson, MD 21252, bborkowski@students.towson.edu), Morgan Barkhouse (Speech Lang. Pathol. and Audiol., Towson Univ., Towson, MD), Chhayakanta Patro (Speech Lang. Pathol. and Audiol., Towson Univ., Towson, MD), and Nirmal Kumar Srinivasan (Speech Lang. Pathol. and Audiol., Towson Univ., Towson, MD)

Spatial release from masking (SRM) is the improvement in speech intelligibility when the masking signals are spatially separated from the target signal. Cochlear implant (CI) users utilizing electric-only (E-Only) stimulation had poorer speech recognition than CI users utilizing electric-acoustic stimulation (EAS). Previous research investigating SRM on hybrid CI users big spatial separations ($\pm 90°$) between the target and the maskers which were unrealistic in conversational settings. Here, we present SRM data using natural speech, simulated CI speech, and simulated EAS speech from young, normal hearing listeners for smaller, yet realistic, spatial separations between the target and the maskers. An eight-channel noise-excited vocoder was used to simulate cochlear implant processing and low-frequency filtering was used to simulate residual low-frequency hearing. Five spatial configurations were used: (colocated (target and the two maskers presented from 0° azimuth) and one of four spatially separated conditions (target at 0°, symmetrical maskers at $\pm 5°$, $\pm 10°$, $\pm 15°$, or $\pm 30°$). Initial analysis of the data revealed that the listeners had significantly poorer speech identification thresholds for the simulated CI speech when compared to EAS speech. The individual variability and the relationship between the speech identification thresholds in the three conditions will be discussed.

4pPP23. Spatial release from masking and pupillometry. Rebecka Livingstone (Speech-Lang. Pathol. and Audiol., Towson Univ., Speech-Lang. Pathol. and Audiol., 8000 York Rd., Towson, MD 21252, rliving1@students.towson.edu), Chhayakanta Patro, and Nirmal Kumar Srinivasan (Speech-Lang. Pathol. and Audiol., Towson Univ., Towson, MD)

Speech perception in complex listening environments is driven by both the auditory factors and the cognitive abilities. There are various ways in which cognitive processing while perceiving speech could be measured. In the present study, we applied pupillometry to assess cognitive processing load resulting due to various demands imposed by the presented speech. PupilCore eye glasses were used to capture pupil dilation while doing two tasks: 1. Speech recognition task when the target and the maskers were either colocated or spatially separated, and 2. a localization task where the listeners had to discriminate whether the two sounds originated form the same location or different locations. Initial data analysis revealed different patterns of pupil dilations for the collocated and spatially separated conditions. For the collocated condition, the listeneing effort had a u-shaped pattern as the target-to-masker increased whereas the listening effort linearly increased as the target-to-masker ration increased for the spatially separated condition. Also, the pupil dilation decreased as the localization task became easier. The intricate relationship between listening effort and listening environment will be discussed in detail.

4pPP24. Relationship between attention and spatial processing abilities. Sadie O'Neill (Speech-Lang. Pathol. and Audiol., Towson Univ., 8000 York Rd., Towson, MD 21252, soneil7@students.towson.edu), Morgan Barkhouse, Chhayakanta Patro, and Nirmal Kumar Srinivasan (Speech-Lang. Pathol. and Audiol., Towson Univ., Towson, MD)

Reduction in an individual's hearing ability and their working memory (WM) capacity are hypothesized to be two of the major contributors to the decline in listener performance seen in complex listening environments. Generally, the improvement in speech intelligibility that may occur when a target is spatially separated from competing talkers is quantified as spatial release from masking (SRM). The goal of this study was to estimate working memory capacity based on a divided attention version of the classic spatial release from masking task (the temporal overlap task) and the classic abbreviated reading span task (aRST). Temporal overlap threshold was estimated by adaptively varying the maximum amount of temporal overlap of the signals where a listener could still correctly identify the speech source presented directly ahead of the listener when the speech material was either colocated or spatially separated. Initial analyses of the data revealed a strong relationship between SRM and temporal overlap thresholds, suggesting that SRM is driven by the listeners' ability to modulate their attentional mechanisms. The relationship between SRM, temporal overlap thresholds, and reading span measures will also be discussed.

4pPP25. Comparison of spectral resolution and spectro-temporal modulation sensitivity well above and near the hearing threshold. Ryan Hildebrandt (Commun. Sci. and Disord., Univ. of South Florida, 4202 E Fowler Ave. PCD 1017, Tampa, FL 33620, rhildebrandt@usf.edu), Erol J. Ozmeral (Commun. Sci. and Disord., Univ. of South Florida, Tampa, FL), Stefan Klockgether, Julia Rehmann (R&D, Sonova AG, Stäfa, Zürich, Switzerland), Matthias Keller (R&D, Sonova AG, Staefa, Switzerland), and David A. Eddins (Commun. Sci. & Disord., Univ. of Central Florida, Tampa, FL)

Spectral resolution and spectro-temporal modulation (STM) sensitivity both show a correlation with speech reception thresholds in noise. Both are also sensitive to the difficulties resulting from sensorineural hearing loss; however, the relationship between the two is not fully understood. The present study investigated the potential relationship between spectral resolution and STM sensitivity and the impact of presentation level in young, normal hearing listeners. Four noise carriers were tested with the two broader bandwidths and two 1-octave wide bandwidths: 200–6400 Hz, 1250–5000 Hz, 1250–2500 Hz, & 2500–5000 Hz. Presentation levels were 70 dB SPL and 20 dB SL relative to the individual's pure-tone frequency-shaped noise carrier threshold. Spectral resolution was tested using a phase reversal paradigm, also known as a spectral ripple discrimination task, with thresholds measured in cycles per octave. Spectral, temporal, and STM sensitivity were each tested using a depth detection task, with thresholds measured in dB. On a test-by-test case, results were aligned with past studies that included a single test. The results show that both sensitivities are distinct at a low presentation level close to individual hearing thresholds, and a mixed relationship between both measures indicate that spectral resolution may only be one contributor to STM sensitivity.

4pPP26. Post-auricular orientation of auditory attention in sound field versus virtual sound space. Melina Markotjohn (Audiol., Dalhousie Univ., 816-1715 Lower Water St., Halifax, NS B3J0J4, Canada, ml314650@dal.ca) and Steven Aiken (Audiol., Dalhousie Univ., Halifax, NS, Canada)

The post-auricular muscle in many species' changes orientation of external ears to improve hearing for biologically relevant sounds. The muscle exists in humans but cannot similarly change direction of their ears. Objectives focused on measuring activity of muscle during speech-in-noise task where orientations of speaker and noise were controlled experimentally to determine how signal-to-noise varies as function of presentation mode and azimuth (target speech and noise co-localized, 45°, or spatially separated, 135° and 45°, respectively). It was hypothesized that activity would be elicited in same proportion of subjects when evoked via earphones compared to speakers; there would be no significant differences in magnitude between conditions; maximum engagement would be observed with speech 135°, noise 45°. Activity was recorded with electrodes affixed around ears, outer canthi and neck, while listeners completed a spatialized listening test (locations of speaker/noise controlled experimentally). There was significant main effect of channel; significant interaction between presentation mode and channel; no significant differences between presentation modes for other muscles; no significant effect of azimuth. Engagement in virtual sound-space suggests that muscle activation occurs consequently of spatially directed attention, even when changes in pinna orientation are unlikely to have effect on sound heard.

4pPP27. Exploring the influence of bilingual experience on speech-in-competition measures. Katia Padilla-Bustos (Lab. of Cognit. and Clinical Neurophysiology, National Inst. of Neurology and Neurosurgery, Av. Insurgentes Sur 3877, La Fama, Tlalpan, Ciudad de México, CDMX, Mexico City 14269, Mexico, katia208padilla@gmail.com), Frederick J. Gallun (Oregon Hearing Res. Ctr., Oregon Health and Sci. Univ., Portland, OR), Aaron R. Seitz (Dept. of Psych. and Ctr. for Cognit. and Brain Health, Northeastern Univ., Boston, MA), Rodolfo Solís-Vivanco (Lab. of Cognit. and Clinical Neurophysiology, National Inst. of Neurology and Neurosurgery, Mexico, Mexico), and Esteban Sebastian Lelo de Larrea-Mancera (Dept. of Psych. and Ctr. for Cognit. and Brain Health, Northeastern Univ., Boston, MA)

Speech-in-Competition (SIC) tasks mimic real-world environments by including background noise and multi-speaker scenarios. However, few SIC tasks have been translated into Spanish and validated. Moreover, the influence of bilingual experience on SIC measures is not well understood. This study aimed to investigate how bilingual experience influences performance on speech-in-competition tasks conducted in Spanish versus English. We tested fifty-six Mexican undergraduate students whose native language was Spanish who self-rated English dominance on a 10-point scale ($M = 5.8$, $SD = 2.3$). Participants performed better on the Spanish version of a spatial release from masking compared to the English version but had similar performance in a digits-in-noise test. The relationship between performances and bilingual experience was tested by creating a linguistic profile based on five dimensions of the second language. Statistically significant medium-size correlations were found between performance and the linguistic dimensions of status, proficiency, and history, but not with demand of use, or stability. This indication that the influence of bilingual experience on SIC measures can be substantial suggests that SIC tasks may be inappropriate for individuals whose native language is not English. This also suggests that other linguistically diverse populations may need specialized SIC measures as well.

4pPP28. Cochlear hair bundle dynamics: modeling calcium effects and row-wise interactions. Varun Goyal (Mech. Eng., Univ. of Michigan, Ann Arbor, 3632, G.G.B. Labs., 2350 Hayward St., Ann Arbor, MI 48109, varungo@umich.edu) and Karl Grosh (Mechancial Eng., Univ. of Michigan, Ann Arbor, MI)

External vibrations cause the eardrum to oscillate, resulting in the excitation of the sensory structures of the cochlea. In this talk, we focus on the hair bundles (HBs) of the outer hair cells situated inside the cochlea. These bundles, protruding from the cells, convert mechanical motion into electrical currents. Experimental observations reveal that the calcium concentration inside the stereocilia (hair-like structures that comprise the HB) influences current adaptation. This regulation impacts shifts of the curve, sensitivity, and active range of bundles. We aim to mechanistically understand the intracellular calcium effects by adjusting the adaptation complex in our HB model. We modified model parameters to match experimental data on current, bundle displacement, and shift trends at different calcium concentrations. This involved increasing the adaptation stiffness, stall force, stereocilia pivot, and gating stiffness. A stiffer adaptation spring reduces ion-channel reclosure, affecting the steady-state response. A higher stall force lessens the effective force on the adaptation complex, replicating experimental observations. Unlike the other properties, pivot, and gating stiffness likely do not depend on calcium concentration. Therefore, we will conduct error minimization analyses to identify adaptation complex properties while maintaining constant pivot and gating stiffness across calcium concentrations. Work supported by NIH grant NIH-NIDCD-R01 04084.

4pPP29. The frequency and level dependence of across-frequency binaural interference for interaural time differences. Allison Choi (Hearing and Speech Sci., Univ. of Maryland-College Park, 0119 Lefrak Hall, College Park, MD, achoi123@umd.edu), Anhelina Bilokon (Hearing and Speech Sci., Univ. of Maryland-College Park, College Park, MD), and Matthew J. Goupell (Hearing and Speech Sci., Univ. of Maryland-College Park, College Park, MD)

Binaural hearing enables listeners to localize sound in the horizontal plane, even when dealing with intricate speech-like signals characterized by distributed energy across a broad range of frequencies and varying sound pressure levels. Low-frequency interaural time differences (ITDs) are the most important cue for human horizontal-plane sound localization. Across-frequency binaural interference tasks (where changes in ITD discrimination thresholds of a target signal are measured in the presence of a remote interferer signal) can be used to measure how a listener weighs different frequency regions for ITD. While it is known that low-frequency (0.5 kHz) targets experience less binaural interference from higher-frequency (4 kHz) interferers than vice versa, the impact of the relative level of the two signals has remained mostly unexplored. ITD discrimination thresholds were measured in normal-hearing listeners with target/interferer intensities of 35, 55, and 75 dB-A and frequencies of 0.5, 4, and 8 kHz. Besides the expected frequency effects, data to date show binaural interference increases with increasing interferer level. These results help better understand how sound level variation and more realistic complex sounds like speech are localized.

4pPP30. Finding the ratio of perceived duration between tones with flat and percussive amplitude envelopes. Connor Wessel (Psych., Neurosci., and Behaviour, McMaster Univ., 187 Oak Ave., Hamilton, ON L8L5M9, Canada, wesselc@mcmaster.ca), Cindy Zhang (Faculty of Health Sci., McMaster Univ., Hamilton, ON, Canada), and Michael Schutz (School of the Arts, McMaster Univ., Hamilton, ON, Canada)

The extensive literature on duration assessment generally uses tones with clear onsets and offsets. However, simplistic sounds can fail to evoke the same processes used when listening to sounds with time varying amplitude envelopes (Schutz & Gillard, 2020). To facilitate future research on the

duration assessment of time varying tones, here we explore the ratio at which constant amplitude 'flat' and varying amplitude 'percussive' tones are perceived as the same duration. We will use an adaptive staircase procedure, in which flat and percussive tones are presented in pairs and participants state which tone sounded longer in duration. Each response changes the duration difference on subsequent trials and continues until responses converge around a specific point, with convergence defined as four consecutive reversals in response direction. One instance of these trials constitutes a single staircase; we will then calculate the millisecond point of subjective equality between flat and percussive tones by finding the average point of convergence between multiple, interleaved staircases. Beyond providing guidance on stimulus durations for studies comparing amplitude envelope, we aim to shed light on the process of duration assessment in sounds with time-varying amplitude envelopes.

THURSDAY AFTERNOON, 16 MAY 2024 ROOM 201, 1:00 P.M. TO 3:05 P.M.

Session 4pSA

Structural Acoustics and Vibration, Education in Acoustics and Physical Acoustics: Structural Acoustics and Vibrations Tutorial

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

Peter Kerrian, Cochair
ATA Engineering Inc., 13290 Evening Creek Dr. S, San Diego, CA 92128

Stephanie Konarski, Cochair
Johns Hopkins University Applied Physics Laboratory, 11100 Johns Hopkins Road, Laurel, MD 20723

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

Chair's Introduction—1:00

Invited Paper

1:05

4pSA1. How to model vibration damping in complex materials and structures. James G. McDaniel (Mech. Eng., Boston Univ., Boston University, Dept. of Mech. Eng., Boston, MA 02215, jgm@bu.edu)

Damping is perhaps the most challenging aspect of any model where damping is relevant. For example, an accurate damping model is often important when modeling vibration of metamaterials, vehicles, and machinery. Mass density is easily and accurately calculated from simple measurements of weight and volume. Elastic moduli are easily and accurately calculated from simple measurements of stress and strain. Large databases of mass density and elastic moduli are readily available and accurate for most materials. But damping is elusive. Many models have been proposed in many different research communities. Published numerical values for the parameters are in short supply. But the choice of damping model may dramatically affect the predicted response. The damping model is likely to make the difference between a useful result and a useless result. This presentation seeks to collect and summarize knowledge related to damping to help us model damping more accurately and more efficiently. The three most common models of damping are reviewed. These are material damping, viscous damping, and Coulomb damping. For each, an analysis is presented that begins with the fundamental assumptions and ends with the inclusion of the damping mechanism in finite element models. [Work supported by ONR under Grant N00014-22-1-2785.]

THURSDAY AFTERNOON, 16 MAY 2024 ROOM 214, 1:15 P.M. TO 5:15 P.M.

Session 4pSC

Speech Communication: Speech Production Poster Session

Kelly Berkson, Chair
Linguistics, Indiana University, Bloomington, IN

All posters will be on display from 1:15 p.m. to 5:15 p.m. Authors of odd-numbered abstracts will be at their posters from 1:15 p.m. to 3:15 p.m. and authors of even-numbered abstracts will be at their posters from 3:15 p.m. to 5:15 p.m.

Contributed Papers

4pSC1. Location of constriction in velar sounds in French. Md Jahurul Islam (Linguist., The Univ. of BC, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, jahurul.islam741@gmail.com), Victor Wong, Dayeon Choi, and Bryan Gick (Linguist., The Univ. of BC, Vancouver, BC, Canada)

This study examines the constriction location (CL) of the velar stop [k] in French. While previous studies have investigated how vocalic contexts influence the CL of velar stops [Liker & Gibbon (2008) *Clin. Ling. & Phon. 22*(2); Tabain (2000) *JPhon 28*(2)], few investigated high-level changes in CL. Building on our prior work [Islam & Gick (2023) *JASA* 154], which showed an unexpected palato-velar articulation of [k] followed by [a] compared with [w] followed by [a], we tested whether this [k]-palatalizing is universal or specific to [a]. Using a French MRI speech corpus [Isaieva *et al.*, *Scientific Data* 8], we measured constriction in [k] before [i], [ɛ], [o], [u], and [a]. MRI video frames were manually traced to mark the upper surface of the tongue and the lower surface of the hard palate, resulting in two contours. Using Euclidean distance, the location of the constriction was identified as the narrowest point and distance, respectively, between the two contours. A Python script that measured Euclidean distance from traced MRI frames calculated these distances from the MRI frames. Results indicate a frontal shift in [k] across all contexts, indicating a general fronting articulation of [k] in French speech.

4pSC2. Gender-related variation in /s/ and /ʃ/ in child-directed speech. Eugene Wong (Univ. of Minnesota, 164 Pillsbury Dr. SE, University of Minnesota, Minneapolis, MN 55455, wong0703@umn.edu) and Benjamin Munson (Univ. of Minnesota, Minneapolis, MN)

Previous research showed that 3-year-old children assigned male at birth begin to produce /s/ differently from children assigned female at birth (Munson, Koeppe & Lackas, 2022). This mirrors the differences in /s/ found between cisgender men and women in American English (e.g., Jongman, Wayland & Wong, 2000; Munson, McDonald, DeBoe, & White, 2006). The current study investigates whether children are exposed to the gender-marking /s/ variants through child-directed speech, a hypothesis suggested by Foulkes, Docherty, and Watt (2005). We collected speech samples from 36 mothers of children aged 2 to 3 years through a story-reading task. Mothers read a male-themed story and a female-themed story to their children. We also measured mothers' attitude towards their children's gendered behavior. Spectral characteristics of /s/ and /ʃ/ were measured. As there was no evidence that /ʃ/ marks gender, the acoustic characteristics of /ʃ/ is measured to control for the potential influence of anatomical variation on fricative acoustics. Analysis is ongoing, and will examine whether the acoustic characteristics of /s/ in male- and female-themed stories resemble the male- and female-typed /s/ variants found in previous studies. The results of this study will inform models of the development of gendered speech in children.

4pSC3. Acoustic cues in perception of reduced speech. Song Yi Kim (Dept. of Linguist., The Univ. of Arizona, 3201 E Fort Lowell Rd., Tucson, AZ 85716, songyikim@arizona.edu) and Natasha Warner (Dept. of Linguist., The Univ. of Arizona, Tucson, AZ)

In conversational speech, sounds undergo reduction and deletion, which challenges listeners to decode reduced forms using various cues. Previous research argues for the predominance of acoustic cues in a study of listeners' perception of reduced speech such as "he's" vs. "he was", which can both be realized [ɚz]. Analyzing data from a previous perception experiment (Warner *et al.*, Brain Sci., 2022), this study explores what types of acoustic cues listeners use to process reduced speech. Ambiguous [ɚz] with low second formant (F2) or long duration might be perceived as past due to an additional consonant (/w/) and potential contractions (low F2 as a cue to /w/). Low F2 is expected to result in smaller Bark F2-F1 and larger Bark F3-F2. Current study tests whether the stimuli show acoustic differences in the predicted direction and whether the formant and duration measures correlate with listeners' identification of tense. It was found that there are no significant acoustic differences between past and present forms due to reduction, but that listeners significantly associated smaller Bark F2-F1 with past tense in singular verbs and longer duration with past tense in plural verbs. This confirms listeners' use of the predicted acoustic cues in perceiving reduced speech.

4pSC4. Bidirectional C-to-V coarticulation across syllable and word boundaries. Scarlet Wan Yee Li (Dept. of Linguist., Univ. of Ottawa, Hamelin Hall, Rm. 401, 70 Laurier Ave. East, Ottawa, ON K1N 6N5, Canada, wli240@uottawa.ca) and Suzy Ahn (Dept. of Linguist., Univ. of Ottawa, Ottawa, ON, Canada)

Extensive work has explored the anticipatory or carryover coarticulatory effects of consonants on F2 of adjacent vowels using locus equation. However, studies examining bidirectional effects remain scarce. This study examines how voicing and place of articulation (PoA) of consonants affect bidirectional C-to-V coarticulation across syllable and word boundaries. Recordings of $/C_1V_1.C_2V_2/$, $/C_1V_1\#C_2V_2/$ and $/C_1V_1C_2\#V_2t/$ sequences from native speakers of Canadian English were analyzed. F0, F1, and F2 were measured at the onset and offset of the target vowels (/i ɑ/) to compare the coarticulatory effects caused by the adjacent consonants with different voicing and PoA (/p b t d g k/). Preliminary results ($N = 6$) indicate that coarticulatory effects vary based on voicing and PoA of consonants. Voiceless consonants exhibited a greater effect on f0 bidirectionally, but smaller effects on F1 and F2. Regarding PoA, only the effect of F2 in anticipatory contexts was dependent on different PoA. The findings also suggest that bidirectional coarticulatory effects are found across both syllable and word boundaries. Within and across these boundaries, larger anticipatory effects were found on f0, while the carryover effect on F1 was more robust. Concerning F2, differences between coarticulatory effects were only evident in voiceless and velar contexts.

4pSC5. Phonetics characteristics of seoul korean tense stops. Harim Kwon (English Lang. and Lit., Seoul National Univ., 4400 University Dr., 3E4, Fairfax, VA 22030, harimkwon@snu.ac.kr) and Suzy Ahn (Dept. of Linguist., Univ. of Ottawa, Ottawa, ON, Canada)

This study investigates Seoul Korean tense stops, asking whether underlying intervocalic tense stops in /at*a ap*a ak*a/ share phonetic characteristics with those derived from a sequence of two lax stops by applying the post-obstruent tensing rule /at.ta ap.pa ak.ka/ → [at*a ap*a ak*a]. To examine whether the tense stops show articulatory fortition compared to lax stops, and if so, whether the underlying and derived tense stops differ from each other, tongue configuration and closure duration of tense stops in [at*a ap*a ak*a] that are underlyingly either /at*a ap*a ak*a/ or /at.ta ap.pa ak.ka/ are compared with those of lax stops in [ata apa aka]. Ultrasound tongue imaging data from six native speakers of Seoul Korean reveal indistinguishable tongue configurations among underlying tense stops, tense stops derived from underlying lax sequences, and lax stops. Both derived and underlying tense stops have longer closure duration than lax stops, but the two surface tense stops did not significantly differ in their closure duration. Taking the articulatory and acoustic evidence together, underlying and derived tense stops in Seoul Korean are phonetically indistinguishable from each other although both have longer closure durations than singleton lax stops. The findings further suggest that Seoul Korean may not use tongue configuration as an articulatory strategy for fortition for either underlying or derived tense stops.

4pSC6. Abstract withdrawn.

4pSC7. Dialect effects in hesitations in older children's spontaneous speech. Ewa Jacewicz (Speech and Hearing Sci., The Ohio State Univ., 1070 Carmack Rd., 110 Pressey Hall, Columbus, OH 43210, jacewicz.1@osu.edu), Emma Cronin (Speech and Hearing Sci., The Ohio State Univ., Columbus, OH), Quinn Baumgartner (Speech and Hearing Sci., The Ohio State Univ., Columbus, OH), and Robert A. Fox (Speech and Hearing Sci., The Ohio State Univ., Columbus, OH)

Natural speech is interspersed with pauses, word repetitions, phrase repairs, and other disfluencies. These hesitations, reflecting speakers' performance delays including word fillers, self-corrections, and filled or silent pauses are typical traits of spontaneous speech. Acoustic characteristics of hesitations have been studied primarily in adults, providing evidence that at least some hesitation types exhibit speaker-inherent strategies that are language specific. It is unknown whether older children, whose abilities to engage in conversation approach those of adults, utilize hesitations that reflect the adult model prevalent in a given speech community. Here, we examine spontaneous conversations of American English-speaking 8–12 years-old children growing up in the South (Western North Carolina) and in the Midwest (Central Ohio). We seek to determine whether hesitation types in their speech are shaped by features of their regional dialect. Acoustic analyzes include breathing pattern, phrase length, articulation rate, silent and filled pauses, filler vocalizations, and repair strategies. Preliminary findings indicate that dialect does have an effect on the nature and frequency of hesitations. The work is currently ongoing, and the results will be discussed.

4pSC8. Where does the gamma go?: Acoustics of the non-labialized voiced velar approximant in Tlingit. Amanda Cardoso (Linguist., Univ. of BC, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, amanda.cardoso@ubc.ca), Simone Brown, Omar Lahlou, Ella Paulin, and James A. Crippen (Linguist., McGill Univ., Montreal, QC, Canada)

The acoustic properties of the voiced velar approximant [ɰ] are poorly characterized, partly from understudy of relevant languages and partly from uncertainty about the sound itself. We investigate this sound in Tlingit, a critically endangered indigenous language of northwestern North America, where [ɰ] contrasts with labialized [w] and palatal [j] in some dialects but not in others where it was lost through sound change (reductive primary split with merge). We use audio recordings of spoken narratives spanning much of the 20th century from speakers with and without [ɰ] to characterize this sound both dialectally and diachronically. We measure formants (f1–f3) to identify place information, amplitude for oral aperture, and harmonic-to-noise ratio for manner. Existing descriptions and phonological patterns predict that [ɰ] is an approximant and not a fricative so we compare against known approximants for differences in turbulence and spectral properties. We also compare against labialized sounds including contrastive [w] for absence of lowered f3 to exclude lip rounding in [ɰ]. We argue that it is the intersection of these acoustic cues that distinguishes [ɰ] from similar sounds, but that these cues are unstable so that dialectal sound change arises from misperception in perturbations of their production.

4pSC9. Signaling phrasal prosody through multimodal coordination. Yoonjeong Lee (Linguist., Univ. of Michigan, 611 Tappan St., 421 Lorch Hall, Ann Arbor, MI 48109-1220, yoonjeol@umich.edu), Alison McGrath, Vikas Tatineni (Linguist., Univ. of Michigan, Ann Arbor, MI), and Jelena Krivokapić (Linguist., Univ. of Michigan, Ann Arbor, MI)

This study examines the relationship between prosodic structure and non-referential co-speech gestures in Seoul Korean, a language that lacks lexical stress and uses phrasing to mark prominence. Acknowledging this crucial role of phrasing in Seoul Korean and considering the importance of prominence in orchestrating speech and co-speech gestures across diverse languages, we predict inter-articulator coordination through prosodic boundaries in Seoul Korean. Eight Seoul Korean speakers read a children's story while their speech, manual, and head gestures were recorded using electromagnetic articulography and camcorders. Results uncover systematic inter-articulator coordination patterns, deviating from those observed in languages that use pitch accents to mark prominence. Manual gestures serve as global phrase markers, synchronizing with both phrase-edge segment and tone gestures, demonstrating substantial stability. The manual gesture onset aligns precisely with the phrase-initial tone gesture onset and constriction gesture target. Similarly, the manual gesture target aligns with both the phrase-final boundary tone target and constriction gesture target. Head gestures, serving as local phrase-edge markers, precisely align with phrase-edge syllables, displaying onset-to-onset and max-to-max inter-articulator coordination patterns. Findings underscore the impact of linguistic structure on coordinating speech and co-speech gestures, elucidating how language-specific prosodic structure is conveyed through multimodal coordination.

4pSC10. Unflappable? Buckeyes say "whatever" to English flapping rules. Sean A. Fulop (Linguist., California State Univ. Fresno, 5245 N Backer Ave. Linguist PB92, Fresno, CA 93740-0001, sfulop@csufresno.edu), Mark P. Ryan (Albany, OR), and Hannah J. Scott (Comput. Sci., Oregon State Univ., Corvallis, OR)

American English dialects generally use a 'flap/tap' consonant in some places where /t/ or /d/ is both indicated in the spelling and used in some other dialects. What environments allow a flap to occur instead of /t/ or /d/? This question has inspired considerable debate in the literature, but a consensus has settled on a lack of stress on the syllable following as being the most critical factor in the occurrence of a flap word-internally. This study investigates the occurrence of flaps in the Buckeye Corpus of American English (40 speakers from Columbus, OH), detailing the frequency of flapping in various phonetic environments. One significant finding is that all speakers produced some flaps word-internally before syllables bearing primary or secondary stress (as in "whatever").

4pSC11. Dynamic auditory-acoustic properties differentiate word-final fricatives in Gitksan. Una Y. Chow (Linguist., Univ. of BC, Totem Field Studios, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, una.chow@ubc.ca) and Molly Babel (Linguist., Univ. of BC, Vancouver, BC, Canada)

Gitksan is an endangered, understudied Tsimshian language spoken in northwestern British Columbia. This study aims to provide an acoustic-phonetic description of Gitksan fricatives. Isolated words containing word-final fricatives were recorded with three Gitksan first-language speakers (2 male, 1 female; age 65+). Postvocalic fricative tokens ([s] = 159, [ɬ] = 123, [ç] = 180, [X^w] = 150, [χ] = 267) were analyzed. Dynamic, spectral measurements of peak frequency in equivalent rectangular bandwidths (peakERB; Reidy, 2016) were estimated from 17 20-*ms* windows centered at equidistant points between the onset and offset of the fricative. The resulting

4p THU. PM

trajectories (windows 3–15) show that the peakERB decreases as the place of articulation shifts further back in the vocal tract (alveolar [s]∼25–31, palatal [ç]∼20–23, uvular [χ]∼17–19; exception: labiovelar [X[w]]∼15–16, likely lowered by labialization). Also, the lateral alveolar [ɬ] shows a lower peakERB (∼22–24) than the central alveolar [s]. A random forest model (Breiman, 2001) was trained on 586 tokens in predicting the fricatives with peakERB measures of the 17 windows. The trained model was tested on 293 tokens; it identified [s] most accurately and [ɬ] least accurately. Additionally, the model's relative-importance ranking of the 17 peakERB predictors suggests that the middle third of the peakERB trajectory is robust in differentiating word-final fricatives in Gitksan.

4pSC12. Classification patterns in conversational English fricatives: Between- and within- speaker analyses. Viktor Kharlamov (Florida Atlantic Univ., 777 Glades Rd., CU-97, Ste 280, Boca Raton, FL 33431, vkharlamov@fau.edu), Daniel Brenner (Alameda, CA), and Benjamin V. Tucker (Northern Arizona Univ., Flagstaff, AZ)

This study examines between- and within- speaker patterns in random forest classification models for fricatives in conversational English. Prior investigations on the categorization of fricatives have mostly focused on group-level analyses and careful speech. We use a corpus of sociolinguistic interview speech from Western Canadian English, which represents a more casual speech style, and we compare group-level results to the findings of classification models on individual speakers' productions. The models use 23 different spectral, temporal, and amplitudinal measures to predict phonetic labels of the fricatives. Our results reveal that certain top-ranked predictors at the group level (e.g., spectral peak frequency, segment duration) are also important for fricative classification in the individual models. Other measures (e.g., midpoint kurtosis, RMS) show substantial variability in their relative prominence and play greater roles in individual- versus group-level modeling. We discuss the implications of these findings for our understanding of individual variation in connected speech and what the findings may imply about models of production and perception.

4pSC13. Spectral properties of Quebec French sibilants. Massimo Lipari (Dept. of Linguist., McGill Univ., 1085, Ave. du Docteur-Penfield, Montréal, QC H3A 1A7, Canada, massimo.lipari@mail.mcgill.ca) and Morgan Sonderegger (Dept. of Linguist., McGill Univ., Montreal, QC, Canada)

Most acoustic work on sibilants has focused on English /s/ and /ʃ/, examining spectra of the middle portion of these consonants. Voiced sibilants, which are rarer cross-linguistically (Ohala, 1983), as well as sibilants in other languages, have received less attention; in particular, there has been no previous work on Quebec French. Previous cross-linguistic studies have suggested important differences in sibilant acoustics exist between languages (Gordon *et al*., 2002), including in the dynamics of spectral properties over the course of the segment (Reidy, 2016). This study adds to the literature on the typology of sibilant acoustics by examining the spectral dynamics of sibilants in Quebec French (/s, ʃ, z, ʒ/). ∼28 k word-initial, prevocalic tokens from more than 100 speakers are extracted from a large corpus of parliamentary speech (Milne, 2014). For each token, multitaper spectra (Reidy, 2013) over a 20 ms window are calculated at 17 equidistant points. A variety of acoustic measures, including segment duration and spectral moments, are reported. Preliminary results examining static measures reveal that the anterior sibilants have higher spectral speak (∼5700 Hz vs ∼4600 Hz for posterior ones), as expected; unlike in English, however, /s/ and /ʃ/ have similar average duration.

4pSC14. Abstract withdrawn.

4pSC15. Regional differences in the production of tones in standard mandarin. Sishi Fei (School of Foreign Lang. and Cultures, Nanjing Normal Univ., School of Foreign Lang. and Cultures, Nanjing Normal University, Ninghai Rd., 122, Nanjing, 210097, China, Nanjing, Jiangsu 210097, China, sishi_fei@163.com)

This study investigates the intricate patterns of tonal variations in monosyllabic words across distinct regional accents of Standard Mandarin, with a focus on Shanghai, Guangzhou, and Beijing. Due to the differing tonal systems of the local Chinese dialects in the three cities, speakers from these areas may exhibit distinct patterns in their production of tones in Standard Mandarin, influenced by their native Chinese dialects. Therefore, to capture the nuances of tonal differences, the study extracts the time-normalized f0 values of four tones (i.e., level, rising, dipping, and falling tones) from monosyllabic words in a Mandarin Chinese speech corpus. The overall f0 contours of the four tones within and across the three regional accents are modeled using growth curve analysis. We expect to find that regional accents of Standard Mandarin exhibit significant tonal variations such that the tone shapes of speakers from Shanghai and Guangzhou may significantly deviate from those of Beijing speakers. Specifically, there are variations in contour shape and temporal dynamics, particularly in the quadratic term for dipping tones and the slopes of rising and falling tones. These findings can provide valuable insights into the intricate characteristics of regional tonal variations in Standard Mandarin.

4pSC16. Phonetic adaptation in conversation: The case of Cantonese tone merging. Ivan Fong (Dept. of Linguist., Simon Fraser Univ., Burnaby, BC, Canada, ivan_fong@sfu.ca), Fenqi Wang, Kira Chan, Tyne Johnson-Dhillon, Jadeyn Trasolini (Dept. of Linguist., Simon Fraser Univ., Burnaby, BC, Canada), Dawn Behne (Dept. of Psych., Norwegian Univ. of Sci. and Technol., Trondheim, Norway), Allard Jongman (Dept. of Linguist., The Univ. of Kansas, Lawrence, KS), Joan Sereno, and Yue Wang (Dept. of Linguist., Simon Fraser Univ., Burnaby, BC, Canada)

Phonetic adaptation occurs when one interlocutor adjusts their speech to converge to or diverge from that of their conversation partner to enhance intelligibility. While most research investigates segmental adaptations, our study focuses on suprasegmentals, specifically Cantonese tone merging. Some Cantonese speakers ("mergers") are found to merge certain lexical tones (e.g., mid-level Tone3 and low-level Tone6), which may cause confusions when interacting with non-merger speakers. Previous research has shown that a merger may unmerge a level tone pair (Tone3/Tone6) when shadowing a non-merger. However, still unclear is whether such changes result from automatic acoustic mimicking or reflect goal-oriented adaptations for intelligibility benefits. This study uses an unscripted conversation task involving a merger and a non-merger playing a video game, where productions of merged tones may cause confusions, thus motivating goal-oriented adaptations. Initial acoustic analyses focus on average F0 and F0 taken at 10 points along the contour in target Tone3 and Tone6 productions by mergers. Differences in these values for Tone3 versus Tone6 provide evidence that a merger is unmerging the tone pair. Preliminary results show increasing unmerging trends as the task progresses, suggesting progressive alignment toward a non-merger's productions for intelligibility gains.

4pSC17. Phonetics relating stance-taking in tonal languages-examples from Mandarin Interjections. Wu Siyu (College of Foreign Lang., Nanjing Univ. of Aeronautics and Astronautics, Jiangjun Avenue, Jiangning District, Nanjing, Jiangsu 211100, China, siyuwu@nuaa.edu.cn)

Previous studies on phonetics and stance-taking mainly focused on non-tonal languages. In this study, we found that tone variation could affect the perception of stance information. The study includes a production experiment and a perception experiment which is aimed to evaluate the stance polarity and stance strength. When the tone was nearer to its citation form as its duration got longer, the participants were more likely to identify the stance information as positive or negative. This gives us insight that there may be a express-inference process during the perception of stance-taking.

4pSC18. Examining the time-varying measures of F0 in sarcasm: Wigglines, spaciousness, and contour clustering. Csilla Tatar (Linguist., Univ. of Michigan, 611 Tappan St., Ann Arbor, MI 48109, cstatar@umich.edu), Jonathan R. Brennan, Jelena Krivokapić, and Ezra Keshet (Linguist., Univ. of Michigan, Ann Arbor, MI)

The present project examines the phonetic correlates, specifically those related to F0, of sarcastic speech. In a production study, American English speakers ($N = 12$) produced identically worded utterance pairs presented in contexts conducive to sarcasm and sincerity. Measures of F0 variability are contrasted; these are wiggliness and spaciousness [Wehrle, Cangemi,

Krüger, & Grice (2018) Proceedings of AISV], F0 range and F0 mean SD. Raw values were entered into by-speaker logistic regression models. Wiggliness and spaciousness together were found to be comparable to F0 mean SD and F0 range in distinguishing sarcasm and sincerity for eight of the speakers. Intonational characteristics were further examined via by-speaker F0 contour clustering [Kaland, 2021]. These by-speaker analyses showed that many speakers produce contours characteristic of sarcasm or sincerity, but that these contours differ by speaker. Further exploration of a subset of nine speakers' data showed that wiggliness and spaciousness alone can capture some of the differences between sincere and sarcastic contour clusters for some speakers. Speaker strategies vary in terms of F0, and sarcastic speech is characterized by reduced wiggliness and spaciousness for some of the speakers.

4pSC19. Abstract withdrawn.

4pSC20. Influence of tone, vowel, and consonant constraints on lexical selection in Cantonese. Minghao Zheng (Dept. of Linguist., Univ. of Florida, 4131 Turlington Hall, Gainesville, FL 32611, minghao.zheng@ufl.edu)

This study explores the impact of tones, vowels, and consonants on lexical selection in Hong Kong Cantonese, employing a word reconstruction paradigm. While prior research in European languages suggested greater mutability in vowels than consonants, the applicability of this hypothesis to tonal languages remained untested. Building on Wiener and Turnbull's (2016) Mandarin Chinese experiment, this study involved ten native speakers in a word reconstruction task with vowel, consonant, and tone substitution conditions, and a free choice condition which allowed participants to change either a vowel, consonant, or tone. Results revealed a preference for altering tone over vowels or consonants when transforming fake words into real ones, indicating that in Cantonese, tone carries the lowest information load and constrains lexical access least tightly. Moreover, participants exhibited greater accuracy in tone substitution compared to consonants and vowels. Contrary to the universal intrinsic vowel mutability hypothesis, this study suggests that, in tonal languages like Cantonese, vowels are less mutable than tones or consonants, carry the highest information load, and constrain lexical access most tightly. Vowels are more mutable only in the absence of lexically contrastive suprasegmental information. These findings contribute to understanding lexical selection in tonal languages, challenging prevailing notions regarding vowel mutability.

4pSC21. Investigating post-focus compression in code-switching: A comparative study of Twain Mandarin and English. Grace Kuo (Foreign Lang. and Literatures, National Taiwan Univ., 1 Section 4, Roosevelt Rd., Taipei, Taipei City 106, Taiwan, graciakuo@ntu.edu.tw)

On-focus pitch range expansion is a documented phenomenon in both English and Mandarin Chinese. However, the presence of post-focus compression (PFC) contrasts with its absence in Taiwan Mandarin, a regional variant of Mandarin Chinese. This discrepancy becomes particularly intriguing in the context of code-switching, which provides a unique lens to observe the interplay between these two languages. This study probes the manifestation and cognitive processing of PFC in code-switching constructs, employing a bilingual paradigm between English and Taiwan Mandarin. Through a carefully designed production experiment, this research addresses two questions: (a) Is PFC evidence in the code-switching utterances of simultaneous and early bilinguals? (b) Is there an observable correlation between the emergence of PRC in linguistic output and the language proficiency of late bilinguals, potentially indicating a transfer effect from Taiwan Mandarin to English? Anticipated results may substantiate the presence of linguistic transfer in the code-switching sentences of bilingual individuals of diverse English proficiencies. Furthermore, the implications of these findings may extend to pedagogical strategies, offering insights into the deployment of code-switching within multilingual educational settings.

4pSC22. The [s – > ʃ] sound change in /str-/ contexts: how temporal changes in the acoustic spectrum affect fricative percept. Christine H. Shadle (Yale Child Study Ctr., Yale Univ., 300 George St Ste. 900, New Haven, CT 06511, shadle@haskins.yale.edu), Laura L. Koenig, and Weirong Chen (Yale Child Study Ctr., Yale Univ., New Haven, CT)

Studies have documented a sound change in some English dialects whereby /s/ in the context /strV/ surfaces as [ʃtrV]. This can be interpreted as /s/ rounding for the rhotic being re-analyzed as [ʃ]. We recently measured $F_{;M}$, the frequency of the main peak in mid-fricative spectra, in seven adults producing words with /s/ and /ʃ/ across phonetic contexts and compared the acoustics to perceptual rankings on an [ʃ] to [s] scale. One speaker exhibited a perceived sound change; two, partial sound changes. It was shown that $F_{;M}$ was correlated with the perceptual rankings for /strV/ words. However, in all speakers, lowered $F_{;M}$ for /s/ in labial contexts did not affect percepts. The time course of lip rounding and tongue-tip retraction in /strV/ could help differentiate coarticulation versus a phonemic change. Interestingly, however, X-Ray Microbeam data [Iskarous, K. *et al.* (2011) Articulatory-acoustic kinematics of /s/, J. Acoust. Soc. Am. 129, 944-954] showed that /s/ constriction location was constant over time except in /str-/ context. This study will assess the amount and timing of $F_{;M}$ lowering, its perceptual consequences and likely articulatory causes, to gain more insight into the [strV] – > [ʃtrV] sound change.

4pSC23. Protective face masks affect clearly produced diphthongs by L1 and L2 English talkers. Baorian Nuchged (Dept. of Linguist., The Univ. of Texas at Austin, Atton Hall, 305 E 23rd St Ste. 4.304, Austin, TX 78712, baorian@utexas.edu) and Rajka Smiljanic (Dept. of Linguist., The Univ. of Texas at Austin, Austin, TX)

The study investigated how the use of protective face masks and language experience shape the production of listener-oriented clear speech. One L1 and one L2 English talker read sentences in a clear and conversational speaking style with and without a surgical mask. Formant trajectories between the onset and offset of two diphthongs, /aɪ/ and /eɪ/, were analyzed using Euclidean distance in the F1–F2 vowel space. The results showed that the distance between the onset /a/ and the offset /ɪ/ was larger when speech was produced without the mask and when speaking clearly. These modifications were larger for L1 talker compared to the L2 talker. The Euclidian distance for /eɪ/was only affected by speaking style. The results suggest that talkers produced hyperarticulated diphthongs characterized by larger formant movements in clear speech. The presence of a mask limited jaw movement for the diphthong containing the low vowel. Additionally, the L1 talker made larger articulatory modifications in response to the presence of a mask and listener-oriented clear speech compared to the talker with less extensive experience with the target language. These production patterns may be related to lower word recognition in noise for masked, conversational, and L2 speech found in previous work.

4pSC24. Register shifts in whistling: Investigating the influence on tongue shaping and gender-related variances. Laryssa A. Lancaster (Univ. of BC, 7480 Belair Dr., Richmond, BC V7A1B6, Canada, laryssalancaster@gmail.com), Dayeon Choi, Justine Cavaco, Chelsea Lisiecki, Jahurul Islam, and Bryan Gick (Univ. of BC, Vancouver, BC, Canada)

The study examines whistling register changes and the influence of tongue shape. In previous work, [Kaburagi *et al.*, 2018] used magnetic resonance imaging (MRI), finding the degree and type of restriction made by the tongue directly impacted whistle F0. [Belyk et.al., 2019] found similar results using real-time magnetic resonance imaging (rtMRI), however, did not specifically observe register shifts. This study aimed to validate whistle register shifts and identify their variations among individuals regarding tongue position and gender. We hypothesize that participants with broader whistling ranges will exhibit register shifts based on tongue shape. Measures based on ultrasound imaging of male and female participants recorded while completing whistling exercises will be included in the analysis. This methodology enables tongue movements to be visualized, offering a view of the interplay between whistling registers and tongue configurations. ImageJ analysis results of tongue positions from ultrasound imaging will be presented with relevance to whether individuals produce a shift in their vocal register while whistling, and whether these conclusions depend on gender, as in sung register. Implications will be discussed regarding potential similarities between register changes during whistling and tongue shapes associated with speech.

Session 4pSP

Signal Processing in Acoustics: Signal Processing in Acoustics Potpourri II

Raymond Plasse, Cochair
Nashville, TN

Trevor Jerome, Cochair
Naval Surface Warfare Center, Carderock Division, 9500 MacArthur Blvd, BLDG 3 #329, West Bethesda, MD 20817

Chair's Introduction—1:00

Contributed Papers

1:05

4pSP1. Range and depth estimates from array invariant and model inversion, compared with machine learning approach. Paul Hursky (Appl. Ocean Sci. LLC, 4825 Fairport Way, San Diego, CA 92130, paul.hursky@gmail.com)

The array invariant is derived from the waveguide invariant, and can be used to predict the shape of multipath arrival patterns on a 2D plane with axes of time and angle of arrival. The shape observed on a vertical line array is an ellipse that is a function of the waveguide invariant and the range. Once the range is obtained, ray tracing can be used to invert the multipath pattern for a depth estimate. We will compare results of using the physics-based range and depth estimator with results obtained by training a neural network on synthetic data produced by a ray tracer. We will produce two sets of exemplars to train our estimator. In one, we will provide the range as an input, and train the network to produce a depth. In the second, we will train the network to produce both range and depth. The method derived from the array invariant is expected to be brittle with respect to range-dependent bathymetry. We will compare how well the machine learning approach deals with such variability, compared to the array invariant approach.

1:20

4pSP2. Inversion of partially spanning array data for normal mode parameters using seabed impedance constraint. Paul Hursky (Appl. Ocean Sci. LLC, 4825 Fairport Way, San Diego, CA 92130, paul.hursky@gmail.com)

Previous work with fully or nearly spanning vertical line arrays has shown that normal mode shapes can be identified as the eigenvectors of the covariance matrix averaged over lengthy source tracks that served to de-correlate the modes. Recent work with partially spanning arrays has used compressed sensing or super-resolution methods like MUSIC to estimate mode wavenumbers. These methods assume the sound speed profile is available, but not the seabed properties. We will present another alternative for inverting for normal modes under these conditions, including a partially spanning vertical line array, in which sets of candidate modes are restricted to modes that satisfy the same impedance boundary condition at the seabed. Each distinct impedance condition defines a physically consistent set of modes. Thus, our inversion is sparse in impedance conditions. We demonstrate how the normal modes resulting from this process are used to form matched field processing replicas that recover source tracks in range and depth.

1:35

4pSP3. Real-time, sample-by-sample estimation of multiple signal waveforms from acoustic array data using B-splines. Garth Frazier (NCPA, Univ. of MS, NCPA, University of MS, P.O. Box 1848, Oxford, MS 38677, frazier@olemiss.edu)

This work presents a real-time signal processing algorithm that estimates *time-domain* waveforms of multiple plane wave signals on a sample-by-sample basis from data measured by an acoustic array. Moreover, the algorithm provides sample-by-sample estimates of the direction-of-arrival (DOA) of the waveforms. In this case sample-by-sample means that as each sample of data is measured by the array estimates of the waveforms and their directions-of-arrival are updated. While the basic idea can be based on almost any finite-dimensional approximation to a function space, this algorithm makes use of B-spline basis for the signal waveforms. Other features of the approach include no restrictions on signal waveform shape, automatic estimation of the number of signals present, automatic step-size adjustment for correction of signal waveform and direction-of-arrival parameters to achieve good performance while guaranteeing stability, and the use of circular (spherical) statistical models for 2-D (3-D) DOA estimates. The approach contrasts with blind source separation (BSS) methods that are based on non-Gaussian statistical assumptions and that do not assume a known array geometry nor a propagation model. This method is closely related to algorithms based on block-of-data by block-of-data processing but does not require stitching results from block processing to create time-domain waveform estimates.

1:50–2:05 Break

2:05

4pSP4. Auditory-inspired adaptive frequency tracking. Vijay Peddinti (NUWC Div. Newport, Howell St., Newport, RI 02841, vijaykumar.peddinti.civ@us.navy.mil)

One of the best frequency trackers to date is the human (mammalian) auditory system, which has evolved through millions of years of resolving classification problems. It is a versatile, elegant and powerful sound processing unit. It excels in detecting, estimating, and classifying multiple targets simultaneously even in noisy environments. Hence, mimicking even some features of the auditory system could be beneficial in developing superior frequency tracking and classification algorithms. An auditory inspired adaptive synchrony capture filterbank (SCFB) signal processing architecture for tracking signal frequency components was proposed in a related paper [JASA-2013]. The SCFB architecture consists of a fixed array of traditional, passive linear, gammatone filters in cascade with a bank of three adaptively tunable bandpass filters that form a frequency-discriminator-loop (FDL).

The SCFB exhibits many desirable properties for processing speech, music, and other complex sounds. In recent work [Dec2021], the algorithm was modified using adaptive tuning parameters, and a generalized way to determine/suppress voiced and unvoiced (silent) regions. This modified algorithm estimates frequencies with higher accuracy even in the presence of closely spaced input tones. Preliminary analysis with synthetic, human speech and humpback/whale-call signals demonstrates that the revised algorithm performs well. This talk will focus on follow-on work.

2:20

4pSP5. Multichannel signal transmission and reception using compact multichannel underwater communication devices: A unified theory and experimental results from different underwater media. Ali Abdi (Elec. Comput. Eng. Dept., New Jersey Inst. of Technol., 323 Dr. Martin Luther King Jr Blvd, Newark, NJ, ali.abdi@njit.edu), Erjian Zhang, and Rami Rashid (Elec. Comput. Eng. Dept., New Jersey Inst. of Technol., Newark, NJ)

Simultaneous transmission of multiple signals over the same bandwidth allows for optimal utilization of the limited bandwidth in underwater systems and networks. To avoid using multiple spatially separated single-channel projectors, one compact multichannel device can be used instead [E. Zhang, R. Rashid, A. Abdi, “Particle velocity underwater data communication: Physics, channels, system and experiments,” IEEE J. Oceanic Eng. (2023); E. Zhang, R. Rashid, A. Abdi, “Underwater communication experiments for transmitting multiple data streams using a vector acoustic MIMO system: OFDM and FSK modulations,” Proc. Oceans (2023)]. Simultaneous reception of multiple signals using one compact multichannel device is also feasible and improves the communication performance, without using several spatially separated single-channel hydrophones [R. Rashid, E. Zhang, A. Abdi, “On the performance of a new wireless communication compact multichannel underwater receiver using a sphere vector sensor,” IEEE Transactions on Vehicular Technology (2023); R. Rashid, E. Zhang, A. Abdi, “Underwater acoustic signal acquisition and sensing using a ring vector sensor communication receiver: Theory and experiments,” Sensors (2023)]. In this paper, we present various experimental results using a number of compact multichannel underwater devices in different setups, and demonstrate their usefulness for underwater communication. [Work supported in part by NSF, Grant IIP-1500123].

2:35

4pSP6. Dual-repetitive pilot aided parameter estimation for underwater orthogonal frequency division multiplexing acoustic communications in the presence of strong composite noise. Zhiqiang Liu (US Naval Res. Lab., 4555 Overlook Ave. Washington, DC 20375, zhiqiang@ieee.org)

In this work, a novel design of signaling parameters and pilot placement is proposed for supporting robust underwater orthogonal frequency division multiplexing acoustic communications in the presence of strong composite noise. The proposed design allows for high-resolution estimation of various channel parameters that are essential to successful symbol recovery. Receiver processing algorithms are developed by exploring special properties of the received signal that hold valid despite complicated channel and noise effects. These algorithms are efficient since they reuse the same set of pilot symbols for multiple tasks of estimation that often require separate sets of pilots. They are also extremely effective because they largely avoid the effects of noise even when the noise is composite. The performance of these algorithms is evaluated via both extensive simulations and an at-sea experiment.

Session 4pUW

Underwater Acoustics: Underwater Acoustic Modeling

Benjamin Cray, Cochair
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Jessica Desrochers, Cochair
Ocean Engineering, The University of Rhode Island, 48 Villa Avenue, Warwick, RI 02886

Contributed Papers

1:00

4pUW1. Acoustic propagation modeling using sound velocity profiles estimated from high resolution temperature data. Brian Amaral (Ocean State Sensing, 21 Admiral Kalbfus Rd., Newport, RI 02840, Oceanstate-sensing@gmail.com), Jennifer L. Amaral (Marine Acoust., Inc., Middletown, RI), Antone B. Eliasen (Ocean State Sensing, Mclean, VA), and Russell Shomberg (Marine Acoust., Inc., Newport, RI)

An accurate sound velocity profile (SVP) is an important input for underwater acoustic propagation modeling. The SVP is largely driven by the temperature of the water column, especially at shallower depths. Measuring the SVP in high spatial and temporal resolution is challenging with traditional instrumentation, however techniques exist to produce high resolution temperature measurements that can be used to derive estimates of the SVP. Fiber optic distributed temperature sensing (DTS) offers an improvement of several orders of magnitude over traditional singular point measurement devices and has the ability to measure temperature over large range and depth scales in an efficient manner. A DTS system was towed for 172 nm off the coast of New England over several days. The measured temperature data was used to estimate the SVP across the range and depth and will be compared to traditional environmental data sets, such as HYCOM. A comparison of modeled acoustic propagation using different resolution SVPs will be presented.

1:15

4pUW2. Understanding model fidelity for training synthetic aperture sonar image classifiers. Thomas E. Blanford (Univ. of New Hampshire, University of New Hampshire, Durham, NH 03824, thomas.blanford@unh.edu), David Williams (The Penn State Univ., La Spezia, SP, Italy), J. Daniel Park (The Penn State Univ., State College, PA), Brian Reinhardt (The Penn State Univ., University Park, PA), Shawn Johnson, and Daniel C. Brown (The Penn State Univ., State College, PA)

Convolutional neural networks (CNNs) are increasingly employed for classification tasks in automated target recognition (ATR) algorithms for synthetic aperture sonar (SAS) images. Training ATR algorithms, however, requires many unique observations of the targets of interest. Data available for training is often limited, and acquiring additional training data through experimentation is prohibitively expensive. Increasingly, training with simulated data is being considered as an alternative, but the fidelity required of the models that generate this data is not yet known. SAS imagery typically contains significant complexity from countless physical mechanisms. Simulating complex scenes requires multiple models to account for these different effects, but some effects may not require accurate modeling for the purposes of training an ATR algorithm. This presentation will describe a study to investigate the fidelity of models required so that simulated data may be used interchangeably with experimental data for training CNNs. Using in-air experimentation, a high-fidelity data set was developed with multiple degrees of complexity. A high-frequency sonar signal model was then used to generate complementary simulated data. This approach allows for specific physical features in the data to be individually isolated, enabling detailed exploration of the relationships between model fidelity and CNN architecture.

1:30

4pUW3. Effects of seabed corrugation and stratification on 3D acoustic propagation along the New England Shelf Break. Brendan J. DeCourcy (Woods Hole Oceanographic Inst., 86 Water St., Falmouth, MA 02543, bdecourcy@whoi.edu), Ying-Tsong Lin (Scripps Inst. of Oceanogr., La Jolla, CA), and Jason Chaytor (U.S. Geological Survey, Woods Hole, MA)

Acoustic signals transmitted and recorded during the New England Shelf Break Acoustics Experiment in 2021 (NESBA) display evidence of complex scattering effects which were not captured in *in situ* modeling efforts. In particular, although the acoustic parabolic equation models used can predict later arrivals of a single transmission, initial arrivals show mismatch between model predictions and observation. In this presentation, estimates of sub-bottom sediment structure are supplemented by a high-resolution model of the bathymetry to simulate 3D sound propagation across a realistic seabed. The influence of the seabed corrugation and stratification on acoustic arrival times is examined in the context of the NESBA environment. [This research is supported by the Office of Naval Research].

1:45

4pUW4. Acoustic arrival patterns in the Beaufort duct using normal mode and ray trace acoustic prediction models. Jessica Desrochers (Ocean Eng., The Univ. of Rhode Island, 48 Villa Ave., Warwick, RI 02886, jessicabfothergill@gmail.com), Lora Van Uffelen (Ocean Eng., Univ. of Rhode Island, Narragansett, RI), and Alexander Muniz (NUWC, Newport, RI)

The stratification of the Beaufort Sea has experienced significant changes over the last few decades resulting in a subsurface duct between 100- and 300-meters depths, known as the Beaufort Duct. This duct allows for long-range acoustic transmissions due to little interaction with the seafloor or sea surface. Acoustic arrival predictions for broadband acoustic sources centered around 250 Hz, such as those deployed in the Beaufort Sea in 2016–2017 show a peak acoustic arrival prior to the final cutoff centered on the sound speed minimum in the duct. This reverse dispersion feature in the acoustic time front can be connected back to the unique ducting features in the sound-speed profile. This relationship is explored using normal mode modeling and geometric optics. Modal speed predictions and ray path lengths and travel times are used to interpret the acoustic arrival patterns, particularly the dispersion feature present in the acoustic time front.

2:00

4pUW5. Temporal evolution of sound pulse in shallow water under conditions of horizontal refraction. Vertical modes and space-time horizontal rays. Boris Katsnelson (Marine Geosciences, Univ. of Haifa, 199 Adda Khouchy Ave. Haifa 3498838, Israel, bkatsnels@univ.haifa.ac.il) and Alexander Kaplun (Marine Geosciences, Univ. of Haifa, Haifa, Israel)

As is known, to describe horizontal refraction in a shallow water waveguide, the "vertical modes and horizontal rays" approach (Burridge & Weinberg, 1974) is used, within which the two-dimensional eikonal equation for the modal amplitude includes the refractive index determined by the eigenvalues depending on the horizontal coordinates (x,y) and frequency (f). In other words, we have an effective two-dimensional inhomogeneous dispersive medium, to describe the propagation of signals in which, the authors propose to use the so-called space-time rays (Rytov, 1940, Connor & Felsen, 1974, Hillion, 1993) applied for the horizontal plane (STHR). The set of STHRs in a given situation is constructed for each mode. Within the framework of this approach, it is possible to obtain the evolution of the pulses of each mode from the solution of the nonstationary eikonal equation in the horizontal plane or from the corresponding Hamilton-Jacobi equations, obtained using non-local integral wave equation. Examples of constructing STHR in some typical situations (in particular in a wedge) are considered, some specific effects during signal propagation are considered, in particular, the so-called the pulse front tilt, previously discovered in laser optics (Bor & Rasz, 1985). Work was supported by ISF, grant 946/20.

2:15

4pUW6. The effects of bathymetry on estimating the depth of a source in range-dependent environments with a single hydrophone. Ivars P. Kirsteins (NUWCDIVNPT, 1176 Howell St., Newport, RI 02813, i.kirsteins@gmail.com)

The effects of varying or range-dependent bathymetry, e.g., such as a sloping bottom, on the depth resolution characteristics of single hydrophone matched field processing (MFP), e.g., width of depth main-lobe as a function of source and receiver nominal positions, is not well understood. A major challenge is that in the case of range-dependent environments there are generally no analytical representations for the pressure field, thus requiring numerical methods for their evaluation. To provide some theoretical insights into how the shape of the bathymetry and locations of the source and receiver influences the MFP depth ambiguity function, we study the case of an ideal wedge with perfectly reflective pressure-release boundaries using the analytical solutions of Buckingham [1]. We derive a simple approximation for the depth ambiguity function main lobe width and compare it against the theoretical result for an ideal range-independent environment [2] and then numerically for a penetrable wedge and more realistic shelf-like environments. M.K. Buckingham, "Acoustic propagation in a wedge-shaped ocean with perfectly reflecting boundaries," NRL report 8793, March 1984. M. Cheney and I.P. Kirsteins, "Resolution of matched field processing for a single hydrophone in a rigid waveguide," J. Acoust Soc. Am. 151, pp. 3186–3197, 2022.

2:30

4pUW7. Deconvolution of the waveguide impulse response for source localization at low frequency. Florent Le Courtois (DGA TN, Ave. de la Tour Royale, Toulon 83000, France, florent.lecourtois@gmail.com), Bazile Kinda (DGA TN, Brest, France), and Myriam Lajaunie (Shom, Brest, France)

Matched-mode processing (MMP) methods have been widely investigated for source localization and geoacoustic inversion. For horizontal line arrays (HLA), it aims at retrieving the horizontal wavenumbers and thus at inferring source range and bearing. In this work, the matched-mode is performed by applying the deconvolution of the waveguide response in a narrowband context in order to improve the localization accuracy. Deconvolution usually refers to a sparse framework when the number of sources is smaller than the number of sensors of the HLA. The Orthogonal Matching Pursuit algorithm is then used in this work. First results from numerical simulations for Pekeris waveguide highlight better localization accuracy than MMP. The deconvolution shows as well stronger robustness to mismatch in the sound speed values of the seabed. Results from measurement campaign, involving large HLA, low to ultra-low frequency and water depth up to 1500 m are investigated as well.

2:45–3:00 Break

3:00

4pUW8. A study of model selection techniques for modeling synthetic aperture sonar backscatter statistics. Derek R. Olson (Oceanogr., Naval Postgrad. School, 833 Dyer Rd., 313b Spanagel Hall, Monterey, CA 93943, olson.derek.r@gmail.com) and Marc Geilhufe (Norwegian Defence Res. Establishment, Kjeller, Norway)

High resolution acoustic imaging of the seafloor, such as with synthetic aperture sonar, can reveal complex environments due to the ability to resolve scatterers of different types. The statistical distribution of the backscattered field is appropriately modeled using a mixture distribution, which consists of a sum of a K pdf for each scatterer type, weighted by the relative frequency with which they occur. In this type of modeling, the number of distributions must be selected before parameters can be fit, and as the number of components increases, so does the danger of overfitting the data. Several methods of model selection are explored here. Two methods, the Bayesian information criterion and the Akaike information criterion are based on point parameter estimates, and a penalty due to the number of model parameters. The other two, the deviance information criterion and the Watanabe-Akaike information criterion, are based on Monte-Carlo sampling of the model parameter space. These four model selection criteria are compared to each other, and to the authors manual selection of the number of components.

3:15

4pUW9. Analysis of seamount propagation through the first convergence zone. Johnathan J. Todd (Penn State, 201 Appl. Sci. Bldg., University Park, PA 16802, jjt6090@psu.edu), William Hodgkiss (Scripps Inst. of Oceanogr., San Diego, CA), Daniel C. Brown (Penn State, State College, PA), and Chad M. Smith (Penn State, State College, PA)

During the recent Office of Naval Research (ONR) New England Seamounts Acoustics (NESMA) Pilot experiment, continuous measurements were made of underwater acoustic propagation and scattering from close range through the first deep-water acoustic convergence zone (CZ) using a stationary mid-frequency source and the ONR Three Octave Research Array (THORA) at Penn State. The source was positioned at a shallow depth over the plateau of one of the Atlantis II seamounts; the THORA was towed from 1 km range to a maximum of 85 km from the source. High-level goals of this work are to: 1) assess the influence of seamount scattering on mid-frequency transmission loss (TL) within the shadow zone, and 2) characterize how seamount interaction and the CZ is influenced by the shallow sound speed profile (SSP) in this region. The data was beamformed and analyzed using narrowband and broadband techniques, while ray-based and parabolic equation (PE) based models were used to develop an understanding of the influence of the seamount bathymetry, roughness, and the SSP. This talk will discuss a frequency dependent decrease in TL (acoustic enhancement) within the shadow zone due to seamount interaction and the limited influence of the seamount within the first CZ.

3:30

4pUW10. A model-experiment validation of near-range phase center breakdown. Jonah Warner (The Penn State Univ., 805 West Aaron Dr., Apt A1, State College, PA 16803, jbw5756@psu.edu), Thomas E. Blanford (Univ. of New Hampshire, Durham, NH), and Daniel C. Brown (Penn State Univ., State College, PA)

Spatial coherence of a scattered field has helped to improve sensor navigation techniques and field imagery. Under the phase center approximation (PCA), a bistatic sensor geometry is approximated as a monostatic phase center. In the far field, the approximation is valid, and signals with common phase centers are highly coherent. The approximation fails when a non-trivial spatiotemporal delay appears in the sensor's near-field, and signals that share phase centers are less coherent. A model by Brown and Blanford shows this near-field degradation of PCA, using comparisons between the van Cittert-Zernike Theorem and a point-based scattering model. Experimental validation of the model was performed using an in-air sonar mounted

on a linear actuator. The sonar collected returns from a shallow bed filled with plastic beads, which approximates rough interfaces typical of underwater environments. Multiple trials were conducted using a variety of transmitters, receiver placements, and ping intervals. These geometries were simulated using the point-based scattering model and the two sets of data were compared in order to validate the model. Results from this model-data comparison, along with future applications for sonar imaging and navigation, will be discussed.

Invited Paper

3:45

4pUW11. Hybrid model for acoustic and vibration predictions based on vessel induced acoustic vibration: A review. Solomon O. Ologe (Mech., Univ. Polytechnic of Catalonia, Calle de Colom, 11, Tarrassa Campus, Barcelona 08222, Spain, ologe.solomon@upc.edu)

This research examines hybrid models for acoustic and vibration predictions based on vessel-induced acoustic vibration. Acoustic and vibration are caused by complex machineries and marine plants, affecting crew members and aquatic life. To protect marine species, urgent action is needed. Previous studies have developed models for estimating and predicting acoustic and vibration levels, but these models have limitations. Hybridization, combining numerical and computational methods, and simulations of vibro-acoustic behavior could improve results. Hybridization is encouraged for greater efficiency.

Contributed Papers

4:00

4pUW12. Analysis of bubble curtain effectiveness in the coastal Virginia offshore wind turbine installation. Gavin Dies (Ocean Eng., Univ. of Rhode Island, Narragansett, RI), Ying-Tsong Lin (Woods Hole Oceanographic Inst., 266 Woods Hole Rd., Woods Hole, MA 02543, ytlin@whoi.edu), Gopu R. Potty, James H. Miller (Ocean Eng., Univ. of Rhode Island, Narragansett, RI), Jade Case (Ocean Eng., Univ. of Rhode Island, Winter Park, FL), Jennifer L. Amaral (Marine Acoust., Inc, Middletown, RI), and Anwar A. Khan (HDR Inc, Fort Lauderdale, FL)

The Coastal Virginia Offshore Wind (CVOW) consists of two turbines roughly 40 km off the coast of Virginia Beach, Virginia. Water- and seabed-borne acoustic signals from impact pile driving at CVOW were recorded during the installation of these turbines in May 2020. In-water pressure signals were measured using two vertical line arrays (VLAs) at ranges of 3 km and 7.5 km. The water depth at the wind turbines and both VLAs was about 26 m. During installation, one of the piles utilized a double bubble curtain during construction while the other pile did not. Bubble curtains are used to reduce the acoustic impacts of pile driving by creating a barrier of bubbles around the source. We found that the bubble curtains were most effective at longer ranges and at frequencies above 200 Hz. Normal modes are an efficient representation of the acoustic field in this relatively range-independent shallow water environment. The amplitudes of acoustic modes at several frequencies were used to analyze the effectiveness of bubble curtains. Energy propagating in the seabed was also measured by an array of geophones and was found to be unattenuated by the bubble screens.

4:15

4pUW13. An evaluation of cross-modal acoustic intensity modulation. Benjamin Cray (NUWC, 1176 Howell St., Newport, RI 02841-1708, benjamin.cray@navy.mil)

Range to a submerged source, in a shallow-water waveguide, is determined using a single vector sensor receiver. Low order cross-modal variations in acoustic intensity (all four components, vertical and radial reactive and active) estimate target range. The algorithm relies on intensity modulations, generated from the superposition of a limited number of low-order trapped modes. Performance was evaluated with Acoustic Toolkit's SCOOTER[1,2] software (a Fast Field Program based on a contour integral representation of the range-independent waveguide's acoustic pressure). SCOOTER, which allows for sediment attenuation, multiple seafloor layers, and various sound speed profiles, generates the corresponding Depth-Dependent Green's Function. Comparisons of SCOOTER predictions to measured data, collected in Monterey Bay (California), and within the New England Mud Patch (Rhode Island), will be presented. Maggi, A. L., Duncan, A. J., "Acoustic Toolbox User interface and Post processor, AcTUP V2.2L", Centre for Marine Science & Technology, Curtin University of Technology, Perth, AU. Alec Duncan, "A Consistent, User Friendly Interface for Running a Variety of Underwater Acoustic Propagation Codes", Proceedings of ACOUSTICS 2006, 20–22 November 2006, Christchurch, New Zealand

Session 5aAA

Architectural Acoustics, Structural Acoustics, and Vibration and Noise: Absorptive and Diffusive Metasurfaces for Architectural Acoustic Application

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Ning Xiang, Cochair
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Chair's Introduction—8:00

Invited Papers

8:05

5aAA1. Optimal bi-state acoustic metamaterial for broadband sound absorption and diffusion: A real-estate dilemma. Eric Ballestero (Institut d'Acoustique - Graduate School, Laboratoire d'Acoustique de l'Université du Mans, UMR CNRS 6613, University of Le Mans, Le Mans 72085, France, eric.ballestero@univ-lemans.fr), Yang Meng (Institut d'Acoustique - Graduate School, Laboratoire d'Acoustique de l'Université du Mans, UMR CNRS 6613, Le Mans, France), Ping Sheng (Hong Kong Univ. of Sci. and Technol., Hong Kong, Hong Kong), Vincent Tournat (Institut d'Acoustique - Graduate School, Laboratoire d'Acoustique de l'Université du Mans, UMR CNRS 6613, Le Mans, France), Vicent Romero-Garcia (Instituto Universitario de Matemática Pura y Aplicada (IUMPA), Universitat Politècnica de València (UPV), València, Spain), and Jean-Philippe Groby (Institut d'Acoustique - Graduate School, Laboratoire d'Acoustique de l'Université du Mans, UMR CNRS 6613, Le Mans, France)

The advent of metamaterials has given a new breath to acoustic treatment design due to their ability to target much lower frequencies within deep-subwavelength dimensions. However, many practical applications may require a multi-functional structure instead of a static one with a single purpose and, thus, overturn geometry-specific performance. Such hybrid acoustic treatments have been an active field of research for the past decades, in which a trade-off between different acoustical mechanisms has to be struck, the major issue being the inversely decreasing returns for each acoustical mechanism, i.e., the more of one the less of the other. The aim of the present work is to build over previous metamaterial strategies and design a dynamic bi-state passive acoustic metamaterial that can change its acoustic properties within the same structural volume, going from highly absorbent ($\alpha \sim 0.85$) to diffusive ($\delta \sim 0.65$) over more than one octave for each acoustic phenomenon, i.e., 400–1100 and 1000–2500 Hz for absorption and diffusion, respectively, with an overall thickness L = 12 cm two times thinner than traditional acoustic treatments. Such design can help enhance the acoustics of rooms, but it can also be introduced to critical environments with limited space, such as aerospace.

8:25

5aAA2. Towards extraordinary sound absorption using coupled resonances. Yun Jing (Acoust., Penn State Univ., 201 Appl. Sci. Bldg., State College, PA 16802, jing.yun@psu.edu) and Jun Ji (Acoust., Penn State Univ., State College, PA)

In this paper, we provide an overview of recent advancements in novel sound absorption designs, predominantly based on optimized coupling between resonances. First, we present a sound absorption panel design capable of achieving high absorption in the low-frequency broadband range (50–63 Hz, one-third octave band). This panel demonstrates an average absorption coefficient of approximately 93%, all while maintaining a thickness of 15.4 cm (equivalent to 1/45 of the wavelength at 50 Hz). Transitioning from the one-port system, we delve into the realm of the two-port system, introducing an ultra-sparse structure for flow-free sound absorption. This structure exhibits near-perfect absorption ($\sim$99%), and this absorption capability is achieved when the spatial period of monopole–dipole resonators is close to one working wavelength (95% wavelength). These developments signify significant strides in sound absorption technology, presenting innovative solutions with enhanced efficiency and performance across various frequency ranges and environments.

8:45

5aAA3. Acoustic performance of screens designed to act as metamaterials. Umberto Berardi (Toronto Metropolitan Univ., 350 Victoria St., Toronto, ON M5B 2K3, Canada, uberardi@ryerson.ca), Antonella Bevilacqua (Università della Campania Luigi Vanvitelli, Parma, Italy), Amelia Trematerra, and Gino Iannace (Università della Campania Luigi Vanvitelli, Aversa, Caserta, Italy)

Metamaterials are, nowadays, a mature field of research in acoustic and many applications with metamaterials have been developed in recent years. In metamaterials, the attenuation of sound is due to the interaction of the sound waves with their geometrically regular structures. The frequency at which the attenuation occurs is calculated with Braggs' law. This work presents sound attenuation measurements of structures made with metamaterials carried out in an anechoic chamber. The arrangements considered are both

two-dimensional and three-dimensional: the 2D arrangement was obtained with a regular distribution of linear elements made with bars with diameters of 9, 15, and 20 mm; the 3D arrangement was obtained with a regular distribution of a lattice of spheres with a diameter of 23 mm. The results of the acoustic measurements are reported in terms of insertion loss in the range from 1000 to 10 000 Hz.

Contributed Papers

9:05

5aAA4. Transparent sound absorption—More than 25 years of applications. Christian Nocke (Akustikbuero Oldenburg, Sophienstr. 7, Oldenburg, Nds. 26121, Germany, nocke@akustikbuero-oldenburg.de)

Optically transparent micro-perforated sound absorbers were introduced more than 25 years ago. One of the first projects carried out was the German Parliament (Bundestag). another project carried out recently is the Canadian Parliament. At first, in the German Parliament, acrylic glas boxes with a sub-millimeter micro-perforations have been developed an applied. Later, on other micro-perforated sheets, such as thin polycarbonate or PVC sheets, have been applied in various projects around the world such as in Ottawa. This contribution concentrates on optically transparent sound absorbers. Fully transparent absorbers can be installed in front of glass facades, under glas roofs and whenever no optical distrubances are wanted. From pandemic times, also transparent screens and other applications are well known. A short review of the applications of various different materials with transparent microperforated sound absorbers is presented. Finally, sound absorption data for different setups are presented.

9:20

5aAA5. Some considerations on the simulation of meta-materials acoustic behavior. Francesco Martellotta (Dept. Architecture, Construction and Design, Politecnico di Bari, Via Orabona 4, Bari 70125, Italy, francesco.martellotta@poliba.it), Ubaldo Ayr, Chiara Rubino, and Stefania Liuzzi (Dept. Architecture, Construction and Design, Politecnico di Bari, Bari, Italy)

One of the major advantages behind the success of metamaterials is related to the possibility to design them based on theoretical assumptions or when theory gets complicated, thanks to numerical methods. Many efforts have been made in the last years to better support researchers with benchmark tools that allow us to make their calculations more reliable, offering the opportunity to directly model the sound propagating inside the material or allow the determination of the parameters that can be used to feed phenomenological models. However, like any other simulation tool, a proper knowledge of the physics and the ways it is approximated are required to get the best results. The paper presents a selection of issues that may play a relevant role to improve the final results, including geometrical discretization, modelling of boundary conditions, and use of periodical structures. A discussion follows also pointing out the differences pertaining to transmission and absorption problems.

Break 9:35–9:50

9:50

5aAA6. Absorptive metasurface inaccuracies made of micropore and microslit panels in multiple structures. Ziqi Chen (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, chenz33@rpi.edu) and Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY)

Microperforated panels in forms of micropores/microslits (MPP/MSP) represent absorptive metasurfaces, which can achieve high absorption with single- or double-layer MSP/MPP absorbers. However, architectural acoustics practice usually requires broadband high absorption. Multiple metastructures in more than two layers become one of possible options. Multiple MPP/MSP metasurfaces arranged in layers inherently complicate the design process. This work applies a model-based Bayesian framework employing a potentially multilayered prediction model. The Bayesian framework involves two-levels of probabilistic inference to design a parsimonious number of layers as the higher level, quantitatively implementing Occam's razor. Once the number of multilayers is selected, the MPP/MSP parameters are readily available within the Bayesian framework, so that the overall metastructure satisfies the design scheme. This work experimentally validates the designed prediction in comparison with the design scheme. However, manufacture inaccuracies may lead to unacceptable deviations. This work applies causation analysis based on a causal model to reveal causal uncertainties/inaccuracies. Using the causal model for causal inference, the Bayesian multilayer design can be satisfactorily validated. This paper discusses comparative investigations of metasurfaces made of micropores and microslits.

10:05

5aAA7. Absorptive performance of layered metasurfaces using microslit panels. Phebe S. Cunningham (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., 110 Eighth St., Troy, NY 12180, cunnip@rpi.edu), Ziqi Chen, and Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY)

This work examines the effectiveness of layered metasurface arrangements, prioritizing space efficiency to create a broadband sound absorber. Innovative metasurfaces, including microslit panels with concentrated and coiled cavities, provide frequency-dependent absorptive properties. While these metasurfaces are independently known as efficient sound absorbers, single-layer microslit panel absorbers with a single cavity, in general, lack a wide effective bandwidth. In this work, theoretical models guide the effective creation of metastructures to achieve broadband absorption. These systems are then assembled and measured using an impedance tube to validate their acoustic performance. This paper discusses the formulation of the theoretical model, experimental validation of the model for layered absorbers, and the design specifications that can be met using various metasurface combinations.

10:20

5aAA8. Robotic fabrication of clay acoustic resonators. Brady Peters (John H. Daniels Faculty of Architecture, Landscape, and Design, Univ. of Toronto, 1 Spadina Crescent, Toronto, ON M5S 2J5, Canada, brady.peters@daniels.utoronto.ca), Nicholas Hoban, John Nguyen, and Nermine Hassanin (John H. Daniels Faculty of Architecture, Landscape, and Design, Univ. of Toronto, Toronto, ON, Canada)

Digital fabrication offers the potential for the creation of complex geometries and mass-customized products; however, most 3D printers do not scale sufficiently to create architectural scale components. Robotic fabrication methods may bridge the gap, offering the possibility of architectural-scale 3d-printing capabilities. It has been found that the combination of multiple Helmholtz resonators tuned to different frequencies can create broadband absorption. This research pairs CAD parametric design with robotic clay extrusion as a method of acoustic resonator mass customization. The history of architectural acoustics together with recent archeological discoveries unveils a long-established practice of using clay vases as acoustic devices. And while the efficacy of these vases in historical settings has been contested, the use of large arrays of carefully tuned acoustic vases remains largely unexplored in contemporary practice. This paper presents the acoustic vases' unique history, defines its geometry and performance, and projects the potentials of the acoustic vase in current practice through modelling, simulation, and fabrication. A full-scale prototype wall with 126 robotically 3d-printed clay resonator vases was designed and constructed. The 1:1 prototype was shown in the "Robotic Clay," which was exhibited at the Canadian Clay and Glass Gallery in Waterloo, Canada.

10:35

5aAA9. Multiphysical design for optimized ventilation and noise attenuation in ducts: An approach using metamaterials. Gioia Fusaro, Rachele Billi, Simone D'Auria (Univ. of Bologna, Bologna, Italy), and Dario D'Orazio (Univ. of Bologna, Viale Risorgimento, 2, Bologna 40126, Italy, dario.dorazio@unibo.it)

Noise generation and propagation through mechanical ventilation and air conditioning (MVAC) ducts are among the main issues related to controlled mechanical ventilation. Commercial solutions allow sensible noise attenuation; however, they also significantly reduce the duct section, forcing the MVAC power to rise to overcome the flow resistance from the commercial silencer, increasing the noise source. The aim of this work stems from such problems, focusing on soundproofing ventilation ducts through acoustic metamaterials without decreasing the inner duct section to limit the flow losses. Moreover, a crucial part of this study falls into the multiphysical interaction between acoustics and fluid dynamics, which is a phenomenon that may result in a challenging analytical model. In conditions of flowing motion, the wave vector has been modeled according to the Lighthill model: the component of the velocity rotor potential becomes significant, influencing the soundproofing performance due to the metamaterial placed on the inner surface of the duct. This work aims to provide a proper setup for wave-based simulation and present some preliminary results.

FRIDAY MORNING, 17 MAY 2024 ROOM 210, 8:00 A.M. TO 9:45 A.M.

Session 5aBAa

Biomedical Acoustics: General Topics in Biomedical Acoustics: Methods

Keith A. Wear, Chair

Center for Devices and Radiological Health, Food and Drug Administration, Bldg. 62 Rm. 2114, 10903 New Hampshire Ave., Silver Spring, MD 20993

Contributed Papers

8:00

5aBAa1. A spatiotemporal deconvolution approach for measurements of therapeutic ultrasound pressures, intensities, and beamwidths. Keith A. Wear (Ctr. for Devices and Radiological Health, Food and Drug Administration, Bldg. 62 Rm. 2114, 10903 New Hampshire Ave., Silver Spring, MD 20993, keith.wear@fda.hhs.gov) and Sam Howard (Onda Corp., Sunnyvale, CA)

Complete characterization of high intensity focused ultrasound (HIFU) systems includes measurements of transmitted pressure at the focal plane with a hydrophone. However, highly focused, nonlinear HIFU beams can result in considerable spatial averaging artifacts due to finite area of the hydrophone active element. Spatiotemporal deconvolution (STD) can correct for hydrophone spatial averaging (Wear, *IEEE Trans. UFFC* **69**, 1243–1256 (2021)]. The objective of the present work is to extend validation of STD from diagnostic levels (6 MPa) to therapeutic levels (49 MPa). Three HIFU transducers (1.45 MHz F/1, 1.53 MHz F/1.5, and 3.91 MHz F/1) were driven in tone bursts (Agilent HP3314A function generator, ENI 240L amplifier) to generate HIFU fields in a degassed, deionized water tank with focal compressional pressures of 49, 16, and 29 MPa. Fields were measured in focal planes with two hydrophones (Onda HFO 100-μm fiber-optic; Onda HNA-0400 400-μm robust needle). STD reduced inter-hydrophone differences from $-18 \pm 11\%$ to $0 \pm 8\%$ (peak compressional pressure), $-8 \pm 4\%$ to $-5 \pm 5\%$ (peak rarefactional pressure), $-20\% \pm 7\%$ to $-3 \pm 8\%$ (pulse intensity integral), and $21 \pm 12\%$ to $-2 \pm 3\%$ (beam FWHM). STD reduces dependence of measurements on hydrophone active element size. STD reduces reliance on fiber optic hydrophones, which can be expensive, difficult to use, and prone to radiation-force-induced sensor misalignment.

8:15

5aBAa2. Absolute temperature estimation using thermal strain imaging. Omar Gachouch (LabTAU, INSERM, Ctr. Léon Bérard, F-69003, 151 Cr Albert Thomas, Lyon, Rhone Alpes 69003, France, omar.gachouch@inserm.fr), Bruno Giammarinaro, Teymour Kangot, Caterina Monini, and Rémi Souchon (LabTAU, INSERM, Ctr. Léon Bérard, F-69003 Lyon, France)

The objective of this study is to present a new thermometry method, based on thermal strain imaging, to estimate the absolute temperature using ultrasound. Unlike the conventional method presented in previous studies, this method does not require knowledge of the initial temperature or of speed of sound. The method was tested in simulations and experimentally. Simulations were performed at different temperatures (from 20 to 90 °C), using k-wave. Then, experimental measurements were performed in Zerdine phantom samples at different temperature (from 23 to 88 °C). A linear array transducer, with a 12-angle plane wave imaging sequence, was used to acquire the thermal strain images and to derive the temperature at the focus. To control the temperature during the measurements, a thermocouple was used. Both simulations and experiments showed a good estimation of the temperatures. The new ultrasound-based method is a promising technique to estimate and monitor temperature during thermal ablation.

8:30

5aBAa3. Using different methods to measure ultrasonic attenuation in cortical bone: A comparison. Brett A. McCandless (Mech. and Aerosp. Eng., North Carolina State Univ., 1840 Entrepreneur Dr., Raleigh, NC 27614, bamccand@ncsu.edu), Kay Raum (Ctr. for Biomedicine, Charité-Universitätsmedizin, Berlin, Germany), and Marie Muller (MAE, North Carolina State Univ., Raleigh, NC)

The assessment of bone mineral density (BMD) and bone microstructure is important in screening for various bone diseases. Dual x-ray absorptiometry is the most commonly used technique for evaluating BMD; however, alternatives are necessary due to the use of ionizing radiation and low resolution, the latter of which prevents obtaining information on bone microstructure. Quantitative ultrasound presents a potentially attractive alternative. The microstructure of porous media, such as cortical bone, influences the attenuation of ultrasound; as such, ultrasound attenuation can be used to obtain information about the microstructure of cortical bone. Different models and measurement methods may be used to estimate ultrasonic attenuation in a cortical bone. In this study, three different methods of measuring ultrasonic attenuation were used. Finite-difference time-domain simulations were conducted in maps mimicking cortical bone. Pore densities and pore size distributions in the bone matrix, as well as absorptive properties of the bone and porous matrices, were specified for all simulations. Attenuation was measured within the pulse bandwidth as a function of frequency using the independent scattering approximation (ISA), a cortical backscatter method (CortBS), and the reflections from the two surfaces of the cortical bone map. Excellent agreement was found between these three methods.

8:45

5aBAa4. Simulation of high frame rate spread-spectrum color Doppler imaging of pulsatile flow. Kian Esmailian (Biomedical Eng., Western Univ., 100 Perth Dr., London, ON N6A 5K8, Canada, kesmaili@uwo.ca) and James Lacefield (Biomedical Eng., Western Univ., London, ON, Canada)

Spread-spectrum Doppler, a method introduced by our lab, preserves the maximum unaliased velocity of ultrafast Doppler while retaining some of the image quality benefits of compounding plane waves transmitted at different angles. The technique employs a sequence of pulses transmitted at M angles that are repeated L times in different random orders. Shuffling the slow-time samples so the angles repeat in the ascending order concentrates echoes from stationary off-focus targets in M harmonic frequency bins while spreading the in-focus signal across all frequencies. Off-focus echoes are suppressed, without compounding, by applying a notching comb filter, while the portion of the in-focus signal spread to the other $(L-1)M$ bins is retained for velocity estimation. Field II simulations were used to assess the method's ability to track pulsatile velocity fluctuations in a straight vessel. The angle-corrected peak velocity was accurate to within $\pm 10\%$ of the true value when imaging at a Doppler frame rate of approximately 115 frames per cardiac cycle. Further improvement of the method will require a filter to attenuate off-focus echoes from non-stationary tissue.

9:00

5aBAa5. An MR-compatible fiber-optic probe for measuring focused ultrasound-induced temperature rises without viscous heating artifacts. Sara L. Johnson (Dept. of Radiology and Imaging Sci., Univ. of Utah, 804 E 300 S, Apt. 25, Salt Lake City, UT 84102, sarajay144@gmail.com), Henrik Odeen, Allison Payne (Dept. of Radiology and Imaging Sci., Univ. of Utah, Salt Lake City, UT), and Harry Vine (OSENSA Innovations, Caldwell, NJ)

Focused ultrasound (FUS) beam interactions with thermocouples and fiber optic (FO) probes cause a "viscous heating artifact" (VHA), which prevent accurate temperature measurements during acoustic sonication. This work demonstrates a novel fiber-optic temperature probe which is insensitive to VHA, validated in an MR-guided FUS treatment setting. The FO probe (OSENSA Innovations; PRB-140, 0.14 mm diameter) was inserted with an 18G catheter into a tissue-mimicking gelatin phantom. The FO probe tip was located with high-resolution MR images ($0.25 \times 0.25 \times 0.5\,\mathrm{mm}^3$) for targeting with FUS. Continuous-wave FUS sonications (50 W, 20 s) were delivered at a distance of $\sim$1.5 mm from the FO probe tip using a 256 phased-array transducer (Imasonic, France; 1 MHz, $2.1 \times 2.3 \times 9.8$ FWHM spot size) and FUS-induced heating was measured with 3D MR temperature imaging (MRTI; $0.5 \times 0.5 \times 1\,\mathrm{mm}^3$ resolution, 3.9 s acquisition). The VHA effect was not observed in the PRB-140 FO probe measurement data. The FO probe heating and cooling curves closely matched those measured by MRTI, with a root mean-squared error (RMSE) of 0.61 °C. In contrast, the RMSE of VHA-sensitive FO probe was 6.00 °C for a similar acoustic power output. This ultrasound artifact-immune FO temperature probe is highly advantageous for MR temperature sequence development and precise temperature monitoring in FUS treatment applications.

9:15

5aBAa6. A pipeline to generate large-scale, realistic cardiac ultrasound recordings including clinically relevant artefacts. Nitin Burman (Dept. of Cardiovascular Sci., Katholieke Universiteit Leuven, UZ Herestraat 49 - Box 7003, Leuven 3000, Belgium, nitin.burman@kuleuven.be), Sophie V. Heymans (Dept. of Cardiovascular Sci., Katholieke Universiteit Leuven, Kortrijk, Belgium), Claudia Manetti, Joost Lumens (Faculty of Health, Medicine and Life Sci., Maastricht Univ., Maastricht, Netherlands), and Jan D'hooge (Dept. of Cardiovascular Sci., Katholieke Universiteit Leuven, Leuven, Belgium)

Simulated ultrasound (US) data are widely used to develop and validate (machine learning-based) US data processing algorithms. In this regard, the quantity and quality of the simulated US data are crucial. Here, we have developed an US simulation pipeline to generate realistic cardiac US recordings on a large scale. In this pipeline, we used clinical cardiac US scans to sample the echogenicity of the US scattering sites. In parallel, a non-linear US simulator, k-wave, was employed to generate clinical artifacts due to the presence of ribs and lungs, including reverberation and shadowing. The position of the ventricle, the probe, and the simulated artifact data were then spatially registered in order to modify the originally sampled echogenicities. Motion of the myocardial scattering sites was kinematically governed by a stable mechanical heart model (CircAdapt). The resulting dynamic echogenicity map was fed into a fast convolution-based ultrasound simulator (COLE) to generate cardiac US recordings with clinical appearance including artefacts. The generated US data follow realistic speckle statistics and is a potential augmentation tool for machine learning based US data processing algorithms. [Work funded by European Union's Horizon 2020 research and innovation programme under the Marie Sklodowska-Curie grant agreement No. 860745.]

9:30

5aBAa7. Optimizing intracardiac flow assessment with high frequency ultrasound in a mouse model. Geraldi Wahyulaksana (Radiology, Weill Cornell Medicine, 416 E 55th St., New York, NY 10022, gew4002@med.cornell.edu), Colin K. L. Phoon (Div. of Pediatric Cardiology, 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 (Radiology, Weill Cornell Medicine, New York, NY)

Cardiomyopathy, a disorder affecting the heart muscle, is associated with severe symptoms, such as heart failure, arrhythmia, or cardiac arrest. Echocardiography is the primary clinical tool for myocardial assessment, which provides diagnostic and prognostic insights. However, current clinical parameters, such as myocardial strain, often detect the condition only after symptoms become pronounced, showing the need for early-stage assessment. Based on preliminary studies, intracardiac flow patterns have the potential to detect abnormal myocardial changes earlier than the current clinical parameters. Mouse models are commonly imaged with ultrasound in the cardiovascular disease research but their use for flow pattern evaluation has been hindered by spatial and temporal resolution limitations. Here, we address this limitation by conducting data acquisition using a Verasonics scanner with a range of high-frequency probes (16 to 42 MHz), transmit cycles, and transmit angles. A singular value decomposition (SVD) filter and frequency de-aliasing were also investigated to optimize flow quantification. Our findings show that a higher transmission frequency, while providing superior spatial resolution, may not yield the best flow quantification due to increased Doppler aliasing. Furthermore, SVD filtering and Doppler de-aliasing should be used with caution because improper implementation can adversely impact the quantification results.

Session 5aBAb

Biomedical Acoustics: General Topics in Biomedical Acoustics: Drug Delivery and Therapy

Anurag N. Paranjape, Cochair
UPMC Heart and Vascular Inst., Univ. of Pittsburgh, 3550 Terrace St., 960 Scaife Hall, Pittsburgh, PA 15213

Virginie Papadopoulou, Cochair
Biomedical Engineering, The Univ. of North Carolina at Chapel Hill, 116 Manning Dr., 9004 Mary Ellen Jones Building, CB 7575, Chapel Hill, NC 27599-7575

Contributed Papers

8:00

5aBAb1. Ultrasound-triggered *in situ* hydrogel formation for spinal disc repair. Veerle A. Brans (Inst. of Biomedical Eng., Univ. of Oxford, Inst. of Biomedical Eng. (Botnar Res. Centre), Old Rd., Oxford OX3 7LD, United Kingdom, veerle.brans@eng.ox.ac.uk), Anna P. Constantinou (Dept. of Mater., Imperial College London, London, United Kingdom), Matthew J. Kibble, Nicolas Newell (Dept. of Bioengineering, Imperial College London, London, United Kingdom), Luca Bau (Inst. of Biomedical Eng., Univ. of Oxford, Oxford, United Kingdom), Molly M. Stevens (Dept. of Mater., Imperial College London, London, United Kingdom), Constantin Coussios (Inst. of Biomedical Eng., Univ. of Oxford, Headington, Oxford, Oxfordshire, United Kingdom), and Michael Gray (Inst. of Biomedical Eng., Univ. of Oxford, Oxford, United Kingdom)

Over 600 million people suffer from lower back pain attributable to spinal disc degeneration. Treatments range from conservative physiotherapy to costly, invasive options like spinal fusion surgery. Injectable engineered materials may allow minimally invasive disc repair, but self-curing lacks process control, and optical *in situ* curing is depth-limited. We, therefore, propose ultrasound-triggered implant formation, enabling spatiotemporal control at clinically relevant tissue depths. We developed an anionic polysaccharide-based hydrogel, seeded with calcium-loaded thermosensitive liposomes. Focused ultrasound was used to heat this injectable precursor material to just above the lipid phase-transition temperature of 41 °C, inducing calcium release and ionically crosslinked network formation. Controlled heating was achieved by elevating the acoustic attenuation of the precursor solution with purified glass microspheres. The heating and gelation processes were controlled in real-time using thermometry and acoustic cavitation emissions. We optimized the ultrasound frequency and pressure amplitude to provide controlled heating with minimal cavitation. After these parameters were employed for ultrasound-mediated gelation, the rheological properties of the resultant gels were compared to literature values for native spinal disc material. Finally, *in situ* gel formation was evaluated in *ex vivo* bovine tail discs, from injection to mechanical assessments, confirming the ability to remotely trigger injectable disc-mimicking materials.

8:15

5aBAb2. Precise, low-intensity triggered release in biological systems using ultrasound-responsive antibubbles. Athanasios Athanassiadis (Heidelberg Univ., Im Neuenheimer Feld 225, Heidelberg 69120, Germany, thanasi@uni-heidelberg.de), Nicolas Moreno-Gomez, Dimitris Missirlis, Yannik L. Trautnitz, and Peer Fischer (Heidelberg Univ., Heidelberg, Germany)

Antibubbles are an emerging carrier for low-intensity, ultrasound-triggered release that can carry large payloads in their liquid cores. As we have previously shown, antibubbles can release their payloads at adjustable pressures, ranging from as low as a few kPa (MI < 0.01) to above a few hundred kPa (MI > 0.2). The specific release threshold and behavior can be determined during fabrication, making it possible to realize either single-shot release or multi-stage dosing of a payload. These characteristics make antibubbles an attractive alternative to conventional microbubble delivery agents particularly in sensitive tissues where risks of ultrasound need to be kept to a minimum. In this talk, we discuss recent experiments in which we demonstrate triggered release from antibubbles in *in vitro* biological systems. We show that payload release can be controlled using patterned or focused ultrasound fields, and we characterize the distribution and cellular uptake of payloads inside cell-laden matrices. These results not only demonstrate the potential of antibubbles for therapeutic applications but also open the door to targeted delivery in complex tissue scaffolds for tissue engineering.

8:30

5aBAb3. Targeted delivery of miR-1 to the heart using clinical contrast ultrasound. Davindra Singh (Biology, Concordia Univ., 1455 de Maisonneuve Blvd., Montreal, QC H3G 1M8, Canada, davindra.singh@mail.concordia.ca), Stephanie He (Biology, Concordia Univ., Montreal, QC, Canada), and Brandon Helfield (Phys., Concordia Univ., Montreal, QC, Canada)

Pathological left-ventricle hypertrophy is a cardiovascular disorder resulting in the thickening of the ventricle wall due to abnormal cardiomyocyte growth. The downregulation of miR-1 in hypertrophic cardiomyocytes has been identified as an early disease marker, with the delivery of miR-1 as a promising treatment strategy. Ultrasound and microbubbles offer an exciting approach to image-guided and site-specific cardiac gene delivery. The objective of this study is to show the feasibility of viable ultrasound and microbubble-mediated delivery of miR-1 *in vivo*. Sprague–Dawley rats were injected via tail vein with a suspension of miR-1 mimic (0.6 mg/kg) and Definity. The rats were then treated with a high MI (1.34) flash sequence for 20 min using a C5-2 probe with a Phillips iU22. Delivery was confirmed on isolated heart tissue with RT-qPCR and Western blots for miR-1 and protein expression, respectively. miR-1 delivery in healthy male rats resulted in a 1.33-fold ($p = 0.05$) increase in miR-1 as compared to sham controls. This resulted in a decrease in hypertrophic protein expression (1.18-fold in TWF1, $p = 0.17$; 1.23-fold in MEF2A, $p = 0.07$; 1.25-fold in CX43, $p = 0.03$). Our data demonstrate the feasibility of using ultrasound and microbubbles as an image-guided delivery method for molecular therapeutics in cardiovascular disorders.

8:45

5aBAb4. Monodisperse microbubble-mediated drug delivery: Influence of microbubbles size on drug delivery outcome. Yuchen Wang (Erasmus MC, Dr. Molewaterplein 40, Rotterdam 3015 GD, Netherlands, y.wang@erasmusmc.nl), Hongchen Li (Biomedical Eng., Erasmus MC, Rotterdam, Netherlands), Bram Meijlink (Biomedical Eng., Erasmus MC, Utrecht, Utrecht, Netherlands), Jiali Luo, Robert Beurskens (Biomedical Eng., Erasmus MC, Rotterdam, Netherlands), Benjamin Johnson (Univ. of Leeds, Leeds, United Kingdom), and Klazina Kooiman (Erasmus MC, Rotterdam, Netherlands)

As the microbubble's resonance frequency is size-dependent, polydisperse microbubbles induce varying drug delivery outcomes at one ultrasound frequency. This study aimed to investigate drug delivery outcome of monodisperse MBs (mMBs) with radii ranging from 1.5–2.9 μm insonified at 2 MHz. Phospholipid-coated mMBs were generated using the Horizon microfluidic flow-focusing device. *In vitro* experiments were conducted on single microbubble-endothelial cells (n = 68) using confocal microscopy and 10 Mfps ultra-high-speed imaging. At 220 kPa PNP for 10 cycles, the 1.5 μm mMBs exhibited the highest PI uptake in 81.3% of cases, followed by 77.3% for 2.2 μm, 36.8% for 2.7 μm and 13.3% for 2.9 μm mMBs. Conversely, the 1.5 μm mMBs had the second lowest excursion amplitude ($R_{;max}$-$R_{;0}$) of 0.8 ± 0.3 μm as this was 1.2 ± 0.3 μm for 2.2 μm mMBs, 1.0 ± 0.3 μm for 2.7 μm mMBs, and 0.7 ± 0.2 μm for 2.9 μm mMBs. Additionally, for the 1.5, 2.2, and 2.7 μm mMBs, PI uptake and tunnel formation occurred more often (1.6, 1.8, and 1.3 times, respectively) than PI uptake and a resealing membrane pore. During insonification, mMB pinch-off occurred more frequently in tunnel formation (70.8%) than resealing pore formation (53.3%). This research revealed the impact of mMBs size on drug delivery outcome.

9:00

5aBAb5. Focused ultrasound-guided delivery of gene editing protein in human induced pluripotent stem cells. Kyle Hazel (Biology, Concordia Univ., 7141 Sherbrooke St W, Montreal, QC H4B 1R6, Canada, kyle.hazel@hotmail.com), Davindra Singh, Mathieu Husser, Elahe Memari, Stephanie He (Biology, Concordia Univ., Montreal, QC, Canada), and Brandon Helfield (Phys., Concordia Univ., Montreal, QC, Canada)

Focused ultrasound (FUS) in combination with microbubbles (MBs) can induce cavitation-mediated plasma membrane permeabilization in nearby cells, thus permitting entry to otherwise impermeable macromolecules. Cas9 is an endonuclease protein currently at the forefront of gene editing due to its efficiency, ease of use, and low cost. In complex with single guide RNA (sgRNA), Cas9 can target specific gene sequences to cause double-stranded breaks, interrupting gene function in the process. Cas9 is best delivered as a ribonucleoprotein (RNP) for the most effective results, however, suffers from inefficient delivery methods for *in-vivo* applications due to its large size (160 kDa). FUS and microbubbles can be an effective alternative to currently applied systems (e.g., adeno-associated vectors) to deliver Cas9:sgRNA RNPs for CRISPR-mediated knockout. Currently, we are exploring the use of FUS for Cas9-mediated knockout of EGFP in both EGFP-expressing human induced pluripotent stem cells (hiPSC) and human cardiomyocytes. Treatment of hiPSC under acoustic conditions (1 MHz, 1000 cycles, 5 ms intervals, >208 kPa) suitable for cavitation-mediated sonoporation led to EGFP knockout in hiPSCs. By modulating FUS parameters and setup, we can optimize delivery of Cas9 as an RNP for treatment of genetic diseases, such as hypertrophic cardiomyopathy.

9:15

5aBAb6. Using ultrasound-targeted microbubble cavitation to open the blood–brain barrier for drug delivery in Alzheimer's disease. Grace E. Conway (Univ. of Pittsburgh, 3550 Terrace St., Scaife Hall 969.3, Pittsburgh, PA 15213, gec36@pitt.edu), Xucai Chen, Stacey J. Rizzo, Afonso C. Silva (Univ. of Pittsburgh, Pittsburgh, PA), and Flordeliza S. Villanueva (Medicine/Cardiology, Univ. of Pittsburgh, Pittsburgh, PA)

Drug delivery for Alzheimer's disease (AD) is challenging due to restricted diffusion across the blood–brain barrier (BBB). Ultrasound-targeted microbubble cavitation (UTMC) can be used to transiently open the BBB. We hypothesized that opening the BBB with UTMC would increase the brain concentration of LY2886721 (LY), an orally administered drug developed for AD, in 5XFAD mice, a model of AD. We treated the right hemisphere using an RK-50 system (FUS Instruments Inc). Definity microbubbles were injected, and ultrasound (1.45 MHz, 2 Hz pulse repetition frequency, 10 ms pulse length, 0.6 MPa for 30 s) was applied to 11 locations across the right hemisphere. Thirty minutes after UTMC, mice were administered LY (30 mg/kg p.o.). In one cohort, 45 min after dosing with LY, there was an increase in the concentration of LY in the UTMC-treated hemisphere compared to the non-UTMC treated hemisphere ($p < 0.05$). In a second cohort, 2.5 h after dosing with LY, there was a decrease in soluble $A\beta_{;40}$ ($p < 0.05$), insoluble $A\beta_{;40}$ ($p < 0.05$), and insoluble $A\beta_{;42}$ ($p < 0.05$) in the UTMC-treated hemisphere compared to the non-UTMC treated hemisphere. UTMC can be used to open the BBB to increase the concentration of LY and decrease $A\beta$ in a preclinical model of AD.

9:30

5aBAb7. The activation of endothelial nitric oxide synthase, induced by calcium influx plays a crucial role in regulating endothelial hyperpermeability caused by ultrasound-targeted microbubble cavitation. Anurag N. Paranjape (Medicine/Cardiology, Univ. of Pittsburgh, 3550 Terrace St., Pittsburgh, PA 15217, anuragnparanjape@gmail.com), Xucai Chen, and Flordeliza S. Villanueva (Medicine/Cardiology, Univ. of Pittsburgh, Pittsburgh, PA)

Ultrasound-targeted microbubble cavitation (UTMC) plays a crucial role in improving drug delivery across the endothelial barrier. For successful application of UTMC in clinic, an understanding of molecular mechanisms involved is essential. Here, we hypothesized that Ca^{2+} and endothelial nitric oxide synthase (eNOS) pathways regulate UTMC-induced endothelial hyperpermeability. We used human coronary artery endothelial cells seeded on transwell inserts, exposed to microbubbles and ultrasound (frequency 1 MHz; PNP 250 kPa; pulse length 10 μs; pulse interval 10 ms; treatment duration 10 s). UTMC caused cellular influx of Ca^{2+}, which was inhibited by blocking mechanosensitive channels ($p < 0.0001$) using GsMTx4. Knockdown of Piezo1 using siRNA showed similar effects. UTMC-induced Ca^{2+} influx activated eNOS and enhanced nitric oxide production, which was necessary for UTMC-induced hyperpermeability. Our data suggest that UTMC induces a Ca^{2+} influx-dependent increase in S-nitrosylation: An increase in S-nitrosylation was observed on both β-catenin and VE-cadherin after UTMC, potentially contributing to the destabilization of adherens junctions that normally maintain barrier integrity. Our study explains the role of Ca^{2+} influx, eNOS activation, and enhanced S-nitrosylation of adherens junction proteins in the regulation of endothelial hyperpermeability. Further investigation of these pathways will aid in clinical translation and optimization of UTMC for delivering cell-impermeant drugs.

9:45–10:00 Break

10:00

5aBAb8. Towards real-time decompression sickness mitigation using wearable capacitive micromachined ultrasonic transducer arrays. Joshua B. Currens (Joint Dept. of Biomedical Eng., UNC - Chapel Hill, 10010 Mary Ellen Jones, Campus Box 7575, Chapel Hill, NC 27599, jcurrens@unc.edu), Muhammetgeldi Annayev, Remzi Erkan Kemal (Elec. and Comput. Eng., North Carolina State Univ., Raleigh, NC), Katherine M. Eltz, Arian Azarang (Joint Dept. of Biomedical Eng., UNC - Chapel Hill, Chapel Hill, NC), Michael Natoli, Rachel M. Lance, Richard E. Moon (Ctr. for Hyperbaric Medicine and Environ. Physiol., Duke Univ., Durham, NC), Paul A. Dayton (Joint Dept. of Biomedical Eng., UNC - Chapel Hill, Chapel Hill, NC), Feysel Yalcin Yamaner (Clearsens Inc., Durham, NC), Omer Oralkan (Elec. and Comput. Eng., North Carolina State Univ., Raleigh, NC), and Virginie Papadopoulou (Joint Dept. of Biomedical Eng., UNC - Chapel Hill, Chapel Hill, NC)

Decompression sickness (DCS) due to inert gas supersaturation remains one of the major risks for scuba divers and can occur despite adherence to prevention schedules for staged decompression. Post-dive echocardiography for venous gas emboli (VGE) detection has low sensitivity for DCS outcome and is unable to provide real-time physiological monitoring underwater. Alternatively, we present progress towards collecting ultrasound data while at pressure and an exploration into a quantitative assessment for

decompression stress. Ultrasound data of an imaging phantom were collected in a hyperbaric chamber up to 9 ATA using a Verasonics V1 system and custom capacitive micromachined ultrasonic transducer (CMUT). The DC voltage requirement for CMUT operation decreased as ambient pressure increased. Separately, a mouse model was used to simulate an extreme pressure profile and echocardiograms were collected every 20 min over 2-h post-decompression. Towards enhancement of DCS assessment, a quantitative analysis of the murine echocardiograms was implemented. Preliminary results show an increase in signal intensity within the venous blood from pre- to post-dive, indicating potential gas presence despite VGE absence. Our findings demonstrate the ability to obtain ultrasound data at pressure and a potential continuous assessment method, which may provide a practical direction for real-time decompression stress quantification.

10:15

5aBAb9. Designing a benign prostatic hyperplasia dual-mode cavitation cloud and boiling histotripsy therapy transducer. Yashwanth Nanda Kumar (Ctr. for Industrial and Medical Ultrasound, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, ynandaku@uw.edu), Kaizer Contreras, Yak-Nam Wang, Wayne Kreider, Stephanie Totten (Ctr. for Industrial and Medical Ultrasound, Univ. of Washington, Seattle, WA), George R. Schade (Dept. of Urology, Univ. of Washington, Seattle, WA), and Adam D. Maxwell (Ctr. for Industrial and Medical Ultrasound, Univ. of Washington, Seattle, WA)

Benign prostatic hyperplasia (BPH) is a condition that causes an enlarged prostate leading to lower urinary tract symptoms (LUTS), affecting quality of life. Histotripsy is a non-invasive focused ultrasound technique that mechanically disintegrates tissue wherein cavitation cloud (CH) and boiling histotripsy (BH) are the two commonly used therapy modes. Recent studies have shown successful ablation of BPH tissue *in vitro* by transducers that were designed for a transabdominal approach. However, this approach may not be successful in treating tissue *in vivo*, as evidenced in some of the clinical studies, due to challenges with the anatomical position of prostate within humans. Therefore, a need exists in treating it transrectally with a transducer that can perform both CH and BH for efficient ablation. Initial design studies were performed to determine the appropriate source parameters and materials needed for fabricating the combinational therapy transducer for *in vivo* use. A 3 MHz multi-element source with a focal length of 40 mm and gain of 64 was chosen. The equivalent aperture is 48.5 mm with a 10 mm central opening for image guidance. Non-linear simulations show sufficient peak positive and negative pressures can be achieved for performing both regimes. Further details on the fabrication and characterization of the transducer with benchtop results will be presented.

10:30

5aBAb10. Correlation of *Escherichia coli* inactivation with histotripsy bubble cloud size. Pratik A. Ambekar (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, pambek@uw.edu), Tatiana D. Khokhlova (Dept. of Medicine, Univ. of Washington, Seattle, WA), Yak-Nam Wang (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Pavel B. Rosnitskiy (Appl. Phys. Lab., Univ. of Washington, Moscow, Russian Federation), Daniel Leotta, Gilles P. Thomas, Stephanie Totten, Shelby Pierson, Matthew Bruce (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Adam D. Maxwell (Urology, Univ. of Washington, Seattle, WA), Keith Chan (Vantage Radiology and Diagnostic Services, Seattle, WA), W. Conrad Liles (Dept. of Medicine, Univ. of Washington, Seattle, WA), Evan P. Dellinger (Dept. of Surgery, Univ. of Washington, Seattle, WA), Adeyinka Adedipe (Dept. of Emergency Medicine, Univ. of Washington, Seattle, WA), Wayne Monsky (Dept. of Radiology, Univ. of Washington, Seattle, WA), and Thomas Matula (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Bacterial loads can be effectively reduced with cavitation-mediated focused ultrasound, or histotripsy. Our previous *in vitro* work with *Escherichia coli* (*E. coli*) established strong trends of bactericide with increasing peak negative pressure amplitude, pulse length, PRF, and treatment time. The current study correlates bactericide as a function of the histotripsy bubble cloud size produced for these treatment parameters at several frequencies. Histotripsy was applied to *E. coli* suspensions in 10-ml sample vials at 810 kHz, 1.2 MHz, or 3.25 MHz for 40 min. Separately, cavitation was recorded using a Photron Fastrax high-speed camera for acoustic parameters equivalent to those used in the *E. coli* studies. The images were used to quantify the maximum size of each bubble cloud, with the assumption that the cloud was axially symmetric around the propagation direction. A strong linear relationship exists between log kill versus cloud size ($R^2 = 0.96$). Remarkably, across all variables studied, the log-reduction in viable bacteria exhibited a direct proportional relationship with the dimensions of the bubble cloud. This strong correlation between bacterial reduction and cavitation bubble cloud size could have significance for clinical applications where bacteria cannot easily be sampled. [Work supported by NIH R01AR080120 and R01EB023910.]

10:45

5aBAb11. Ultrasound diagnosis and treatment of heterotopic ossification. Fea Morgan-Curtis, Lucas Ruge-Jones, Grace M. Wood (Graduate Program in Acoust., Penn State, University Park, PA), Lisa Berntsen (Dept. of Biomedical Eng., Penn State, University Park, PA), Jacob C. Elliott (Graduate Program in Acoust., Penn State, Res. West, State College, PA 16801, jce29@psu.edu), Daniel Hayes (Dept. of Biomedical Eng., Penn State, University Park, PA), and Julianna C. Simon (Graduate Program in Acoust., Penn State, University Park, PA)

Heterotopic ossification (HO), or the presence of bone in soft tissues, can occur after musculoskeletal trauma, causing pain and reduced mobility. However, even the most sensitive diagnostic modality requires 2–3 weeks after initiation to detect HO, and the only treatment is surgical resection after HO matures (>2 years). Here, we evaluate the color Doppler ultrasound twinkling artifact for early diagnosis of HO and focused ultrasound (fUS) for treatment of early HO. We began by evaluating twinkling and fUS parameters in ossified cell culture. We then evaluated imaging and treatment parameters in mice with HO. Results show that twinkling correlated well with the presence of ossified cells; although the exact size/number of ossified cells required for twinkling was unclear. In mice, twinkling was found to detect HO in some mice as early as 3 days-post-injection, which is earlier than other imaging modalities. For treatment, fUS at 1.07-MHz and 0.2% duty-cycle with $p+/p- = 17/7$ MPa was found to be most successful at disrupting mineralizations with little damage to surrounding cells. Mice treated with fUS were found to have less advanced HO compared to sham mice. These results highlight the promise of ultrasound for the diagnosis and treatment of early HO. [Work supported by CDMRP PR201164].

11:00

5aBAb12. The roles of pulse length and duty cycle in the fractionation of tendinopathic tendons. Grace M. Wood (Graduate Program in Acoust., Penn State, 1 Old Main, University Park, PA 16802, gmw5253@psu.edu), Jacob C. Elliott, and Julianna C. Simon (Graduate Program in Acoust., Penn State, University Park, PA)

Collagenous tissues, like tendon, are resistant to fractionation by focused ultrasound (fUS). Prior work has shown thermal pretreatments increase susceptibility to fractionation by fUS, suggesting the potential for successful fractionation by tuning boiling histotripsy parameters. Here, we investigate the roles of pulse length (PL) and duty cycle (DC) on fUS treatments of tendinopathic tendons. *Ex vivo* bovine tendons were injected with collagenase to induce tendinopathy. The following day, tendons were treated for 15–20 min with 1–3 MHz fUS ($p_{;+} \leq 127$ MPa/$p_{;-} \leq 35$ MPa). Three PLs were chosen below, slightly above, and well above the calculated time-to-boil. The pulse repetition frequency varied between 0.2–1 Hz, allowing for evaluation of DCs between 0.1% and 0.8%. Preliminary results show focal fiber separation at the PL slightly above the calculated time-to-boil; areas of tissue disruption are smaller and less frequent for the other PLs and accompanied by thermal damage at the highest PL. As frequency increases, the tissue disruption becomes smaller and infrequent for the lowest and highest PLs. When the PL was held constant, minimal change in tissue fractionation was found when changing DC. Thus, for the tested combinations, PL is more influential than DC for tuning the fractionation of tendinopathic tendons. [Work supported by NIH RO1EB032860.]

11:15

5aBAb13. Tendon as a model for testing the efficacy of histotripsy for chronic deep vein thrombosis. Kevin Zhao (Radiology, Univ. of Chicago, 5841 South Maryland Ave., Chicago, IL 60637, kzhao@som.geisinger.edu), Erik Saucedo, and Kenneth B. Bader (Radiology, Univ. of Chicago, Chicago, IL)

Histotripsy is a focused ultrasound therapy under development for multiple diseases, including deep vein thrombosis (DVT). Benchtop studies to gauge the efficacy of histotripsy for chronic DVT present a challenge due to pathologic changes in composition that are difficult to replicate *in vitro*. Tendon is a readily available tissue with an extensive extracellular matrix similar to chronic DVT. This study aims to assess histotripsy-induced changes to tendon at driving parameters effective for acute thrombus ablation. Porcine patellar tendons were bisected into control ($N = 19$) and treatment ($N = 21$) segments. Histotripsy pulses were applied for 1 to 20 min at rates of 250 or 500 Hz. The pulse duration (6.7 μs) and peak negative pressure (35 MPa) were previously shown to be effective for acute thrombus ablation. Treated specimens exhibited gross swelling in targeted regions. Manual segmentation of ultrasound images was used to estimate the area of tissue affected by histotripsy, which was found to expand by up to 30% over the exposure durations for both pulsing rates. Histological analysis revealed loss of linear fiber organization and liquefaction within treated regions. Overall, these data indicate histotripsy effectively disrupts tendon integrity, making it a potential tool for the management of chronic DVT.

11:30

5aBAb14. Active targeting of nanotherapeutics using power cavitation imaging with a linear array transducer. Kamso Onyemeh (Biomedical Imaging Program, Weill Cornell Graduate School of Medical Sci., 1300 York Ave., New York, NY 10065, kao4006@med.cornell.edu), Mark Burgess (Dept. of Medical Phys., Memorial Sloan Kettering Cancer Ctr., New York, NY), Raashed Raziuddin (Pharmacology Program, Weill Cornell Graduate School of Medical Sci., New York, NY), and Daniel Heller (Molecular Pharmacology Program, Memorial Sloan Kettering Cancer Ctr., New York, NY)

Power cavitation imaging (PCI) is an emerging strategy for monitoring blood–brain barrier (BBB) opening, enabling spatial discrimination of cavitation intensity through power Doppler-analogous processing of passive cavitation images. Single-element cavitation detectors, while beneficial in signal monitoring for cavitation controllers, provide limited spatial information, conferring the need for an image-guided approach to ensure accurate cavitation localization and regulation. This study aims to evaluate the capability of PCI to spatially correlate real-time acoustic cavitation emissions with mechanical bioeffects as a predictor of P-selectin-targeted nanocarrier delivery. Preliminary sonoporation experiments were performed *in vitro* with brain tissue derived mouse endothelial cells and 3 kDa tetramethylrhodamine (TMR)-dextran. A focused ultrasound transducer (0.5 MHz, 5000 cycles/pulse, 5 Hz PRF, 30 second treatment duration) was used to sonicate cells in small volume (25 μl) cell suspension. Homogenous distribution of TMR-dextran in the cytosol and nucleus was observed via fluorescence microscopy at lower pressures ($7\% \pm 2\%$ fluorescent cells), while higher pressures had no significant differences above control ($3\% \pm 2\%$ vs 1% nontreated) due to microbubble destruction. Future experiments will correlate sonoporation with P-selectin expression and validate a linear array PCI system against contrast-enhanced MRI as a guide for P-selectin-targeted nanocarrier delivery in mouse models of medulloblastoma.

Session 5aNS

Noise, Structural Acoustics, and Vibration and Engineering Acoustics: Pickleball Noise

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

Chair's Introduction—8:00

Contributed Papers

8:05

5aNS1. Is pickleball a good neighbor? Evelyn Way (Maxxon Corp., 920 Hamel Rd., Hamel, MN 55340, eway@maxxon.com)

Upscale multi-family residences need the latest amenities to attract tenants. A recent project featured a party room, exercise facility, and pickleball court on the top floor of a low-rise, adaptive reuse of an office building to residential renovation. Pickleball courts located directly above residential units allowed for measurements of sound transmission during pickleball play and comparison to traditional airborne and impact isolation metrics.

8:20

5aNS2. Preliminary analysis of more than 60 pickleball noise consultant reports. Charles E. Leahy (Law Office - Pro Bono Public Interest, 151 E Summit St., Harbor Springs, MI 49740, charles.leahy@sbcglobal.net)

Over 60 consulting reports authored by over 30 different US and Canadian consultants have been gathered from municipal agendas, court filings, and HOA members. The reports are analyzed with respect to tools, metrics, and criteria used to make findings and recommendations to city officials, HOA Boards, and lawsuit litigants. The reports range greatly in length and purpose, many with sound meter data, excerpts from municipal ordinances, and specifications for noise mitigation including setbacks, full enclosures, barriers, and equipment recommendations. Particular attention is given to the consultant's choice of sound meter metrics and the attention or lack of attention to the impulsive nature of pickleball noise and applicability of ANSI Standard 12.9 Part 4. When a local ordinance is referenced, we examine whether the consultant limited the report to provisions specifying a decibel limit and excluded comment on the general nuisance and plainly audible standards that are often present. Anecdotes are drawn from the data to identify best practices and worst practices of the consultants and their reports. This paper may be useful to consultants in planning their future pickleball studies; useful in deciding whether to refer these projects to a more experienced pickleball noise boutique firm; and/or useful to attorneys in recognizing the challenges of using the consultant report in the litigation setting.

8:35

5aNS3. Improving the persuasiveness of the noise consultant report—A critique and proposal. Charles E. Leahy (Law Office - Pro Bono Public Interest, 151 E Summit St., Harbor Springs, MI 49740, charles.leahy@sbcglobal.net)

The author, a retired mechanical engineer and patent attorney, sat on an HOA Board that converted four tennis courts to 13 pickleball courts without installing the noise absorbing barrier recommended in three reports by two acoustical consultants. A nuisance lawsuit seeks an injunction to close the courts and money damages. Consultants study the pickleball noise and recommend mitigation options to the deciders. Deciders are laymen, such as city officials and HOA boards, unfamiliar with noise science and under pressure to meet the insatiable need for more pickleball courts. When mitigation is not implemented, or not successful, litigation ensues. Judge and jury become deciders. This paper starts from the premise that the consultant makes technically competent recommendations and is motivated to fully educate the lay deciders. We explore the communication gap between consultants and deciders. Proposals include embedding wav. noise files in the written report, reference to ANSI standards for highly impulsive noises, deference to the lack of research on health issues of long-term exposure to persistent repetitive impulsive noise. We introduce the missing metrics of "pickle pops per hour" and the "predictable worst case noise impact" occurring when intensively used courts are introduced into residential neighborhoods.

8:50

5aNS4. Challanges in enacting pickleball noise regulations. Dana S. Hougland (Shen Milsom & Wilke, LLC, 1801 Wewatta, Fl. 11, Denver, CO 80202, dhougland@smwllc.com) and Tammy Maurer (City Council, City of Centennial, Centennial, CO)

Outdoor pickleball courts installations vary widely in environmental conditions, including terrain and construction. With the rapid rise in popularity of pickleball, new court installations and conversions of existing tennis courts have generated community pressure for local governments to enact noise restrictions. Local governmental agencies have been challenged to establish regulations, which must apply across a number of unique installation conditions and terrains. This presentation examines the challenges by a selection of local governmental agencies to enact and enforce regulations.

9:05

5aNS5. Categorization of pickleball paddles as a highly impulsive sound source. Lance Willis (Spendiarian & Willis, Tucson, AZ) and Charles E. Leahy (Law Office - Pro Bono - Public Service, 151 E Summit St., Harbor Springs, MI 49740, charles.leahy@sbcglobal.net)

As the sport of pickleball has grown in popularity and tennis courts close to homes are being converted to this use, noise regulation for pickleball courts has become a topic of discussion among city councils, planners, home owners associations, residents, and, increasingly, attorneys seeking injunctions against poorly sited or mitigated courts. Accurate noise impact assessment of the short duration impulsive sound produced by the impact of the ball on the paddles requires the use of the highly impulsive adjustment in ANSI S12.9 Part 4. Examination of the definitions of impulsive sound in this standard and in ISO 1996 Part 1 identifies characteristics of rapid onset, significant energy in the most sensitive part of the auditory spectrum, and duration as primary differentiators for categorization. The noise impact assessment methodology in ANSI S12.9 Part 4 for highly impulsive sound has proven an effective means of determining the amount of mitigation needed for pickleball facilities, being more repeatable and precise than other commonly used assessment methods while avoiding underestimation of annoyance.

9:20

5aNS6. Advancements in nanotechnology for acoustic management in pickleball. Eliot Arnold (4400 Shawnee Mission Parkway, Fairway, KS 66205, earnold@slncr.com)

This research investigates the use of advanced nano-fiber technology for sound and noise management in pickleball courts. The technology, known for its flexibility and adaptability, addresses the unique acoustic challenges of pickleball, a sport with a distinctive noise profile characterized by impulsive and unpredictable sounds. These nano-fibers are particularly effective in absorbing mid to high-frequency noises (800–5000 Hz) common in pickleball. Incorporating these nano-fibers into acoustic foams and textiles significantly enhances sound absorption, allowing for thinner materials while doubling performance compared to conventional materials. These fibers, about 1/500th the diameter of human hair, have a high surface area to volume ratio, aiding in sound scattering and increased friction with air molecules. This structure enables the efficient transformation of sound energy into heat, which is then effectively dissipated. Aligned with the Acoustical Society of America's standards, this abstract emphasizes a scientific breakthrough in sports acoustics, contributing to the reduction in urban noise pollution. The study underscores the impact of cutting-edge material technology in improving environmental acoustics and community well-being.

9:35

5aNS7. Understanding pickleball noise at the source: The vibroacoustics of the pickleball paddle and ball. Daniel A. Russell (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, dar119@psu.edu)

Pickleball continues to be America's fastest-growing sport, growing by roughly 150% every year with 48.3 million US adults (19% of the adult population) having played at least one game in 2023. This immense popularity and growth is accompanied by a similar increase in the number and voracity of complaints from the residential communities surrounding pickleball courts. This paper aims to explain the source of the pickleball noise—a short duration impulse with a strong tonal component near 1250 Hz—due to the impact of the pickleball ball and paddle, with a specific focus on the vibrational modes of the paddle that contribute most strongly to the tonal nature of the sound. Experimental modal analysis was performed on several paddles (wood, aluminum, composite, and carbon fiber) as well as the first paddle to meet the new USA Pickleball Quiet Category requirements. Spectral analysis of the impact identifies the offending tonal component near 1250 Hz to be a strongly radiating "membrane-type" mode of the paddle surface. Paddles for which this mode shape has a significantly higher or lower in frequency and amplitude result in a much less annoying impact sound. The role of vibrational modes in the ball will also be considered.

9:50–10:00 Panel Discussion

FRIDAY MORNING, 17 MAY 2024 ROOM 206, 8:00 A.M. TO 10:55 A.M.

Session 5aPA

Physical Acoustics, Biomedical Acoustics, and Structural Acoustics and Vibration: Nonlinear Acoustics in Solids

John M. Cormack, Cochair
Div. of Cardiology, Dept. of Medicine, Univ. of Pittsburgh, Pittsburgh, PA 15261

Christopher M. Kube, Cochair
Engineering Science and Mechanics, The Pennsylvania State Univ., 212 Earth and Engineering Sciences Bldg., University Park, PA 16802

Invited Papers

8:00

5aPA1. Direct, model-free acoustic evaluation of stresses and strains in loaded nonlinear soft solids. Michel Destrade (School of Math. Stat. Sci., Univ. of Galway, University Rd., Galway, Galway H91 TK33, Ireland, michel.destrade@nuigalway.ie), Guoyang Li (Dept. of Mech. and Eng. Sci., Peking Univ., Beijing, China), Artur L. Gower (Mech. Eng., Univ. of Sheffield, Sheffield, United Kingdom), Yanping Cao (Dept. of Eng. Mech., Tsinghua Univ., Beijing, China), Seok-Hyun Yun (Harvard Med. School and Wellman Ctr. for Photomedicine, Massachusetts General Hospital, Boston, MA), Zhaoyi Zhang, and Ziying Yin (Dept. of Eng. Mech., Tsinghua Univ., Beijing, China)

We show analytically and experimentally that the states of stress and strain existing in a mechanically loaded elastic material can be accessed directly from elastic wave speed measurements, without having to determine, or even know, its material linear and nonlinear elastic parameters. These techniques are expected to have important applications in the health monitoring of loaded structures. Examples include stressed hydrogels, muscles, and thin membranes, such as a stretched rubber sheet, a piece of cling film ($\sim$10 μm thick), and the

animal skin of a bodhrán, a traditional Irish drum. References: Z. Zhang, G.-Y. Li, Y. Jiang, Y. Zheng, A. L. Gower, M. Destrade, and Y. Cao, "Non-invasive measurement of local stress inside soft materials with programmed shear waves," Sci. Adv. 9, eadd4082 (2023); G.-Y. Li, A.L. Gower, M. Destrade, and S.-H. Yun, "Non-destructive mapping of stress and strain in soft thin films through sound waves," Commun. Phys. 5, 231 (2022).

8:20

5aPA2. Hysteretic friction between unfused surfaces as a potential source of acoustic nonlinearity and anelasticity in additively manufactured aluminum. Ward Johnson (National Inst. of Standards and Technol., 325 Broadway, MS 647, Boulder, CO 80305, wjohnson@boulder.nist.gov), Paul Heyliger (Dept. of Civil and Environ. Eng., Colorado State Univ., Fort Collins, CO), Jake Benzing, Orion L. Kafka, Newell H. Moser (National Inst. of Standards and Technol., Boulder, CO), Derek Harris, Jeremy Iten (Elementum 3D, Erie, CO), and Nik W. Hrabe (National Inst. of Standards and Technol., Boulder, CO)

Acoustic nonlinearity and loss determined by noncontacting nonlinear reverberation spectroscopy at resonant frequencies near 670 kHz are found to be anisotropic and correlated with porosity in commercially pure additively manufactured (AM) aluminum. These results, combined with results from Ritz vibrational modeling, x-ray computed tomography, and scanning electron microscopy, point towards a predominant role of acoustic shear displacement gradients across lack-of-fusion (LOF) defects. The nonlinearity and loss are also found to be dependent on accumulated duration of mode-specific acoustic excitation. Potential candidates for nonlinearity and anelasticity localized at LOF defects include dislocations and hysteretic friction/contact between unfused surfaces. Maps of geometrically necessary dislocation (GND) densities determined from electron backscatter diffraction provide no evidence of greater GND densities near LOF defects or significant dependence on porosity, and electron channeling contrast imaging similarly provides no evidence for greater dislocation densities within grains near LOF defects. In light of these results on dislocations, hysteretic friction of contacting surfaces within LOF defects is suggested as the most likely source of anisotropy and porosity dependence of the nonlinearity and loss. Under this hypothesis, the dependence on duration of acoustic excitation may be attributed to reconfiguration of surface asperities under dynamic stress.

8:40

5aPA3. Plasticity in nonlinear elastic metamaterials: Low-order dynamic modeling. Samuel P. Wallen (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, sam.wallen@utexas.edu), Michael R. Haberman (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Washington DeLima (Honeywell FM&T, KCNSC, Kansas City, MO)

Nonlinear elastic metamaterials have been shown to admit a variety of rich, dynamical features that can be leveraged to tailor the propagation of mechanical waves. Since these materials derive their properties from intricate, subwavelength geometries, direct numerical simulations are often prohibitively expensive at scales of interest. To overcome this limitation, reduced-order models, typically in the form of effective continua or discrete lattices that capture the essential features of the material at sufficiently long wavelengths, have been developed. While many prior studies have implemented these models successfully, the vast majority have considered only recoverable elastic deformations with linear damping and neglected history-dependent effects, such as plasticity and friction. In this presentation, we introduce an effective lattice modeling framework for nonlinear elastic metamaterials undergoing plastic deformation. Due to the history-dependent nature of plasticity, this framework generally yields a system of differential-algebraic equations whose computational cost is significantly greater than a purely elastic system of similar size. We apply the method to several examples of interest and explore means to obtain phenomenological elastic-plastic models for general material architectures.

9:00

5aPA4. Slow dynamics and the role of moisture. John Yoritomo (Acoust., Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375, john.yoritomo@nrl.navy.mil)

The elastic behavior of rocks and other composite materials cannot be entirely captured by the traditional theory of nonlinear elasticity, where the stress field is related to powers of the strain field. Rather, these materials display non-classical nonlinear elastic behavior, such as hysteresis, end-point memory, fast dynamics, and slow dynamics. Slow dynamics (SD) is characterized by a drop in material stiffness due to a minor mechanical conditioning, followed by a slow recovery back to the original macroscopic elastic state. SD has drawn particular attention because the recovery, often logarithmic in time, has been observed in a wide variety of materials, on lengths scales from the laboratory to the seismic, and on time scales from milliseconds to years. The universal character suggests a simple, fundamental mechanism for the SD recovery. This talk will present recent experiments that seek to test proposed SD mechanisms. In particular, the role of moisture will be investigated. The main experimental venue is the simplified structure introduced in previous work—a single bead confined between two slabs of a similar material. The main benefit of this structure over more commonly studied SD materials (sandstones and concrete) is the ability to control the environment around the contact points.

Contributed Papers

9:20

5aPA5. Wave transmission in 2D nonlinear granular-solid composite systems. Chongan Wang (Dept. of Mech. and Aerosp. Eng., Univ. of California San Diego, 3353 Lebon Dr., 101, San Diego, CA 92122, robertqhd1@gmail.com), Qifan Zhang (Wuhan Univ. of Technol., Wuhan, Hubei, China), Sameh Tawfick, and Alexander F. Vakakis (Univ. of Illinois at Urbana Champaign, Urbana, IL)

Wave propagation in granular media composed of contacting discrete elastic granules attracted considerable attention due to its highly tunable acoustic properties. While prior investigations focused on granular systems with natural boundaries (e.g., fixed or free), our study delves into two-dimensional nonlinear wave propagation within hybrid metamaterials composed of ordered granular media interacting with linearly elastic solids, employing discrete element modeling for the granular media and finite element analysis for the elastic solids. Challenges resulting from numerical instabilities arise from non-smooth contact nonlinearities and friction at granular and granular–solid interfaces. To tackle the numerical challenges, we developed an interrelated interpolation-iteration algorithm with a self-adaptive time scheme. Special consideration was given to the convergence

of contact forces at the granular–solid interface. Specifically, to ensure robust computational modeling without numerical instabilities, we monitored the eigenvalues of appropriately defined local maps governing the iterative computations of the contact forces. We demonstrated the capacity of these hybrid metamaterials for shock mitigation and non-reciprocal acoustics with properties that are passively tuned (self-adaptive with) energy. Such results and computational methods contribute to the predictive modelling and design of 2D granular media with flexible interfaces, with diverse applications, e.g., in shock protectors and acoustic diodes.

9:35–9:55 Break

9:55

5aPA6. Elastic bit and Berry phase: Investigating topological phenomena in a classical granular network. M. Arif Hasan (Mech. Eng., Wayne State Univ., 5050 Anthony Wayne Dr., Detroit, MI 48202, Hasan.Arif@Wayne.Edu) and Kazi Tahsin Mahmood (Mech. Eng., Wayne State Univ., Detroit, MI)

This study investigates the Berry phase, a key concept in classical and quantum physics, and its manifestation in a classical system. We achieve controlled accumulation of the Berry phase by manipulating the elastic bit (a classical analogue to a quantum bit) in an externally driven, homogeneous, spherical, nonlinear granular network. This is achieved through the classical counterpart of quantum coherent superposition of states. The elastic bit's state vectors are navigated on the Bloch sphere using external drivers' amplitude, phase, and frequency, yielding specific Berry phases. These phases distinguish between trivial and nontrivial topologies of the elastic bit, with the zero Berry phase indicating pure states of the linearized granular system and the nontrivial π phase representing equal superposed states. Other superposed states acquire different Berry phases. Crucially, these phases correlate with the structure's eigenmode vibrations: trivial phases align with distinct, in-phase, or out-of-phase eigenmodes, while nontrivial phases correspond to coupled vibrations where energy is shared among granules, alternating between oscillation and rest. Additionally, we explore Berry's phase generalizations for non-cyclic evolutions. This research paves the way for advanced quantum-inspired sensing and computation applications by utilizing and controlling the Berry phase.

10:10

5aPA7. Predicting second-harmonic generation in shear wave beams in tissue-mimicking phantoms. Philip G. Kaufinger (Appl. Res. Labs., University of Texas at Austin, Austin, TX 78758, pkaufinger@utexas.edu), 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)

Planar nonlinear shear waves in isotropic media are subject to only cubic nonlinear effects and generate only odd harmonics during propagation. However, wavefront curvature in shear wave beams breaks the symmetry and yields quadratic nonlinear effects, and therefore, a second harmonic may be present in shear wave beams depending on the polarization of the wave. Focused radially polarized shear wave beams have been generated in tissue-mimicking phantoms [Cormack *et al*., IEEE TBME (2024)], and it is postulated that the second harmonic may be used to estimate the quadratic nonlinearity in such a medium as an additional biomarker for diseased tissue. Here, an analytical solution obtained in the paraxial approximation for nonlinear propagation of a shear wave beam in an absorbing medium [Spratt *et al*., AIP Conf. Proc. **1685**, 080007 (2015)] is used to characterize the strength of second-harmonic generation. Feasibility of measuring the second harmonic experimentally in a weakly nonlinear, radially polarized focused shear wave beam propagating in a tissue-like medium is explored. Second-harmonic generation in focused shear wave beams with other polarizations is discussed. [P.G.K. was supported by the ARL:UT Chester M. McKinney Graduate Fellowship in Acoustics.]

10:25

5aPA8. Strong diffraction of nonlinear surface acoustic waves in crystals. Brittany A. McCollom (Appl. Res. Labs. and Walker Dept. of Mech. Eng., Univ. of Texas at Austin, Austin, TX 78713, bmccollom@utexas.edu), John M. Cormack (Div. of Cardiology, Dept. of Medicine, Univ. of Pittsburgh, Pittsburgh, PA), Edoardo Baldini (Dept. of Phys., Univ. of Texas at Austin, Austin, TX), and Mark F. Hamilton (Appl. Res. Labs. and Walker Dept. of Mech. Eng., Univ. of Texas at Austin, Austin, TX)

Nonlinear distortion and shock formation in planar surface acoustic waves in anisotropic crystals have been modeled without [Hamilton *et al*., JASA (1999)] and with [Cormack *et al*., JASA 2022] piezoelectricity. Weak diffraction has been included in the paraxial approximation for nonlinear surface wave beams in isotropic solids [Shull *et al*., JASA (1995)]. Here, a procedure for including strong diffraction, i.e., without the paraxial approximation, in the model for nonlinear surface waves in crystals is presented, which incorporates the angular spectrum approach described by Kharusi and Farnell (JASA 1970). Anisotropy is defined by expressing the phase speed of a plane wave, and therefore, the magnitude of the corresponding wavenumber, as a function of the direction of propagation. Next, an explicit expression relating the two wavenumber components in the planar surface along which the wave propagates must be obtained as a function of direction, requiring iterative solution of a transcendental relation. The beam is then propagated incrementally away from the source, advancing the angular spectrum in k-space and including nonlinear interactions in the spatial domain to characterize the combined effects of diffraction and nonlinearity. Preliminary results are presented for converging nonlinear surface waves in crystals. [Work supported by IR&D at ARL:UT.]

10:40

5aPA9. Studying the nonlinearity effects in ultrasound-assisted water purification and treatment systems. Pooja Dubey (George W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., 2 Rue Marconi, Georgia Tech-Europe, Metz 57070, France, pooja.dubey@gatech.edu) and Nico Declercq (George W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., Metz, France)

In light of industrial sectors' rapid and exponential growth, the deleterious specter of water pollution looms ominously over our environment. One of the significant challenges is obtaining a sustainable and energy-efficient water purification/treatment system. Since the 1990s, several research studies have been proposed to demonstrate the usefulness of ultrasound as a water purification tool. As a result, newly designed devices such as ultrasound-assisted electrochemical treatment and ultrasound-assisted heat exchanging devices are becoming more common. However, such devices' high voltage ultrasonic emission is a significant problem because the nonlinear acoustic effects are not well understood and are, therefore, not adequately integrated into the design of the devices. Furthermore, the presence of biofilms in these devices creates more complexity due to the interaction of high-amplitude ultrasonic waves with the biofilm network. To better overcome the inefficient functioning of such devices and adverse operational issues, the current study aims to investigate and explain the nonlinear ultrasonic effects in ultrasonic-assisted purification devices, with and without biofilm deposits. The obtained insight will help develop an effective design strategy for high efficiency.

Session 5aSA

Structural Acoustics and Vibration: General Topics in Structural Acoustics

Alyssa Bennett, Cochair
Naval Surface Warfare Center Carderock, 9500 MacArthur Blvd, Bethesda, MD 20817

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

Contributed Papers

8:00

5aSA1. Acoustic source localization on finite structures using remote sensors. Allison M. King (Dept. of Mech. Eng., Univ. of Michigan, 1231 Beal Ave. Ann Arbor, MI 48109, kingalli@umich.edu) and David R. Dowling (Naval Architecture and Marine Eng., Univ. of Michigan, Ann Arbor, MI)

Acoustic waves are well-suited for remote sensing applications and structural health monitoring purposes because they convey information about their source and can be measured using non-contacting methods. Source localization is an important structural health monitoring task; however, traditional time-of-flight array signal processing techniques used to localize acoustic sources are ill-suited for many structural engineering applications due to the potential for complicated propagation paths, the dispersive propagation of acoustic waves in structures, and the coupling of the vibrating structure and the surrounding medium. Thus, source localization experiments were conducted using matched field processing (MFP) for a 0.9-m diameter round aluminum plate excited by the impact of a 1.3-cm stainless-steel ball bearing dropped from 7.6-cm. A 14-sensor linear remote acoustic array placed 8.9-cm above the plate measured the sound radiated by the 0.64-cm thick plate. MFP array signal processing localization techniques were used along with a physics-based finite element acoustic model to localize the excitation on the structure. Source localization results in both a quiet environment and environment with additive white Gaussian noise are discussed, and these results are compared to those from an acoustic model where plate edge reflections are neglected. [Work was sponsored by a SMART Scholarship and the NEEC.]

8:15

5aSA2. Passive structural health monitoring of a vibrating shell using data-based matched field processing. Sandrine T. Rakotonarivo (Marine Physical Lab., SCRIPPS Inst. Oceanogr., UCSD, LMA - UMR 7031 AMU - CNRS - Centrale Marseille, 4 impasse Nikola Tesla, Marseille 13453, France, sandrine.rakotonarivo@univ-amu.fr), Theo Langlet (CEA Cadarache, MISTRAL Lab., LMA, AMU, Marseille, France), Jit Sarkar (Marine Physical Lab., SCRIPPS Inst. Oceanogr., UCSD, San Diego, CA), and William Kuperman (Marine Physical Lab., SCRIPPS Inst. Oceanogr., UCSD, La Jolla, CA)

This study presents experimental results on passive structural health monitoring of a vibrating elastic structure for defect localization. The approach is based on matched field processing (MFP), which requires a model of the pristine structure. An MFP methodology for localizing a defect in a shell equipped with vibration sensors was developed and numerically demonstrated in [JASA EL **2**, 025601 (2022)]. This paper further extends this study to experimental implementation of this MFP methodology without requiring full knowledge of the structure parameters and the boundary conditions of the pristine structure. The method is tested on experimental data for localizing a defect on a rectangular steel plate with non-perfectly, fixed boundary conditions.

8:30

5aSA3. Sensing touch location on an elastic surface by monitoring structural vibrations. Benjamin R. Thompson (Elec. & Comput. Eng., Univ. of Rochester, 120 Trustee Rd., Rochester, NY 14620, bthomp23@ur.rochester.edu), Tre DiPassio, Jenna Rutowski, Mark Bocko, and Michael C. Heilemann (Elec. & Comput. Eng., Univ. of Rochester, Rochester, NY)

In recent years, touchscreens have become ubiquitous, and projected capacitance (p-cap) has arisen as the dominant touch-sensing technology. Though powerful, p-cap has some characteristics that make it less suited for certain applications. Active acoustic sensing (AAS) systems work by sensing the vibration response of an elastic surface to a system-generated excitation and associating specific response characteristics with touch locations on the surface. These systems have the potential to offer advantages over p-cap in terms of scalability, cost, and performance in harsh environments. AAS systems may also offer the ability to sense touch pressure as an additional input parameter. We developed an empirical model of surface vibrational response as a function of the location of an applied force and then used this model to inform the signal processing and machine learning approaches employed in a prototype AAS system. In this presentation, we provide results from our empirical model as well as details of the prototype system, including its construction and performance.

8:45

5aSA4. Direction of arrival estimation in reverberant environments using a single vibration sensor on an elastic panel. Jenna Rutowski (Elec. & Comput. Eng., Univ. of Rochester, 500 Joseph C. Wilson Blvd, Rochester, NY 14627, jrutowsk@ur.rochester.edu), Tre DiPassio, Benjamin R. Thompson, Mark Bocko, and Michael C. Heilemann (Elec. & Comput. Eng., Univ. of Rochester, Rochester, NY)

The vibrational response of an elastic panel to an incoming acoustic pressure wave is dependent on the coupling between the incident angle and the panel's bending modes. By examining the relative modal excitations recorded by a single structural vibration sensor affixed to the panel, the direction of arrival (DOA) of the incident wave may be inferred. In reverberant environments, determination of the DOA of a sound source is complicated by acoustic reflections. Panel microphones may be particularly susceptible to this effect due to their large surface areas and finite modal decay times. A study on the impact of reverberation on DOA estimation using panel microphones was conducted. The panel's response to wake word utterances was recorded in eight spaces, with reverberation times (RT60s) ranging from 0.27 to 3.00 s. These responses were used to train neural networks for DOA estimation. Results indicate an inverse relationship between RT60 and DOA estimation reliability. Within $\pm 5^{\circ}$, DOA estimation reliability was measured at 95% in the least reverberant space and decreased to 78% in the most reverberant space. Results also suggest that DOA estimation using panel microphones can adapt to diverse acoustic environments by training the system with data from multiple spaces with different RT60s.

9:00

5aSA5. Experimental modal testing of growing thin ice. Carl R. Hart (U.S. Army Engineer Res. and Development Ctr., Cold Regions Res. and Eng. Lab., 72 Lyme Rd., Hanover, NH 03755, carl.r.hart@erdc.dren.mil), Cody M. Best, Emily Asenath-Smith, Kiera L. Thompson Towell (U.S. Army Engineer Res. and Development Ctr., Cold Regions Res. and Eng. Lab., Hanover, NH), Michelle E. Swearingen (Construction Eng. Res. Lab., US Army ERDC, Champaign, IL), and Michael B. Muhlestein (U.S. Army Engineer Res. and Development Ctr., Cold Regions Res. and Eng. Lab., Hanover, NH)

Current practices for assessing the thickness of growing ice include drilling and coring. This is inherently risky and motivates the use of a stand-off method. Laser Doppler vibrometry is a potential technique, however, the interpretation of vibrometer signals must be informed by a physical understanding of ice subjected to mechanical vibrations. The vibrational response of growing, thin ice is largely unknown. For thick ice, adequate predictions for the vibrational response assumes ice behaves as an elastic plate. It is hypothesized that thin ice responds in a fashion somewhere between an elastic membrane and an elastic plate. A modal test of thin ice growing within an insulated plastic basin was conducted over the course of two tests. The resulting dataset spans a range of ice thicknesses from 8 to 45 mm. Examination of Bode and Nyquist plots showed that the two lowest observable ice resonances, as measured at the center of the ice sheet, exhibit a dependence on ice thickness distinctly different from a fixed elastic plate loaded by a half-space of water. Preliminary finite-element analysis calculations indicates that the basin wall compliance is a significant factor in modifying the ice resonance versus thickness relationship.

9:15

5aSA6. Coupling the force analysis technique and full-field vibration measurements for the identification of a time-space-varying sound pressure loading on a membrane. Anaïs Mougey (Université de Sherbrooke, 2500 boulevard de l'université, Sherbrooke, QC J1K 2R1, Canada, anais.mougey@usherbrooke.ca), Manuel Melon, Félix Foucart (Le Mans Université, Le Mans, France), and Olivier Robin (Université de Sherbrooke, Sherbrooke, QC, Canada)

This work is grounded on the force analysis technique, an identification method that directly uses a structure's equation of motion to formulate an inverse problem, explicitly identifying the force causing the structure's motion. Prior research employing this technique has been predominantly conducted in the frequency domain and was limited to stationary, mechanical excitations using physical sensor arrays such as accelerometers. The objective of this research was mostly quantitative (amplitude, location), while the proposed approach is rather qualitative identification. Indeed and by combining the force analysis technique with full-field and non-contact vibration measurements conducted on a system, here a membrane, this communication describes a proof-of-concept for the identification of a time-space-varying sound pressure loading. A compact and tonal sound source is used to draw freehand shapes against the membrane surface, and the objective is to follow/reconstruct the trajectory followed by this source. Results are provided for different drawn shapes or letters, and the effect of mechanical or calculation parameters on the reconstructed information is studied. Finally, potential research directions are discussed and fed by preliminary measurements on a percussion instrument.

9:30

5aSA7. Forces on a singing wineglass rim. Megan F. Orzolek (Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, mfo4@psu.edu), Alexander M. Mertz, and Michael L. Jonson (Penn State, University Park, PA)

The characterization of friction is a multiscale problem involving a balance between asperity contacts and fluid film forces, most commonly modeled with a velocity-dependent coefficient of friction. Periodic stick–slip phenomena are often associated with the friction force that causes mechanical vibration. This type of vibration can cause noise to radiate via structural modes of a system. Notable examples include journal bearings, brake squeal, turkey calls, and wineglasses. Singing wineglasses have been used for centuries as musical instruments, so musical acousticians have studied the structural vibrations and acoustic output. However, the tri-axial force applied to the rim has not been measured. A test rig was designed to measure the vertical, tangential, and radial dynamic forces applied to the spinning glass rim and simultaneously capture the radiated noise. The sound radiated from the wineglass was observed to depend on the relative velocity at the rim and applied vertical load, so a stability region for a clear sound was determined. Additionally, the strongest radiating modes and harmonics in the pressure were found in the concurrently measured nonlinear force. This work may be extended to confirm the force measurement by existing friction models, a water medium, and more complicated systems.

9:45–10:00 Break

10:00

5aSA8. Abstract withdrawn.

10:15

5aSA9. Leak detection in an operational underground water distribution network using active acoustics. Pranav Agrawal (Civil and Environ. Eng., Univ. of California, Los Angeles, 580 Portola Plaza, 5731 Boelter Hall, Los Angeles, CA 90095-1593, pranav0505@g.ucla.edu), Stan Fong, Dirk Friesen (Digital Water Solutions, Waterloo, ON, Canada), and Sriram Narasimhan (Civil and Environ. Eng., Univ. of California, Los Angeles, Los Angeles, CA)

Leaks present in the water distribution networks (WDNs) lead to an enormous loss of a valuable resource, not only affecting the economic efficiency of water utilities but also posing significant potential safety hazards, and an increased burden on infrastructure maintenance. In the past, several studies have focused on passive acoustic-based leak detection by primarily sensing and analyzing the acoustic signals emitted from the leak source. In this study, we installed a low-frequency transducer in the water column of an operational WDN near Los Angeles, California, and excited the system using steady-state sinusoidal and sine burst signals. The acoustic pressure signals inside the pipe network were measured at multiple locations using state-of-the-art hydrophone-enabled devices retrofitted to fire hydrants. The experiments were conducted in the presence of a simulated leak in the network. The leak acts as an impedance discontinuity for the acoustic wave propagation, and therefore, the excited signals undergo partial reflection at the leak location. Based on signal processing techniques, we attempt to detect and localize the leaks in the WDN. The goal of using the active acoustics is to detect small leaks and increase the range of sensing.

10:30

5aSA10. Tri-axial ground-borne vibration measurements during rail pass-bys. Harry Ao Cai (HGC Eng., 2000 Argentia Rd., Ste. 203, Plaza 1, Mississauga, ON L5N 1P7, Benin, hcai@hgcengineering.com), Nathan Gara (HGC Eng., Mississauga, ON, Canada), and Brian Howe (HGC Eng., Mississauga, Cambodia)

Ground-borne vibrations from rail pass-bys, transmitted from train wheels rolling on the rails, have the potential to cause various adverse effects at nearby receptors, such as annoyance and re-radiated noise. To assess the impact of rail pass-bys, surface-level vibration can be characterized in three components: one vertical and two horizontal directions. Prior experience, best-practice guidelines from the Federal Transit Administration Transit Noise and Vibration Impact Assessment Manual, and theory of propagation of Rayleigh surface waves indicate that the vertical component dominates the horizontal components, such that vibrations from rail pass-bys can be adequately characterized by the vertical component only. This purpose of this study is to assess the sufficiency of characterizing the impact of ground-borne vibrations caused by rail pass-bys based solely in the vertical direction. Tri-axial vibration measurements were conducted for freight and passenger trains in Ontario, Canada, using a multi-channel signal analyzer for simultaneous measurement of multiple-axis vibration levels. The measured levels were then examined across the three components.

10:45

5aSA11. Analysis, design, and installation of vibration isolation for lightweight helipads. Alfredo Rodrigues (CDM Stravitec, 100 Sunrise Ave., 202, Toronto, ON M4A1B3, Canada, a.rodrigues@cdm-stravitec.com), Ashwin Dias (CDM Stravitec, Overijse, Belgium), and Freddy Saddik (CDM Stravitec, Dallas, TX)

This case study discusses the design and manufacturing steps for the installation of a vibration isolation solution for two lightweight helidecks installed on the rooftop of an existing medical center building. The design stage involved the analysis of the relevant information, involving several specialties, such as Acoustics (for sources of vibration and acoustic performance requirements), Heliport Design[AR1] (for the type of aircraft, MTOW, number of supports, static and dynamic loads), and Structural (for the types of connections to the existing building structure). A collaborative effort between CDM Stravitec and the helideck supplier led the design through a progressive and iterative process where a final design that responded to the strong acoustical requirements and the complex nature of a fully functional and integrated lightweight helideck. Along the design, compromises where made ensure that requirements are met for both acoustic and structural performance. The solution delivered was a prefabricated box, which could easily be installed on site, comprised the necessary components, such as springs and uplift restraints to deal with the challenging aspects of the isolation of lightweight helidecks. The paper also discusses the challenges and best practices encountered during the production and installation of the system.

11:00

5aSA12. Optimization of an aircraft fuselage assembly to minimize radiated sound power. Jordan Howes (Mech. and Mater. Eng., Queen's Univ., 130 Stuart St., Kingston, ON K7L2V9, Canada, jordan.howes@queensu.ca), Adam McKenzie, Wesley Dossett, Luke Crispo, and Il Yong Kim (Mech. and Mater. Eng., Queen's Univ., Kingston, ON, Canada)

During the structural design of an aircraft fuselage, the acoustic performance is generally not considered. However, vibrations transmitted through the fuselage generate noise leading to passenger discomfort. Currently, the main source of noise reduction comes from the addition of damping material to the fuselage. This work investigates the use of various design optimization techniques to reduce radiated sound power by changing the fuselage structure's geometry which consists of a skin panel, frames, and stringers. Design optimization tools, such as topology, size, and shape optimization were used to determine an optimal design for each design variable that minimized radiated sound power in the fuselage assembly. The design variables that were studied included skin panel thickness and design, frame and stringer spacing, and frame and stringer cross section. Equivalent radiated power was used as an objective function for numerical optimizations within Altair OptiStruct as an indirect method for minimizing radiated sound power numerically. To understand the source of the sound power improvements, a physics analysis was conducted for each design variable that compared sound power, equivalent radiated power, and radiation efficiency.

11:15

5aSA13. Comparison and analysis of onboard vibration and underwater noise radiated by ships of different classes. Kamal Kesour (Innovation Maritime, 53 Rue St-Germain Ouest, Rimouski, QC G5L 4B4, Canada, kkesour@imar.ca), Paul Camerin, Jean-Christophe G. Marquis (Innovation Maritime, Rimouski, QC, Canada), and Cédric Gervaise (SenseaFR, Grenoble, France)

The shipping industry plays a major role in the increase in anthropogenic noise in the marine environment. To understand and measure the underwater noise emitted by ships, the MARS (Marine Acoustic Research Station) project has deployed an acoustic station in the St. Lawrence River to measure ship signatures between 2021 and 2023. The acoustic station complies with the ANSI/ASA S12.64 2009 standard. In addition, the project team carried out simultaneous vibration diagnostics to identify sources of underwater noise on board several vessels. This paper presents the main results of the onboard vibration measurements, including a comparison of vibration levels measured on bulk carriers, tankers, and general cargo ships. The contribution of diesel generators, propulsion system, and propeller cavitation to the acoustic signature is also presented and analyzed for each vessel class. As expected, propeller cavitation was found to be the main source of shipborne noise of the investigated vessels, especially above 50 Hz. Finally, the acoustic signatures of ships equipped with electric and diesel propulsion systems are also compared, while identifying the frequencies related to each source.

11:30

5aSA14. Mode-matching analysis of acoustic scattering in bifurcated circular cylindrical waveguides with liner conditions and step-discontinuity. Qamar Abbas (Mathematics, Indus College Kahuta, Rawalpindi, Pakistan, Indus College Kahuta, Rawalpindi Rd., Kahuta, Punjab 47330, Pakistan, qamar299ICK@gmail.com) and Rab Nawaz (Mathematics, COMSATS Univ. Islamabad, Islamabad, Pakistan)

The study delves into the acoustic scattering from a bifurcated circular cylindrical waveguide, considering liner conditions and step-discontinuity. Specific impedance of acoustic liners is developed for analyzing scattering characteristics. Two problems, with and without liners, are formulated and solved using Mode-matching technique, based on eigenfunction expansions. The solution's accuracy, dependent on eigenfunction properties influenced by the medium and wall conditions, is validated through power conservation and matching conditions. Results reveal the considerable impact of reacting liners and step-discontinuity on scattering concerning frequency and duct dimensions. The study implicates to various fields where the control and understanding of acoustic scattering phenomena are crucial for optimizing the performance of devices and systems.

Session 5aSC

Speech Communication: Speech Production and Speech Tech Poster Session

Zhaoyan Zhang, Chair
UCLA School of Medicine, Los Angeles, CA 90095

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

Contributed Papers

5aSC1. Comparison of automated and manual measures of speaking rate in individuals with dysarthria. Lian J. Arzbecker (Communicative Disord. and Sci., Univ. at Buffalo, 103D Cary Hall, Buffalo, NY 14207, arzbecker.1@osu.edu) and Kris Tjaden (Communicative Disord. and Sci., Univ. at Buffalo, Buffalo, NY)

Speech patterns in speakers with dysarthria exhibit a broad range of prosodic and articulatory variations, potentially posing a significant challenge for automated speech recognition systems. This study seeks to address a gap in the literature by exploring a fully automated approach to determine speaking rates in English-speaking individuals with dysarthria. This research aims to elucidate whether a syllable-based automated method yields results akin to manual processes when calculating speaking rates across diverse groups [i.e., individuals with dysarthria secondary to multiple sclerosis (MS) or Parkinson's disease (PD), and healthy controls]. Moreover, this study examines the performance of an automated method across three distinct speech tasks: isolated sentences, paragraph reading, and spontaneous speech. The methodology involves adjusting parameters within a version of a published script specifically designed for calculating speaking rate through syllable detection (Praat Script Syllable Nuclei; de Jong *et al.*, 2021). Using a sample of 60 speakers (20 MS, 20 PD, and 20 healthy controls), this study aims to evaluate the accuracy and validity of an automated script in calculating speaking rate. Ultimately, these findings are intended to guide speaking rate measurement of dysarthria in research and clinical practice.

8:00

5aSC2. Subglottal resonances in healthy older adults. Steven M. Lulich (Speech Lang. & Hearing Sci., Indiana Univ., 200 South Jordan Ave., Bloomington, IN 47405, slulich@indiana.edu)

Resonances of the subglottal airways impact the vocal tract transfer function and voice production during speech through linear and non-linear acoustic coupling. It has been posited that subglottal resonances form acoustic boundaries between contrasting sets of vowels and consonants based on their coupling effects on formant frequencies. Thus far, studies of subglottal resonances have reported data almost exclusively from children and young adults; data from older adults are rare. This study investigates subglottal resonances in eight healthy older adults between 50 and 80 years of age (5 males, 3 females). Initial analyses indicate that the first and especially the second subglottal resonances are at lower frequencies than expected when compared against height-matched young adults but that the subglottal resonance frequencies are not dependent on posture, vowel acoustics, or standard measures of pulmonary function.

5aSC3. Modeling syllable rhythms for interactive alignment studies. Brandon Copping (Commun. Sci. and Disord., Univ. of Memphis, 4055 North Park Loop, Memphis, TN 38122, bcopping@memphis.edu) and Eugene Buder (Commun. Sci. and Disord., Univ. of Memphis, Memphis, TN)

Many acoustic cues have been proposed as indicators and coordinators of upcoming turn-exchanges in conversational interactions, including syllable rhythm. Rhythmic coordination can help to explain close coordination of speech during these exchanges with intervals on the order of 250 ms or less, or with fluent and non-disruptive overlaps. While general questions surrounding speech rhythm remain unanswered, investigations of syllable alignment across partners in conversation can bypass such questions by assessing directly for patterns that align across vocal exchanges. We develop models of syllable rhythm for this purpose using hand-coded syllable durations from spontaneous conversations and from turn-taking responses elicited by recorded stimuli. Our approach builds on evidence that turn-takers begin planning their utterance approximately 0.5 s prior to turn endings. Syllable duration strings from appropriately identified frames are replaced by Gaussian curves to create oscillatory waveforms, and cross-correlation functions are applied to identify rhythmic continuities across exchanges. Patterns identified using this technique are explanatory of close coordination in the spontaneous exchanges, and supra-syllabic units such as the foot and phrase are identified as well. This syllable framework can be developed further by incorporating additional acoustic measures such as f_{o} and amplitude to build a multidimensional construct for interactional rhythm alignment.

5aSC4. Mapping acoustic characteristics of emotional prosody in Mandarin disyllabic words: A machine-learning approach. Xuyi Wang (Speech-Language-Hearing Ctr., School of Foreign Lang., Shanghai Jiao Tong Univ., Dongchuan Rd. 800, Shanghai 200240, China, wxy_evie@sjtu.edu.cn), Hongwei Ding (Speech-Language-Hearing Ctr., School of Foreign Lang., Shanghai Jiao Tong Univ., Shanghai, China), and Yang Zhang (Speech-Language-Hearing Sci. and Masonic Inst. for the Developing Brain, Univ. of Minnesota, Minneapolis, MN)

This study conducted an acoustic-prosodic mapping analysis of emotional prosody in Mandarin Chinese. It utilized a validated audiometry corpus with 450 disyllabic words. The spoken words covered five basic emotions produced by a female speaker: angry, sad, happy, fearful, and neutral. A machine-learning approach was adopted to map key acoustic-prosodic features for Mandarin emotional vocalization. The results revealed distinctive acoustic profiles for each emotion, highlighting variations in fundamental frequency, intensity, speaking rate, and voice quality. Emotional utterances consistently exhibited higher mean F0 values than neutral expressions. Fear displayed the highest crest in F0. Angry and happy utterances showed greater vocal intensity and a faster speaking rate compared to fearful and sad expressions. While anger was associated with a creaky voice quality, sadness corresponded with a breathier voice quality. The current findings are limited with the use of the single-speaker corpus. Ongoing efforts aim to expand the corpus with more speakers to test the

generalizability and scalability of the analysis approach for subsequent investigations.

5aSC5. How nasal airflow can affect Nasalance magnitude. Liran Oren (Otolaryngol., Univ. of Cincinnati, University of Cincinnati, PO Box 670528, Cincinnati, OH 45267, orenl@ucmail.uc.edu)

Nasometry is a method for evaluating the function of the velopharyngeal valve. This technique provides a measure of Nasalance, calculated from acoustic energy captured with a device (nasometer) that can separate the oral and nasal acoustic signals using a sound separation plate. We previously observed that the presence of nasal emission can dramatically increase nasalance magnitude. The unintended elevation of nasalance occurs because airflow from the nares impinges on the nasometer's nasal microphone. We aim to quantify this effect using a customized nasometer in patients diagnosed with nasal emission. The customized nasometer has six microphones (three pairs) placed in a radial configuration on the top and bottom of the separation plate. Ideally, all microphones on the same side (i.e., top or bottom) should measure the same Nasalance value. A difference in the measurement of the nasal microphones would be attributed to flow interference. Preliminary results show that the airflow can artificially elevate the Nasalance magnitude by as much as 15 points depending on the produced syllable. These findings can help explain the common discrepancy observed in speech clinics between the Nasalance magnitude and the perceived severity of nasal emission.

5aSC6. Optimization of classifying accurate and misarticulated speech sounds for use in a gamified real-time ultrasound biofeedback system. Sarah C. Biehl (Biomedical Eng., Univ. of Cincinnati, 3159 Eden Ave., Cincinnati, OH 45219, biehlsc@mail.uc.edu), Sarah Dugan (Rehabilitation, Exercise, and Nutrition Sci., Univ. of Cincinnati, Dayton, OH), Sarah R. Li, Alex Knapp (Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH), Renee Seward (School of Design, Univ. of Cincinnati, Cincinnati, OH), Michael A. Riley (Rehabilitation, Exercise, and Nutrition Sci., Univ. of Cincinnati, Cincinnati, OH), Suzanne Boyce (Commun. Sci. and Disord., Univ. of Cincinnati, Cincinnati, OH), and T. Douglas Mast (Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH)

Integrating ultrasound biofeedback therapy (UBT) into a real-time, gamified interface to provide articulatory feedback for speech remediation promotes an external focus of attention, thereby reducing the complex cognitive demands required for standard UBT. Previous studies have shown that accuracy of American English rhotic /r/ can be predicted by a single parameter, the difference between tongue dorsum and blade displacements measured by ultrasound imaging during speech production. This parameter has classified speech productions of rhotic syllables as correct versus misarticulated with a classification accuracy up to 85%. However, implementation of this classification approach into real-time gamified UBT, including both measurement timing and establishment of difficulty levels for progressive therapy, would benefit from optimization using a larger dataset. 2,450 productions of 10 distinct rhotic syllables (including prevocalic and postvocalic contexts) from 50 children, with and without misarticulations, were analyzed. For each production, ultrasound image sequences were processed by TonguePART software to acquire tongue displacement trajectories, and accuracy was judged by trained listeners using a visual analog scale. Analyses were conducted to optimize selection of the image frame for classification, determine parameter thresholds appropriate for real-time prediction of /r/ production accuracy, and integrate these thresholds into a difficulty level design for gamified UBT.

5aSC7. Control parameters for coordinative structures in speech production. Matthew Masapollo (McGill Univ., 630 William St., Montreal, QC H3C 4C9, Canada, matthew.masapollo@mcgill.ca) and Susan Nittrouer (Speech, Lang., and Hearing Sci., Univ. of Florida, Gainesville, FL)

In skilled speech production, sets of articulators work cooperatively to achieve task-specific movement goals, despite rampant contextual variation. Efforts to understand these functional units, termed coordinative structures, have focused on identifying the essential control parameters responsible for allowing articulators to achieve these goals, with some research focusing on temporal parameters (relative timing of movements) and other research focusing on spatiotemporal parameters (phase angle of movement onset for one articulator, relative to another). Here, we compared findings across three recent studies where both types of control parameters were investigated, using electromagnetic articulography recordings. In each study, talkers produced VCV utterances, with alternative V (/ɑ/−/ɛ/) and C (/t/−/d/ or /p/−/b/), across variation in rate (fast–slow) and stress (first syllable stressed–unstressed). Two measures were obtained: (i) the timing of tongue-tip or lower-lip raising onset for intervocalic C, relative to jaw opening–closing cycles, and (ii) the angle of tongue-tip or lower-lip raising onset, relative to the jaw phase plane. All three studies showed that the correlations of tongue-tip/lower-lip movement onset latencies and jaw opening-closing cycle durations were stronger and more reliable than the correlations of tongue-tip/lower-lip phase angles and jaw opening-closing cycle durations, demonstrating that timing is the critical control parameter.

5aSC8. Generalization of inter-articulator timing control: Evidence from tongue-jaw and lip-jaw kinematics using electromagnetic articulography. Matthew Masapollo (McGill Univ., 630 William St., Montreal, QC H3C 4C9, Canada, matthew.masapollo@mcgill.ca), Ana Rodriguez, Kara Kent (Speech, Lang., and Hearing Sci., Univ. of Florida, Gainesville, FL), Rosalie Gendron (McGill Univ., Montreal, QC, Canada), Hannah Thomas (Speech, Lang., and Hearing Sci., Univ. of Florida, Gainesville, FL), Nathan Maxfield (Univ. of South Florida, Tampa, FL), and Susan Nittrouer (Speech, Lang., and Hearing Sci., Univ. of Florida, Gainesville, FL)

Recent research in speech production indicates that talkers reliably control the relative timing of articulator movement onsets across variation in production rate, syllable stress, and segmental makeup and that this precision of inter-articulator timing control instantiates phonetic structure. To date, these timing relations have been highly reliable for tongue-jaw kinematics. In the present study, we address the generality of these timing relations to lip-jaw kinematics. Eleven talkers recorded 240 /tV#Cat/ and 240 /bV#Cab/ utterances using electromagnetic articulography, with alternative V (/ɑ/-/ɛ/) and C (/t/−/d/ or /p/−/b/), across changes in production rate (fast-normal) and stress (first syllable stressed-unstressed). To quantify inter-articulator temporal coordination, the timing of either tongue-tip or lower-lip raising onset for the intervocalic C, relative to the jaw opening-closing cycle for V, was obtained. Results indicate that the same kinematic pattern occurred among both sets of articulators: any manipulation that shortened the jaw opening-closing cycle reduced the latency of either tongue-tip or lower-lip movement onset, relative to the onset of jaw opening. Furthermore, the movement onset latencies of the tongue-tip and lower-lip were both highly differentiated by utterance type, bolstering the view that inter-articulator timing relations instantiate phonetic structure in the resulting acoustic signal.

5aSC9. Real-time speech adaptations in conversations between human interlocutor and AI confederate. Fenqi Wang (Dept. of Linguist, Simon Fraser Univ., 8888 University Dr., Burnaby, BC V5A 1S6, Canada, fenqiw@sfu.ca), Jetic Gu (School of Computing Sci., Simon Fraser Univ., Burnaby, BC, Canada), Meagan Durana, Chihiro Mabohang (Dept. of Linguist, Simon Fraser Univ., Burnaby, BC, Canada), Dawn Behne (Dept. of Psych., Norwegian Univ. of Sci. and Technol., Trondheim, Norway), Allard Jongman (Dept. of Linguist, The Univ. of Kansas, Lawrence, KS), Joan Sereno (Dept. of Linguist, The Univ. of Kansas, Kansas City, KS), and Yue Wang (Dept. of Linguist, Simon Fraser Univ., Burnaby, BC, Canada)

Compared to human-directed adaptations, less is known about how humans adjust their speech for intelligibility benefits while interacting with an AI-powered voice interface. In this study, we investigate human speech adaptations in human-to-human versus human-to-AI unscripted conversations. Specifically, we examine the production of words containing intervocalic /t-d/ in a conversation between a speaker who distinguishes these two stops (e.g., metal–medal) and a speaker ("flapper") who merges the two stops into a flap /ɾ/. We predict that misperceptions of intervocalic /t-d/ may cause confusions, thus motivating adaptations. We record native Canadian-English speakers (flappers) while playing a video game on Zoom in two conversation settings: with (1) a human non-flapper and (2) an AI non-

flapper (computer-generated speech). Acoustic analyses of the productions by human flapper speakers include features specific to stop-flap distinctions as well as global features (e.g., overall duration). In both human- and AI-directed speech, we expect human interlocutors to change flapped productions to stops to enhance intelligibility, particularly late in the conversation. Moreover, we expect differences between human- and AI-directed adaptations, with the former dominantly employing sound-specific features and the latter relying more on global hyperarticulation. Understanding these interlocutor-oriented adaptations may inform the technology behind human-computer interfaces.

5aSC10. A new experimental design to study speech adaptations in spontaneous human-computer conversations. Jetic Gu (School of Computing Sci., Simon Fraser Univ., 8888 University Dr., Burnaby, BC V5A1S6, Canada, jeticg@sfu.ca), Fenqi Wang, Ivan Fong, Samuel To (Dept. of Linguist, Simon Fraser Univ., Burnaby, BC, Canada), Dawn Behne (Dept. of Psych., Norwegian Univ. of Sci. and Technol., Trondheim, Norway), Allard Jongman (Dept. of Linguist., The Univ. of Kensas, Lawrence, KS), Joan Sereno (Dept. of Linguist., The Univ. of Kensas, Kansas City, KS), and Yue Wang (Dept. of Linguist., Simon Fraser Univ., Burnaby, BC, Canada)

Interest is growing for how human interlocutors make phonetic adaptations during spontaneous conversations. Given the increasing popularity of AI chatbots, research also needs to account for adaptations in human–computer interactions, an area under-investigated presumably due to methodological challenges in generating controlled conversational responses. Most studies involve scripted computer output, which may obstruct the dynamicity and the oral–aural medium of a natural conversation. To circumvent these constraints, we present a new experimental design that generates unscripted audio computer responses in human–computer conversations during a collaborative game played on Zoom. This design is unique in several aspects. First, the game (Escape Room) requires discussions on placing pictures (depicting target words/sounds) in specific locations, where misperceptions of target words between interlocutors may cause confusions, thus motivating natural adaptations. Second, to enable real-time computer responses, we adopt the wizard-of-oz paradigm typically used in the field of human–computer interaction, where a human confederate inputs text responses behind-the-scenes. Third, a programmable text-to-speech synthesizer converts the text input to audio output. The design demonstrated in this presentation opens the door to new analyses, tracking the dynamicity of speech adjustments over time. Moreover, it is generalizable to studying speech adaptations across interlocutor backgrounds.

5aSC11. Abstract withdrawn.

5aSC12. PrEgg: A free and open source Praat script for electroglottography measurements. May Pik Yu Chan (Dept. of Linguist, Univ. of Pennsylvania, 3401-C Walnut St., Ste. 300, C Wing University of Pennsylvania, Philadelphia, PA 19104-6228, pikyu@sas.upenn.edu) and Jianjing Kuang (Linguist, Univ. of Pennsylvania, Philadelphia, PA)

Electroglottography is a popular method of phonation analysis due to its non-invasive nature. In order to ease the extraction of systematic EGG measures, we wrote PrEgg, an open source Praat script designed to extract Contact Quotient (CQ), Skew Quotient (SQ) and Peak Increase in Contact (PIC) values. The script offers two measures for CQ and SQ, calculated using the EGG threshold method developed by Rothenberg (1988) and the DEGG method from Henrich *et al*. (2004), and PIC is further calculated from the DEGG signal. To test the validity of the script, we ran the Praat script on EGG data on Yi provided in the "Production and Perception of Linguistic Voice Quality" project at UCLA. We compared our results on CQ measurements using both the threshold method and the DEGG method with the provided CQ and CQ_{PM} measurements extracted from EggWorks, a UCLA software. Overall results are comparable between PrEgg and EggWorks. PrEgg is accompanied by a manual and available at https://github.com/maypychan/praat-egg.

5aSC13. Personality perception in synthetic versus natural speech: The effects of voice quality and prosody. Minjeong Kim (Graduate School of Culture Technol., Korea Adv. Inst. of Sci. and Technol., 291, Daehak-ro, Yuseong-gu, Daejeon 34141, Republic of Korea, minjeong.kim@kaist.ac.kr), Jaehan Park (KT Corp., Seoul, Republic of Korea), Minhong Jeong (Graduate School of Culture Technol., Korea Adv. Inst. of Sci. and Technol., Daejeon, Republic of Korea), and Jieun Song (School of Digital Humanities and Computational Social Sci., Korea Adv. Inst. of Sci. and Technol., Dajeon, Republic of Korea)

Synthetic speech technology, now approaching human-like naturalness due to advancements in deep learning, has turned to focusing on personality or persona design as a common practice in the industry. The present study aimed to identify speech characteristics affecting personality impressions in synthetic and natural speech. Thirty native Korean speakers participated in a personality rating experiment in which they evaluated natural Korean sentences and their synthetic counterparts in terms of the Big-Five personality model. Acoustic analyses were performed to examine voice quality and prosody, including Intonational Phrase (IP) boundary tones. The results revealed that scores of agreeableness, conscientiousness, and emotional stability increased overall when the voices contained greater aperiodicity in the harmonics (i.e., were likely breathier) and were weaker in energy. The results also demonstrated that different prosodic features affected personality perception in synthetic and natural speech; synthetic speech with a wider F0 range received higher scores on extroversion, openness, and emotional stability. In contrast, the effect of IP boundary tones was most frequently found for female natural speech, which contained a wider range of tones, including multitonals (e.g., LHL%). Our findings suggest that intonation is one of the key factors which can be adjusted to generate synthetic speech with various personalities.

5aSC14. Modeling the neural encoding of vowel formants in the midbrain. Daniel D. Pyskaty (Depat. Linguist, Neurosci., Univ. of Rochester, Rochester, NY 14627, dpyskaty@u.rochester.edu), Joyce M. McDonough (Linguist, Univ. of Rochester, Rochester, NY), and Laurel H. Carney (Biomedical Eng., Univ. of Rochester, Rochester, NY)

The first (F1) and second (F2) formants of a vowel determine its identity, making it essential to understand how these formant regions are encoded in the brain. We investigated the neural encoding of vowel formants using computational model responses of the auditory nerve and midbrain to natural vowel tokens [Hillenbrand et al., JASA 97, 3099 (1995)]. The first formant peak can be difficult to identify based on model population responses. Responses in the F1 region are more affected by the surrounding harmonic structure, whereas F2 often has a more straightforward representation in the model response. This result is surprising given the sensitivity of listeners for F1 discrimination compared to F2, reflecting Vowel Dispersion Theory [Liljencrants & Lindblom, Language 48, 839 (1972)], which favors the F1 dimension over F2. While peaks are difficult to identify, complex patterns of neural discharge rates observed across low-frequency midbrain responses may work to encode F1. These results suggest revisiting assumptions of the neural representations of formants and highlight the need for a system that evaluates overall response trends to formant regions instead of searching for formant peaks. Neural models can play an important role in understanding the structure of vowel systems. [Work supported by NIDCD-R01-010813.]

5aSC15. Collecting non-native English speech through a web-and mobile-based application. Bozena Kostek (Audio Acoust. Lab., Gdansk Univ. of Technol., Narutowicza 11/12, Gdansk 80-233, Poland, bokostek@audioakustyka.org)

The motivation behind this study is to collect non-native English speech recordings seamlessly. Even though numerous corpora containing non-native English speakers exist, there are not so many resources of Polish speakers uttering sentences in English. That is why a web-based application is built to record L2 English speakers. The same application is also developed in the mobile form as it provides easy access to the software created

and, at the same time, built-in microphones ensure sufficient quality of recordings. A set of sentences is prepared, including four main types, i.e., declarative, imperative, interrogative, and exclamatory. Recordings are saved in WAV audio format. Local data are first saved in the memory of the device on which the application is installed, and then, the user may listen to the recordings and then send them to the server using FTP (File Transfer Protocol). The user interface allows for English language level selection, by choosing one of the four possible options, i.e., basic, intermediate, advanced, and fluent. A group of volunteers provided speech samples, resulting in the collection of utterances from more than 600 speakers, available in the created corpus. Future development envisions adding additional modules that will allow for speech signal analyses.

5aSC16. Automatic detection of nasal closure and nasal release landmark acoustic cues. Janette Park (RLE, MIT, 50 Vassar St., Cambridge, MA 02139, janp@mit.edu), Jeung-Yoon Choi (MIT, Cambridge, MA), and Stefanie Shattuck-Hufnagel (RLE, MIT, Cambridge, MA)

This study describes the detection of nasal closure and nasal release landmarks, as part of a larger system for speech recognition based on acoustic cues. Landmarks are produced as a result of closures and releases in the oral region and are indicated by abrupt changes in the speech signal. Nasal closure and release landmarks have proven particularly challenging to detect and are the focus of this report. The process for implementing the nasal detection module includes extracting and processing a set of speech-related measurements, such as formant frequencies, spectral band energies, and their derivatives, from a large database of labeled speech files, and determining which of these measurements are potentially effective, using ANOVA analysis. Next, Gaussian mixture models are trained and tested on these measurements to classify nasal closures, nasal releases, and all other landmark cues. The resulting nasal closure and release landmark detection module will be used with other landmark modules for vowels, glides, fricative closures/releases, and stop closures/releases, as well as other acoustic cues to place and voicing, in the overall speech recognition system. The current performance of the module will be assessed and discussed.

5aSC17. Investigation of machine-learning-based stimuli for the remediation of children's speech errors. Kathryn Cabbage (Speech and Hearing Sci., Washington State Univ., 412 E Spokane Falls Blvd., Rm. 125P, Spokane, WA 99202, klcabbage@wsu.edu), Elaine R. Hitchcock (Commun. Sci. & Disord., Montclair State Univ., Bloomfield, NJ), Michelle T. Swartz (Speech Lang. Pathol., Thomas Jefferson Univ., Philadelphia, PA), and Thomas Carrell (Lincoln, NE)

Children with speech sound disorders (SSDs) demonstrate difficulty producing phonemes correctly and may exhibit poor speech perception compared to their age-matched peers; however, group differences in speech perception skills remain largely unexplained. Developmental models of speech production posit that children's ability to discriminate correct and incorrect productions in their own speech may be critical for developing accurate speech production. Historically, when children regularly mispronounce a phoneme, it has been essentially impossible to assess whether they perceive correct versus errors productions in their own speech, thus creating a clinical conundrum. How can we assess a child's ability to perceive the accuracy of their own phoneme production when they cannot produce a correct production? Recent technological developments allow for acoustic alteration of children's speech that digitally corrects speech sound errors while preserving natural characteristics of the child's voice. This machine-learning-based stimuli may then be used as training and feedback tokens for remediation when treating children with speech sound disorders. Such acoustic alteration is possible within an accessible, user-friendly environment that is clinically feasible for speech-language pathologists with little acoustic training. Thus, the purpose of this study is to evaluate the acoustic and perceptual accuracy of acoustically-altered child-speech compared to natural speech tokens.

5aSC18. Simulation of sentence-level speech with an acoustically driven model of speech production. Brad H. Story (Speech, Lang., and Hearing Sci., Univ. of Arizona, 1131 E. 2nd St., P.O. Box 210071, Tucson, AZ 85721, bstory@email.arizona.edu) and Kate Bunton (Speech, Lang., and Hearing Sci., Univ. of Arizona, Tucson, AZ)

In a model of speech production developed by Story and Bunton [JASA, 146(4), 2522–2528 (2019)] speech segments are encoded by specifying relative acoustic events along a time axis that consist of directional changes of the vocal tract resonance frequencies called resonance deflection patterns (RDPs). These events are transformed via acoustic sensitivity functions, into time-varying modulations of the vocal tract shape. For example, RDPs specifying /b, p/, /d, t/, and /g, k/ would typically be coded as $[-1\ -1\ -1]$, $[-1\ 1\ 1]$, and $[-1\ 1\ -1]$, respectively. The RDP changes, indicate, from left to right, the targeted directional shift of the first, second, and third resonances of the vocal tract. In addition, events that produce nasalization, voiced versus voiceless sounds, and changes in fundamental frequency are also specified the time axis. The purpose of this study was to demonstrate the use of this model to generate word- and sentence-level speech for eventual speech intelligibility studies. The process for simulating several words and sentences will be demonstrated.

5aSC19. Acoustic analysis as phonetic basis for phonological paraphasia in persons with aphasia: A preliminary study. J. Niranjana (Speech Lang. Sci., All India Inst. of Speech and Hearing, Kapila Ladies Hostel, AIISH, Mysore, Karnataka 570006, India, niranjanajkammath@gmail.com) and N. Hema (Speech Lang. Sci., All India Inst. of Speech and Hearing, Mysore, Karnataka, India)

Phonological planning deficit is considered to be the cause of phonemic paraphasia in persons with Aphasia. The errors are considered to be because of substitution of incorrect phoneme due to deficit in phoneme selection. However, the cascaded activation model suggests that phonemic paraphasia can result in the partial activation of the target and the competing phoneme that transcends to the lower articulatory level as a phoneme that reflects the acoustic properties of both sounds. Hence, this study aims to understand the existence of a phonetic basis for phonemic paraphasia in adults with Broca's Aphasia in the Tamil language. The study recruited five persons diagnosed with Broca's Aphasia in the age range of 50–60 years and five age and gender-matched controls. Confrontation naming task from WAB-R Tamil was used as the stimuli to elicit speech samples in affected individuals. All the recordings were done using PRAAT software. The burst duration and VOT of /p/ and /b/ sounds were compared between the experimental and control groups. Results suggested the presence of an acoustic trail of target sound in the substituted phoneme, challenging the existing literature on phoneme selection deficit at the planning level as the reason for the occurrence of phonemic paraphasias.

5aSC20. A module for automatic analysis of burst spectra for consonant place detection. Ella Tubbs (Res. Lab. of Electronics, Massachusetts Inst. of Technol., 77 Massachusetts Ave. Cambridge, MA 02139-4306, tubbs@mit.edu), Jeung-Yoon Choi, and Stefanie Shattuck-Hufnagel (Res. Lab. of Electronics, Massachusetts Inst. of Technol., Cambridge, MA)

Widespread methods for automatic speech processing have become increasingly powerful, but they do not aim to model aspects of human speech processing. Previous research has shown that individual acoustic cues are important in human speech perception. Developing a model that is capable of identifying individual acoustic cues in speech enhances our ability to extract meaningful information from a speech signal, irrespective of speaker variations or phonemic differences, in a way that provides a transparent and testable model of human speech processing. This research investigates a module for automatic analysis of the spectral burst cue in fricative and plosive speech sounds. The method determines the place of articulation of the spectral burst by utilizing spectral moment measurements near locations where a spectral burst is likely to occur as features in a Gaussian mixture model. This research lays the groundwork for a dynamic model that is developed to be consistent with the Bayesian belief updating framework, in alignment with prior work in human speech perception.

5aSC21. Effects of motor practice on the temporal coordination of articulatory movements for non-native onset clusters: Kinematic and acoustic evidence using electromagnetic articulography. Allen Shamsi (Linguist, Univ. of Florida, 289 Corry Village, Apt 10, Gainesville, FL 32607, allenshamsiev@ufl.edu), Matthew Masapollo (Psych., McGill Univ., Montreal, QC, Canada), Rachel Meyer, and Ratree Wayland (Linguist, Univ. of Florida, Gainesville, FL)

Research on cross-language speech production has shown that part of the challenge of producing non-native clusters arises from difficulties with temporally coordinating the successive consonantal gestures within a cluster. However, it remains unclear whether the application of practice-based motor learning paradigms can improve or stabilize this aspect of non-native cluster production. This study uses electromagnetic articulography (EMA) to measure the effects of motor practice on the temporal coordination of articulatory movements for non-native onset clusters. Monolingual speakers of American English intensively practiced producing monosyllabic sequences containing non-native onset clusters (e.g., MGAT) over two consecutive days. EMA was used to capture lingual, labial, and jaw motion during successive repetitions. For properly-sequenced cluster repetitions, analysis of the EMA sensor trajectories showed that, over the course of practice, there was a general reduction in inter-gestural timing variability and increased overlap between adjacent consonantal gestures. Furthermore, some of these improvements were maintained to the second day, indicating that subjects started forming durable performance gains. Acoustic analyses also showed a reduction in the duration of epenthetic vowel errors (/mgæt/ → /məgæt/) produced throughout practice. Collectively, the findings suggest that motor practice can improve the temporal coordination of articulatory gestures affiliated with non-native clusters.

5aSC22. Developing an algorithm to characterize context-based speech patterns as cue production profiles. Sofie C. Chung (Massachusetts Inst. of Technol., 50 Vassar St., Cambridge, MA 02139, scchung@mit.edu), Jeung-Yoon Choi, and Stefanie Shattuck-Hufnagel (MIT, Cambridge, MA)

The surface articulation of an underlying phoneme commonly fluctuates depending on the context. For instance, the associated acoustic cues for a /t/ phoneme produced at the beginning of an utterance may be distinct from the acoustic cues of a /t/ phoneme that is preceded and followed by a vowel. Thus, for an individual speaker or speaker group, we can investigate what specific phonemic contexts result in the associated acoustic cues to be produced. An algorithm has been developed which matches a target phoneme to its corresponding acoustic cues. For a given database, we can recover all of the contexts in which a phoneme was produced and tabulate the various acoustic cue production patterns that arose. Finally, we label these patterns by their speech production type (e.g., standard production, flapping, etc.) and produce an output in the form of a cue production profile. As a result, this profile would allow for the association of a set of acoustic cues in specific phonemic contexts for an individual speaker or speaker group, such as speakers from a specific dialect region. The algorithm can also be expanded to include prosodic cues to account for how stress and intonation can affect phoneme production.

5aSC23. Automatic detection and labeling of glides for the English and Spanish language. Edgar Morfin (MIT, 50 Vassar St., Cambridge, MA 02139, emorfin@mit.edu), Jeung-Yoon Choi, and Stefanie Shattuck-Hufnagel (MIT, Cambridge, MA)

This study outlines the development of a module for automatic detection of glide landmarks. We define glide landmarks as acoustic events that are observed during a narrowing in the oral cavity that does not result in cessation of airflow or in the conditions for turbulence noise. Glide landmarks are commonly associated with standard productions of underlying glides, a set that includes semivowels, liquids, and sounds produced by narrowing at the glottis, such as glottal stops, or the aspirant /h/. We lay out a framework that can be used to determine the acoustic measurements that are useful for detecting glide landmarks, and a Gausian Mixture Model is trained and tested, for automatic detection. A first investigation is carried out for /r/ production in English and Spanish. A closer look reveals that acoustic cues that describe abrupt onsets and offsets is useful for describing the trilled /r/ in Spanish, resulting in the need for an additional set of glide landmark cues for glide closure and glide release.

5aSC24. Analysis of physiological measures around conversational state changes. Benjamin Masters (Systems Design Eng., Univ. of Waterloo, 200 University Ave W, Waterloo, ON N2L 3G1, Canada, bpmasters@uwaterloo.ca), Susan Aliakbary Hosseinabadi, Dorothea Wendt (Eriksholm Res. Ctr., Snekkersten, Denmark), and Ewen MacDonald (Syst. Des. Eng., Univ. of Waterloo, Waterloo, ON, Canada)

The goal of this work is to extend the use of physiological measures of listening effort to interactive conversation. The initial work here investigates variations in head movement, eye gaze, and pupil dilation around conversational state changes during task-based conversations. Here, conversational state changes are defined as the points in time at which speakers start and stop talking. Windows around each of these types of state changes are analyzed for systematic differences of these parameters, which could be indicative of changes in attention and/or differences in speech production versus perception. Additionally, we calculate state change response functions, derived from a multivariate regression that maps from the state changes to the measured parameters and extracted features. The predictive power of these functions is explored, alongside comparisons of various considerations in their derivations. Our findings, based on data collected from 12 sets of interactive conversations taking place in varying levels of noise and simulated hearing loss, offer insight into how physiological responses during complex interactions can be measured and interpreted to infer when and where effort is directed throughout conversation.

5aSC25. Comparison of spectrograms and continuous wavelet transforms for multi-class classification of vocal pathologies by using convolutional neural networks. Bhawna Rathi (Music Technol., IUPUI, 535 W. Michigan St., IT371, Indianapolis, IN 46202, brathi@iu.edu) and Timothy Hsu (Music and Arts Technol., Indiana Univ. - Purdue Univ., Indianapolis, Indianapolis, IN)

The emergence of artificial intelligence has encouraged researchers to explore non-invasive methods for classifying vocal disorders of different pathologies. This paper introduces a framework designed to classify multiple distinct vocal pathologies, building upon the groundwork laid by the previous two-class classification research. This study uses datasets obtained noninvasively from the Indiana University (IU) Health Voice Center, implementing a Convolutional Neural Network (CNN) algorithm to differentiate vocal pathologies. In this approach, continuous wavelet transforms (CWT) and spectrogram images are derived from audio files of pathological voices and serve as inputs for the CNN classifier. Additionally, the research investigates the effects of data augmentation and the integration of a dropout layer, exploring how these variations affect the results. The findings reveal differing accuracy levels associated with parameter adjustments, such as learning rates and the different number of filter layers. Notably, CWT is found to yield higher accuracy compared to spectrograms. Furthermore, the study has achieved a high accuracy of 96% in multiclass classifications of diverse vocal pathologies within the training dataset, which underscores previous research advancements, demonstrating the feasibility of multiclass classifications.

5aSC26. Predicting vocal tract shape information from tongue contours and audio using neural networks. Sarah R. Li (Biomedical Eng., Univ. of Cincinnati, University of Cincinnati, 231 Albert Sabin Way, CVC, 3960, Cincinnati, OH 45267, lisr@mail.uc.edu), Alex Knapp, Jing Tang (Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH), Suzanne Boyce (Commun. Sci. and Disord., Univ. of Cincinnati, Cincinnati, OH), and T. Douglas Mast (Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH)

Midsagittal ultrasound imaging of the tongue is a portable and inexpensive way to provide articulatory information. However, although ultrasound images show a portion of the tongue surface, other vocal tract structures (e.g., palate) are not typically visible. This missing information may be useful for speech therapy and other applications, e.g., by characterizing vocal tract constrictions and informing how morphological variations affect speech patterns. Prediction of the vocal tract shape from information

available during ultrasound imaging (e.g., tongue contours and audio recordings) is, thus, potentially valuable. Recent advancements in articulatory prediction from audio recordings (i.e., acoustic inversion) and speech recognition using combined articulatory and acoustic data have used neural network models. Inspired by these models, this study investigates how well fusion of articulatory and acoustic features in speaker-independent models can predict expanded articulatory information. Specifically, recurrent neural network models will be trained to predict the vocal tract shape based on partial tongue contours and acoustic features, during production of vowels and central approximants. Features will be extracted from simultaneously recorded audio and 2D MRI (USC 75-Speaker Database). Different acoustic features and network architectures will be compared, with the goal of refining future models to predict vocal tract shapes during ultrasound imaging.

5aSC27. The effects of three-way consonant distinction in Northern Saami. Emily Posson (Linguist, Univ. of Minnesota - Twin Cities, 1000 SE 8th St., Apt. 10, Minneapolis, MN 55414, posso011@umn.edu) and Christopher Geissler (Linguist, Carleton College, Northfield, MN)

Of the world's languages, very few have been attested to possess a three-way consonant length distinction, among them Northern Saami (Uralic). However, the effects of this rare contrast on adjacent segments remain unexplored. This study reports on dialectal variation within Northern Saami and sheds light on the relationship between consonant length and the duration of preceding vowels. Three speakers were recorded: one Western Guovdageaidnu speaker and two Eastern Kárášjohka speakers. Target words were bisyllabic of the form $CV_V(C)$, with a medial short, long, or extra-long consonant, and were elicited in frame sentences. Results show dialectal differences in the number of consonant length distinctions, as well as the effects on preconsonantal vowels. The Western dialect exhibits three surface consonant lengths, while the Eastern dialect only has the longest and shortest. Interestingly, the Eastern speakers' extra-long C are systematically longer than their Western counterparts. Preconsonantal vowels vary with consonant length in the Western dialect, while Eastern preconsonantal vowels present a broader range of contrasts. These findings suggest that the Western dialect resembles other Saami languages with three consonant lengths (e.g., McRobbie-Utasi 2007). In comparison, the Eastern dialect may instantiate some of this contrast on the preceding vowel instead.

5aSC28. Clinical applications for automatic detection of creaky voice. Sarah R. Bellavance (Dept. of Commun. Sci. and Disord., New York Univ., 665 Broadway, 6th Fl., New York, NY 10012, srb664@nyu.edu) and Aaron Johnson (Dept. of Otolaryngol.-Head and Neck Surg., New York Univ. Grossman School of Medicine, New York, NY)

Creaky voice is a voice quality in which a low amount of subglottal air pressure, a condensed vocal fold structure, and a high closed quotient of vibration combine to create the auditory percept of a series of pulses at a low pitch. While this voice quality is often nonpathological, it can also co-occur with vocal pathologies. Identification of creak in the speech signal is most often done manually. Automatic creak detection algorithms have been created to streamline and produce replicable workflows. These algorithms have steadily increased in reliability, with COVAREP (Degottex *et al*., 2014) as the newest state-of-the-art. While preliminary studies have demonstrated promising findings using artificial neural networks with clinical data, artificial neural networks typically improve with diverse data testing. The current study implements COVAREP on a novel dataset, both in terms of speakers and speech types. Deidentified patient diagnoses were matched to audio recordings collected from January 2021 through September 2023. Relevant portions of audio recordings were extracted using a Praat script, and COVAREP was implemented on the extracted audio files in MATLAB. Ongoing analyses correlating percentage of creak detected and vocal pathology diagnoses will be discussed. Finally, the results will be compared to those of previous work.

5aSC29. Comparing face-tracking action units with electromyography during speech. Hastiossadat Nozadi (Linguist, Univ. of BC, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, hnozad01@student.ubc.ca), Emma Irwin, Jamie Cheung, Yadong Liu, and Bryan Gick (Linguist, Univ. of BC, Vancouver, BC, Canada)

Video face-tracking software, such as OpenFace 2.0, can be used to make inferences about facial muscle activation [Baltrušaitis *et al*. IEEE 13, 59–66 (2018)]. However, the accuracy of these inferences based on the facial action units (FAUs) calculated by OpenFace 2.0 compared to corresponding muscle activity during speech is unclear. A previous study investigated muscle activation when both smile and speech occur simultaneously, focusing on the zygomaticus major (ZM) muscle [Liu *et al*. ISSP, 130–133 (2021)], but only presented data for a single speaker and did not compare FAU and EMG results. The present study compares OpenFace 2.0 action units with surface electromyography (EMG) data during speech in order to assess the validity of these inferences about facial muscle activation. We compare ZM activity and the lip corner puller FAU intensity results from a dataset collected for the previously mentioned [Liu *et al*. 2021] study. Data include four speakers producing read speech in smile conditions; 2 s will be extracted from EMG and FAU data before and after each utterance. Results will be reported on the relative accuracy of the FAU and EMG data. Implications will be discussed for speech communication research. [Work supported by NSERC.]

5aSC30. Feature generalizability for speaker-dependent detection of alcohol intoxication. Xinglei Liu (Tenvos Res. Labs, Sacramento, CA), Arian Shamei (Linguist, UBC, Vancouver, BC, Canada), and Rima Seiilova (Tenvos Res. Labs, Sacramento, CA, rima@tenvos.com)

Impairments to speech motor control from alcohol intoxication are variable across individuals, making speaker-dependent approaches ideal for speech-based intoxication detection [Schiel *et al*., 2010. Proc. INTERSPEECH 2010]. Here, we evaluated whether individual acoustic features have high generalizability across speaker-dependent models. We selected 97 speakers (54 male, 43 female) from the Alcohol Language Corpus [Schiel *et al*., 2012. LRE. 46, 503-521] who had sufficient sober and intoxicated (>0.08% blood-alcohol concentration) recordings for speaker-dependent modeling. For each speaker, we extracted 9 features from vowels (F0–F3, jitter, shimmer, harmonics-to-noise ratio, duration, and duration variability) and 7 from consonants (spectral skewness and kurtosis, center of gravity, duration and duration variability, harmonics-to-noise ratio), and fitted these to speaker-dependent random forest models with 5-fold cross-validation to evaluate feature importance from the associated mean decrease in Gini impurity (GI). Across all speakers, consonant-based features tended to have stronger generalizability than vowel-based features, with spectral skewness and kurtosis being the most generalizable (GI: 0.11 and 0.09), and vowel duration and F2 being the least generalizable (GI: 0.04 and 0.03). Further experiments to explore additional features and evaluate sex-specific generalizability are ongoing. [Research funded by Tenvos Incorporated for the development of commercial speaker state-detection algorithms.]

5aSC31. Real-time magnetic resonance imaging of velopharyngeal port posture during long pauses in naturalistic speech. Joshua O. Diebel (Linguist, Univ. of BC, 2511 East 2nd Ave., Vancouver, BC V5M1C7, Canada, joshdiablo135@gmail.com), Bryan Gick, Jahurul Islam (Linguist, Univ. of BC, Vancouver, BC, Canada), and Jade Weinstein (Speech Sci., Univ. of BC, Vancouver, BC, Canada)

Velum's behavior as a postural substrate in naturalistic speech is not well understood and is largely based on studies of inter-utterance velopharyngeal port posture (VPP). While existing literature [e.g., Gick *et al*., 2004 Phonetica, 61(4); Ramanarayanan, 2013 JASA, 134(1)] describes posture variations based on rest positions, ready positions, and inter-speech pauses, a less explored aspect is the behavior of the velum during extended pauses in naturalistic speech. Addressing this gap, the present study analyzes velum posture during naturalistic speech to characterize VPP during prolonged pauses. This study draws from a corpus of real-time magnetic resonance imaging (rtMRI) videos of L1 English speakers including those engaging in unstructured speech tasks. Results based on sequences corresponding to long pauses in naturalistic speech will be reported, outlining VPP

characteristics, and quantifying findings in terms of time spent in and shifts between velum positions. Implications for motor control differences in naturalistic and elicited speech data will be discussed.

5aSC32. Exploring facial gestures and visual speech articulation cues during the production of Canadian English voiced stops. Theresa Rabideau (Linguist, Univ. of Ottawa, 25 Elterwater Ave. Apt #14, Ottawa, ON K2H5J1, Canada, trabi037@uottawa.ca) and Suzy Ahn (Dept. of Linguist, Univ. of Ottawa, Ottawa, ON, Canada)

Decades of research on visual information for speech has demonstrated the informativeness of visual cues to enhance (Cho *et al*., 2020; Kawase *et al*., 2014) and influence (MacDonald and McGurk, 1978) speech perception. However, it is unclear which specific visual cues are spatially and temporally correlated to certain features of speech segments. This study explored visual cues of voicing during the production of Canadian English stops using facial recognition technology (Baltrušaitis *et al*., 2018) and manual coding (using ELAN 2022). We recorded the audio along with two videos, capturing both front and side views simultaneously, from six native Canadian English speakers. We paid special attention to the throat (larynx), chin, and neck areas which have been understudied by the previous literature. Preliminary data shows expanding movement in the submental triangle and throat during the production of voiced stops compared to voiceless stops. This finding supports tongue body lowering and larynx lowering found in the production of English voiced stops (Westbury, 1983). The comparison between utterance initial stops with post-vocalic stops shows that certain visual cues may be related to phonological voicing categorization irrespective of actual voicing during closure, while others reflect phonetic voicing reality.

5aSC33. Performance analysis of a dilated attention fast GAN for speech enhancement. Vahid Ashkani (Western Univ., 14-534 Platt's Ln., London, ON N6G 3A8, Canada, vashkani@uwo.ca) and Vijay Parsa (Western Univ., London, ON, Canada)

Recent advancements in speech enhancement have witnessed the emergence of generator-based methodologies. However, several of these approaches exhibit complexity in handling input variations, either excelling at low signal-to-noise ratios (SNRs) by utilizing intricate representations of noisy and clean speech or demonstrating superior performance only at higher SNRs. In this work, we investigated speech enhancement using a Dilated Attention Fast Generative Adversarial Network (DAF-GAN). The proposed DAF-GAN framework achieves stability in performance across different SNR conditions by efficiently processing large-scale signal lengths. The DFS-GAN features a dilated discriminator model operating via patches. The generator architecture incorporates multi-decoding and attention gates facilitated through skip-connections, strategically integrated within the Fast-U-Net model to optimize processing speed. An ideal ratio mask was used in the test phase to further refine the enhanced signal by emphasizing target speech while suppressing residual noise or artifacts. The DAF-GAN performance was assessed using objective metrics such as PESQ on a number of noisy speech databases. Results revealed that the DAF-GAN performed modestly in comparison with the state-of-the-art models. For example, analyses of the VoiceBank-DEMAND dataset yielded a PESQ score of 2.50 for the DAF-GAN.

5aSC34. Influence of airflow and ligament tension on the acoustics of a biomimetic larynx model. Bogac Tur (Phoniatrics, Univ. Hospital Erlangen, Waldstrasse 1, Erlangen 91054, Germany, bogac.tur@uk-erlangen.de), Lucia Gühring, Olaf Wendler, and Stefan Kniesburges (Phoniatrics, Univ. Hospital Erlangen, Erlangen, Bavaria, Germany)

The phonation process is a complex interaction involving the airflow from the lungs, the oscillation of the vocal folds' tissue, and the resultant acoustics. To understand the underlying physical mechanisms, an experimental model has been designed that allow the control of flowrate and longitudinal tension of the vocal folds. A synthetic biomimetic larynx model was applied that features airflow-driven vocal folds' oscillations. The longitudinal stiffness of the vocal folds is controlled using embedded ligament fibers. The model enables to measure aerodynamic and acoustic signals as function of airflow rate and the fiber tension. Based on the measured signals, the influence of airflow and fiber tension was statistically analyzed using the parameters F0, Psub, CPP, HNR, Shimmer, and Jitter. The statistical analysis revealed that both flowrate and fiber tension significantly influence acoustic and aerodynamic parameters. In general, the parameters showed different trends for increasing flowrate and fiber tension. Increase in fiber tension produced increasing parameters up to a maximum tension level followed by a saturation of the parameters. In contrast, the flowrate showed varying trends depending on the respective parameter. The results clearly show how flowrate and longitudinal tension control the phonation process and the resulting acoustic quality.

5aSC35. Analysis of potential influencing factors for acoustic quality in patients with ectodermal dysplasia. Marion Semmler (Div. of Phoniatrics and Pediatric Audiol., Univ. Hospital Erlangen, Waldstrasse 1, Erlangen 91054, Germany, marion.semmler@uk-erlangen.de), Bogac Tur, Ludger Schlautmann, Sophie Wolfsteiner, Laura Ziller, Ann-Katrin Hellmann, Maximilian Eckhardt, Olaf Wendler, and Anne Schützenberger (Div. of Phoniatrics and Pediatric Audiol., Univ. Hospital Erlangen, Erlangen, Bavaria, Germany)

Subjects with ectodermal dysplasia (ED) suffer from an inherited disorder in the development of ectodermal structures. They show a significantly reduced formation of teeth, hair, and a reduced number and activity of sweat and salivary glands. Recently, the voice of ED subjects has come into focus. It is assumed that the generally reduced glandular function is responsible for the altered vocal sound, although no specific cause has been identified. Previous findings included significant changes in the acoustic quality, the parameters derived from high-speed videoendoscopy, the laboratory analysis and rheological analysis of saliva samples as a substitute for laryngeal mucus. Based on these findings, we performed an extended statistical analysis directed towards the correlations between the different affected domains in order to reveal the contributing factors for the resulting acoustic voice quality. Although there were distinct statistical differences between ED males and male controls in the individual domains, only few statistical correlations were found between the subdomains i.e., acoustics, vocal fold vibrations and saliva composition/consistency. These results do not yet allow any definitive conclusions on the influencing factors of voice quality. A larger group of test subjects and analyses of the laryngeal mucus are required.

5aSC36. Comparing speech to fine and gross motor skills in Parkinson's patients. Brian Diep (Linguist, Univ. of BC, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, brdiep@mail.ubc.ca), Sylvia Cho (Linguist, Simon Fraser Univ., Burnaby, BC, Canada), Arian Shamei, and Bryan Gick (Linguist, Univ. of BC, Vancouver, BC, Canada)

Parkinson's Disease (PD) is a neurodegenerative motor disorder resulting from damage to dopaminergic neurons. The goal of this study is to evaluate the correspondence between speech and non-speech motor impairments. To explore this, we extract features from the mPower dataset [B. M. Bot *et al*., Sci Data 3, 160011 (2016)] containing mobile data from PD patients and healthy controls along with their performance on a vowel phonation, finger tapping, and walking task. We hypothesize that there is a shared motor system underlying each of these modalities and that disease progression will manifest in impairments to both speech and non-speech systems that rely on motor control. For acoustic features, we measure temporal consistency via F0-independent features (shimmer, jitter, and harmonics-to-noise ratio). For non-acoustic tasks, we adapt this set to measure spatial consistency and accuracy in finger tapping or walking. We perform clustering and multidimensional scaling (MDS) on our features to understand their correspondence across the modalities. Results will be reported with relevance to the relationship between PD and its effects on articulatory and general motor processes.

5aSC37. Using prosody to produce trust and doubt. Abbey L. Thomas (Brain and Behavioral Sci., The Univ. of Texas at Dallas, 800 W Campbell Rd., Richardson, TX 75080, abbey.thomas@utdallas.edu)

Work presented at a 2022 ASA meeting [Thomas, 2022] demonstrated that American English talkers use acoustic prosodic variables to distinguish attitudes of incredulity and trust from a "neutral tone of voice" when asking

WH-questions. The present study largely replicates these findings in new data. These data comprise both WH-questions and Yes/No questions, read by a new set of talkers in six imagined scenarios. In the previous study, "doubt" was likely confounded with "authority" due to the imagined scenario used to elicit the stimuli. In the present study, the three attitudes (doubt, neutral, trust) were crossed with dominance (participant was +/− dominant relative to their imagined conversational partner). This study examines additional acoustic variables, including the variability of word duration within utterance and the F0 slope of the longest word in each utterance. Averaging across the +/− dominance conditions and the utterance type (yes/no or WH question), duration-based variables reliably distinguished doubting from neutral utterances (with doubting utterances showing greater duration and greater variability in word duration). In contrast, F0-based variables differentiated trusting from neutral utterances, with trusting utterances showing a higher F0 over the course of the entire utterance.

5aSC38. Broad and fine acoustic categories in *bod*, *bond*, *bald*, and *bard*: A step toward acoustic phonology. Matthew C. Kelley (English, George Mason Univ., 4400 University Dr., Fairfax, VA 22030, mkelle21@gmu.edu)

Acoustics is central to the study of speech communication, but it is conspicuously under-represented in abstract representations of speech. Many flavors of phonological analysis tend toward articulatory descriptions, and transcriptions focus on strings of articulatory actions. All this is despite acoustics being easier to measure than articulation with current technology. The present study explores basic concepts for an acoustic phonology, with two types of postulated categories: broad and fine. Resonant, turbulent, transient, and occludent types of sounds comprise the broad categories, as general methods of filtering the speech source. Fine categories are conceptualized as specific types of acoustic actions within a broad category. These acoustic actions are goal-oriented, as for achieving a particular acoustic effect like the presence of antiformants or a lowered F2 or F3. However, these actions are not explicitly restricted to manipulating traditional phonetic features like formants. By default, fine categories are assumed to be produced in parallel when possible, yielding overlap effects like anticipatory nasalization, lateralization, and rhotacization. These concepts are explored in a microanalysis of *bod*, *bond*, *bald*, and *bard* from the speaker in the Massive Auditory Lexical Decision data set, with an eye to seeding the ground for a future acoustic phonology.

5aSC39. Respiratory laryngeal coordination during vowel-plosive-vowel transition in shouted speech. Zhaoyan Zhang (UCLA School of Med., 1000 Veteran Ave., 31-24 Rehab. Ctr., Los Angeles, CA 90095, zyzhang@ucla.edu)

This study investigates differences in respiratory-laryngeal coordination between normal and shouted speech in the production of a vowel-plosive-vowel utterance, with the goal toward synthesis of such utterances in loud speech. The results showed that compared to normal condition, shouted speech produced higher intraoral pressure during vocal tract closure, higher airflow at the release of vocal tract closure, increased fundamental frequency and increased duration of glottal closure during the following vowel, and increased respiratory activities. For unvoiced plosives, shouted speech also produced increased voice onset time of the following vowel, but its importance to the perception of shouted speech was small. Computational simulations further showed that such increased voice onset time is likely due to increased maximum vocal fold abduction and/or delayed initiation of vocal fold adduction, which are necessary to avoid voice onset immediately after the release of vocal tract closure in conditions of high subglottal pressure.

5aSC40. Principal dimensions of laryngeal vocal control in a computational model of voice production. Zhaoyan Zhang (UCLA School of Medicine, 1000 Veteran Ave. 31-24 Rehab Ctr., Los Angeles, CA 90095, zyzhang@ucla.edu)

Voice production is constrained by laryngeal physiology and physics. Such constraints may present themselves as principal dimensions that are shared among speakers in how they produce and perceive voice. In this study, we attempt to identify such principal dimensions in voice outcome measures and the underlying laryngeal control mechanisms in a three-dimensional computational model of voice production. A large-scale voice simulation was performed with parametric variations in vocal fold geometry, stiffness, glottal gap, and subglottal pressure. Principal component analysis was applied to data combining both the laryngeal control parameters and acoustic and aerodynamic voice outcome measures. The results showed two dominant dimensions of vocal control. The first dimension describes interaction between respiration and vocal fold adduction in a way that increases glottal flow amplitude and vocal intensity at the cost of decreasing high-frequency harmonic production. The second dimension mainly describes control of medial surface thickness and glottal closure, which allows simultaneous increase in vocal intensity and high-frequency harmonic production but also increases risk of vocal fold injury. A third dimension with a reduced weight describes control of the fundamental frequency. The importance of these principal control dimensions to vocal expression of emotion and vocal health is discussed.

5aSC41. Spectral energy properties of falsetto voice. Yuan Chai (Linguist, Univ. of Washington, Guggenheim Hall 4th Fl. Box 352425, Seattle, WA 98105, yuanchai@uw.edu) and Patricia A. Keating (Linguist, UCLA, Los Angeles, CA)

Breathy voice and falsetto voice have distinct articulatory mechanisms and auditory percepts. They clearly differ in their noise spectrum and typical f0. However, previous literature has described the harmonic energy profile of each in similar terms, namely high energy in H1 and low energy in higher-frequency harmonics. Are the harmonic spectra of these two voice types in fact the same? We will provide a quantitative comparison of the harmonic energy in falsetto versus breathy (and also modal and creaky) voice using a corpus of readings by Taiwan Mandarin speakers of a "Little Red Riding Hood" story where they enacted character voices. We annotated each vowel in this corpus for its perceived phonation type. In raw values, H1*−H2* is the same in falsetto and breathy voice, but when f0 is controlled statistically, H1*−H2* is lower in falsetto than in breathy voice (falsetto has less falloff from H1 to H2). H1*−H5k*, a measure of overall spectral tilt, is the same in falsetto and breathy voice when f0 is controlled, though falsetto voice has overall higher energy than breathy. Other harmonic differences will be presented in detail, along with comparisons on other voice measures.

5aSC42. Duration imitation is not mediated by phonological contrast: Evidence from a checked-unchecked tonal contrast in Taiwanese Southern Min. Wei Zhang (McGill, 3620 Rue Lorne Crescent, 318, Montreal, QC H2X2B1, Canada, weizhang201707@gmail.com), Meghan Clayards (McGill, Montreal, QC, Canada), and Yu-An Lu (National Yang Ming Chiao Tung Univ., Taipei, Taiwan)

Phonetic imitation is mediated by phonological contrast, as evident in features such as formant, VOT and F0. However, a recent study observed that duration imitation was not mediated by phonological contrast. In contrast to other studies, duration served as a non-primary cue to the phonological contrast in this recent study. This current study further investigates duration imitation in a case where duration serves as the primary cue. We utilized the tonal contrast of T3 versus T33 in Taiwanese Southern Min (TSM), to which duration was identified as the primary cue. We created a seven-step tonal continuum between T3 and T33 by manipulating the tone durations, and recruited seventeen native TSM speakers to imitate each step as closely as they could. The bi- or uni-modality of the distribution of the imitated durations for all seven steps was analyzed using Bayesian regression models. Results showed that the imitated durations were more consistent with an unimodal distribution, suggesting that, unlike other features, the imitation of duration is not mediated by tonal contrast, whether it acts as a primary cue or not. Thus, features exhibit different resistance to phonological mediation in phonetic imitation.

5aSC43. You good?: Examining the role of intonation and eyebrow movements in sentence type distinction. Kendall Lowe (Linguist., Univ. of Michigan, 611 Tappan St., Ann Arbor, MI 48109, loweke@umich.edu), Yoonjeong Lee, Jelena Krivokapić, and Natasha Abner (Linguist., Univ. of Michigan, Ann Arbor, MI)

By analyzing the pitch accents, edge tones, and eyebrow movements of African American English (AAE) speakers, this study examines the coordination of prosody and co-speech gestures. Twelve self-identified AAE speakers engaged in scripted dialogues with an AAE-speaking confederate via Zoom. Half of the dialogues contained the polysemous AAE idiomatic phrase "you good" which has declarative and interrogative forms hypothesized to be distinguished by various eyebrow and intonation patterns. The other dialogues included non-idiomatic phrases semantically related to their "you good" equivalents. These dialogues were designed to explore whether the coordination patterns for the idioms occur with non-idiomatic utterances of the same sentence type. Acoustic data is prosodically annotated using the MAE-ToBI transcription system and visual eyebrow data is tracked using OpenFace 2.0, a facial landmark detection toolkit. The derived eyebrow movement trajectories are time-aligned with the pitch trajectories for the semi-automatic labeling of multimodal signals. The temporal relationship between eyebrow movement landmarks, pitch accents, and edge tones is examined to investigate how different meanings and sentence types are created through their coordination. Preliminary examination indicates intonationally similar contours for the idiomatic and non-idiomatic interrogative phrases, but shared and consistent intonational contours are not evident in the declarative data.

5aSC44. The impact of remote microphones and facial masks on speech production and conversational behaviors in hearing-impaired individuals. Menatalla K. Ellag (Syst. Des. Eng., Univ. of Waterloo, 200 University Ave W, Waterloo, ON N2L3G1, Canada, mkellag@uwaterloo.ca), Jinyu Qian, Ieda Ishida (Sonova Canada, Mississauga, ON, Canada), and Ewen MacDonald (Syst. Des. Eng., Univ. of Waterloo, Waterloo, ON, Canada)

This study investigates the acoustics and conversation behavior of the speech produced during conversations among groups of hearing-impaired individuals. Four groups of four hearing-impaired individuals, all using hearing aids, engaged in discussions on provided topics in the presence of background noise. Conversations were held in four conditions based on two factors (using versus not using a remote microphone; wearing versus not wearing a face mask). Analysis of recorded conversations focused on speech production measures (e.g., fundamental frequency, articulation rate, formant frequencies, etc.) and conversational behaviors (e.g., inter-pausal unit length, floor-transfer offsets, turn duration, etc.). Although both influence the potential difficulty of holding a conversation, distinct effects of mask, remote microphone, and their interaction were observed for measures of speech production and conversational behaviors.

5aSC45. The effect of task on speech production and conversation behavior during conversation. Menatalla K. Ellag (Syst. Des. Eng., Univ. of Waterloo, 200 University Ave. W, Waterloo, ON N2L3G1, Canada, mkellag@uwaterloo.ca), Kate Avison, and Ewen MacDonald (Syst. Des. Eng., Univ. of Waterloo, Waterloo, ON, Canada)

The purpose of this study was to investigate how native-English, healthy-hearing individuals adapt their speech production and conversation behavior in the presence of noise and how this can vary based on conversational goal. Pairs of participants engaged in both free-form conversations as well as conversations based on solving a task (a "spot the difference" task using the Diapix UK pictures). Although seated in separate rooms, talkers could communicate via headset microphones and headphones with gains set to simulate levels that would be present if they were seated in the same room. The effects of task and noise on measures of speech production (e.g., articulation rate, speech level, etc.) and conversational behaviors (e.g. floor transfer offsets, turn length, etc.) are investigated. These results provide insights into how to infer listening effort via acoustical measures of communication in a broader range of settings.

5aSC46. Voice analysis for intoxication detection in laboratory versus law enforcement contexts. Arian Shamei (Linguist, UBC, 2613 West Mal, Vancouver, BC V6T 1Z4, Canada, arianshamei@gmail.com), Xinglei Liu, Rima Seiilova (Tenvos Res. Labs, Sacramento, CA), and Bryan Gick (Linguist, UBC, Vancouver, BC, Canada)

There is substantial interest in deploying voice biomarkers for the detection of alcohol intoxication, yet it remains unknown how other mental and physical states (e.g. emotion, stress) influence voice biomarkers in intoxicated speech. We compared measurements of voice quality (jitter, shimmer, noise-harmonics ratio) across two datasets of alcohol-intoxicated speech: (1) The alcohol language corpus, which contains laboratory elicited speech from 167 individuals in both sober and intoxicated conditions and (2) a custom dataset of police (control, $n = 14$) and suspect (intoxicated, $n = 32$) interactions during traffic stops where intoxication was verified via breath analysis. Measurements were extracted from all stressed vowel tokens and compared across conditions using two-sample t-tests within sex-specific groupings of each dataset. For both males and females, jitter was significantly lower during intoxication as measured from laboratory-elicited speech, but significantly higher for intoxicated individuals when measured from police–suspect interactions. These results suggest that voice biomarkers for alcohol-intoxication are easily confounded by other emotional and physical states (e.g., stress during police interaction), and thus, present a particular challenge for speaker-independent detection systems where baseline voice quality measurements across different emotional states are unknown. [Research funded by Tenvos Incorporated for the development of commercial speaker state-detection algorithms.]

Session 5aUW

Underwater Acoustics: Underwater Sonar and Communication

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Contributed Papers

8:00

5aUW1. Sonar image generation using time-domain scattering echo simulated by Kirchhoff approximation extensions. Yan Wu (Shanghai Jiao Tong Univ., No. 800, Dongchuan Rd., Minhang District, Shanghai 200000, China, wuyan01@sjtu.edu.cn) and Jun Fan (Shanghai Jiao Tong Univ., Shanghai, China)

Sonar images are a topic of great interest in many underwater applications, such as detecting and classifying proud targets. Given the high cost of experimental measurement and the lack of measurement data, especially the difficulty in obtaining non cooperative target image data, a sonar image generation method of target and seabed joint modeling based on echo simulation is proposed in this work. The single and multiple scattering contributions arising from the target, the seabed, and their interactions have been derived under the Kirchhoff approximation extensions solution. And the occlusion effect is considered in the calculation, which is reflected in the sonar image as the target shadow. This study demonstrates that images generated based on echo can provide more complete target feature information and can be used as an alternative method to obtain sonar image quickly and accurately.

8:15

5aUW2. Evaluating underwater communication: A atatistical analysis of shallow water acoustic channels. Hammad Hussain (Electron. System, Norwegian Univ. of Sci. and Technol., Trondheim 7491, Norway, hammad.hussain@ntnu.no) and Syed Sajjad H. Zaidi (PNEC, National Univ. of Sci. and Technol., Islamabad, Pakistan)

Precise channel information is crucial for effective communication systems, particularly in underwater environments. This paper focuses on simulating acoustic channels in underwater communication, using an existing statistical method instead of introducing a new one. The traditional bellhop method cannot account for channel variations, whereas the statistical-based channel simulator we utilize can handle both small-scale and large-scale fading phenomena. Large-scale fading includes uncertainties such as location, time-varying environmental statistics, and variations in received power. Small-scale fading has Doppler shifts due to motion (e.g., vehicular or wave motion) and scattering, rapidly altering channel statistics. To assess the effectiveness of the statistical-based simulator, we used the five-element underwater acoustic dataset collected from an experiment on November 6, 2009. Our findings indicate that this simulator, which leverages the existing statistical method, outperforms the bellhop method in accurately modelling the acoustic channel. This approach shows potential for improving the design and deployment of underwater communication systems.

8:30

5aUW3. Extraction of comprehensive feature for active target detection using two hand-crafted acoustic features. YoungSang Hwang (Dept. of Ocean Syst. Eng., Sejong Univ., 209, Neungdong-ro, Gwangjin-gu, Seoul 05006, Republic of Korea, ldzezl@naver.com), Geunhwan Kim, Wooyoung Hong, and Youngmin Choo (Dept. of Ocean Systems Eng., Sejong Univ., Seoul, Republic of Korea)

We propose a model for creating a new feature suitable for classifying targets and non-targets using a small active sonar data. The proposed model extracts a comprehensive feature from two different hand-crafted features using deep-learning (DL). To extract comprehensive feature, we contrived a complementary learning module (CLM), which consists of two stages. In the first stage, a transformer encoder was adopted to reinforce the input features and then complementarily learned through attention mechanism in the following stage. The output features from the CLM are passed through a shallow CNN to classify targets and non-targets. In addition, an uncertainty quantification method is proposed using two CLMs to quickly add new samples acquired in the real ocean dataset. Using the decisions from two CLMs, we calculate the variance and determine which samples from the entire dataset are worthy of prioritized addition. Two real-ocean datasets were used to examine the model generalization and compare the classification results with conventional DL models. The generalization performance of the proposed model was comparable or superior to other DL models and uncertainty quantificaion method improved the generalization performance. [Work supported by Korea Research Institute for defense Technology planning and advancement(KRIT)—Grant funded by the Korea government (DAPA—Defense Acquisition Program Administration) (No. KRIT-CT-23-026, Integrated Underwater Surveillance Research Center for Adapting Future Technologies, 2024)]

8:45

5aUW4. Acoustic channel estimation via adaptive regularized NLMF algorithm for underwater communication. Hammad Hussain (Electron. Syst., Norwegian Univ. of Sci. and Technol., Trondheim 7491, Norway, hammad.hussain@ntnu.no) and Syed Sajjad H. Zaidi (PNEC, National Univ. of Sci. and Technol., Islamabad, Pakistan)

Estimating shallow underwater acoustic channels is challenging due to the time-varying and sparse nature of the multi-path profile. This paper proposes a novel training-based channel estimation technique that leverages the efficient time-varying regularized normalized least mean fourth (R-NLMF) algorithm. We demonstrate that this algorithm effectively caters to the sparse structure of the channel and significantly improves the performance of the channel estimator. The proposed technique is validated using experimental data from shallow underwater acoustic experiments conducted on Nov 6, 2009, named the "Five-Element Underwater Acoustic dataset." Our results show that the adaptive learning rate and regularized NLMF algorithm provide accurate channel estimation and effectively mitigate the effects of large-scale and small-scale fading.

9:00

5aUW5. Waveguide invariant navigation of an AUV with a towed line array. Junsu Jang (Scripps Inst. of Oceanogr., UC San Diego, 9679 Caminito Del Feliz, San Diego, CA 92121, jujang@ucsd.edu) and FLORIAN MEYER (Univ. of California San Diego, La Jolla, CA)

Passive acoustics navigation of an autonomous underwater vehicle (AUV) can reduce localization errors while being relatively low-cost. A promising approach for navigation in shallow water is to exploit the waveguide invariant, which makes it possible to estimate the range between an acoustic source with a known position and an acoustic receiver on the AUV. This range estimate can then be fused in a sequential Bayes framework to improve the navigation information of the AUV. While previously proposed methods consider a range-independent environment with a single acoustic source and a single receiver, we investigate a more realistic scenario with multiple sources in a potentially range-dependent environment. In particular, a shallow water environment with a constant slope and sound speed is considered. To resolve multiple sources, it is assumed that the AUV tows an acoustic line array. Simulated acoustic data provided by a normal mode program are used to study the feasibility of the AUV navigation in shallow water by exploiting the waveguide invariant in this setting.

9:15

5aUW6. Performance analysis of passive acoustic ranging using a single hydrophone in shallow waters. Jeong Bin Jang (Korea Univ. of Sci. and Technol., 32, Yuseong-daero 1312 beon-gil, Yuseong-gu, Daejeon 34103, Republic of Korea, jangjb@kriso.re.kr) and Sung-Hoon Byun (Korea Res. Inst. of Ships and Ocean Eng., Daejeon, Republic of Korea)

Passive source range estimation is one of the important research fields on the sonar system. Recent research has carried out range estimation using single hydrophone. This study presents the result of analyzing the impact of environmental conditions on range estimation performance when estimating the range of a moving source with a single hydrophone. Range estimation of a moving source using a single hydrophone estimates source velocity from the cross-correlated fields of the signal measured at time intervals. This method of estimating the source range from estimated source velocity is used. Therefore, it is important to accurately estimate the source velocity for ranging, and the accuracy of estimating the source velocity is affected by the propagating mode characteristics in waveguide. This study investigates the changes in velocity estimation performance of a moving source according to various underwater environment and mode formation characteristics through simulation analysis and discusses ways to reduce velocity estimation error. [Work supported by KRIT (Contract No. 22-305-B00-001).]

9:30–9:45 Break

9:45

5aUW7. Generalization performance analysis of anomaly detection-based active sonar classifier using anomaly score landscape. Geunhwan Kim (Dept. of Ocean Syst. Eng., Sejong Univ. 209, Neungdong-ro, Gwangjin-gu, Seoul 05006, Republic of Korea, kimgw200@sejong.ac.kr), YoungSang Hwang, Keunhwa Lee, and Youngmin Choo (Dept. of Defense Syst. Eng., Sejong University, Seoul, Republic of Korea)

This study explores generalization performance of anomaly detection-based active sonar classifier using the deep neural network (NN). Despite its superior performance, NN remains challenging to interpret its decision due to black-box nature, which, consequently, makes it hard to trust the results. To address this, we employ the loss landscape method used to analyze the generalization performance of supervised learning-based NN. Owing to the inapplicability of the conventional loss landscapes to anomaly detection-based classifier, we propose a novel anomaly score landscape for the anomaly detection-based classifier. Experimental results using active sonar datasets reveal that anomaly detection-based classifier maintains consistent landscape structures between train and test datasets, compared to supervised learning counterparts. This study provides insights into the interpretability and generalization capabilities of supervised- and anomaly detection-based active sonar classifiers.

10:00

5aUW8. Estimation of sound source direction by holographic method in the presence of shallow water internal waves. Sergey A. Pereselkov (Math. Phys. and Information Technol., Voronezh State Univ., Russia, Voronezh, Universitetskay pl, 1, Voronezh 394018, Russian Federation, pereselkov@yandex.ru), Venedikt Kuz'kin (Sci. Ctr. for Wave Res., General Phys. Inst., Moscow, Russian Federation), Nikolai Ladykin, and Alexey Pereselkov (Math. Phys. and Information Technol., Voronezh State Univ., Voronezh, Russian Federation)

A holographic signal processing method for estimation of sound source direction in the presence of intensive internal waves (IIWs) is considered in the paper. It is assumed that IIW propagate along acoustic track between source and receiver. The direction estimation is based on holographic processing [Pereselkov S. and Kuz'kin V., JASA 151(2), 666–676] of vector receiver (VR) channels (x-th and y-th). The two source interferograms (x-th and y-th) is generated in frequency-time domain. The results of the two-dimensional Fourier transformation (2D-FT) of interferograms are source holograms (x-th and y-th). The direction estimation is calculated by ratio of the absolute values of the holograms angle distributions. Numerical experiment results of source direction estimation in the presence of IIW for low-frequency band (150-250 Hz) are considered. It is shown that holograms (x-th and y-th) consist of two disjoint regions corresponding to nonperturbed field and field perturbation by IIW. This structure of the holograms allows to separate the unperturbed field component in absence of IIW (x-th and y-th). So, IIW influence on direction estimation can be reduced significantly. The error in direction estimation by holographic method caused by IIW is analyzed in the paper. [Work supported by grant from the Russian Science Foundation 23-61-10024.]

10:15

5aUW9. Holographic signal processing for sound source depth estimation by single receiver in shallow water. Sergey A. Pereselkov (Math. Phys. and Information Technol., Voronezh State Univ., Russia, Voronezh, Universitetskay pl, 1, Voronezh 394018, Russian Federation, pereselkov@yandex.ru), Venedikt Kuz'kin (Sci. Ctr. for Wave Res., General Phys. Inst., Moscow, Russian Federation), Pavel Rybyanets, and Alexey Pereselkov (Math. Phys. and Information Technol., Voronezh State Univ., Voronezh, Russian Federation)

The holographic method for estimating of the low-frequency broadband source depth by single receiver in a shallow water is developed in the paper. By using broadband signals from receiver, the source interferogram (sound intensity distributions) is generated in frequency-time domain. Within framework of holographic signal processing (Pereselkov S. and Kuz'kin V., JASA 151(2), 666–676) is analyzed by two-dimensional Fourier transform (2D-FT). The result of the 2D-FT is called the Fourier hologram (source hologram). The structure of source hologram allows to separate noise and spectral spots relating to modes interference. By using inverse 2D-FT, the cleared source interferogram with information of the mode amplitudes is reconstructed. The ratio of neighboring modes amplitudes allows to estimate the source depth. Results of a numerical experiment of source depth estimation by holographic method in the low-frequency band (100–300 Hz) are analyzed. The stability of the holographic method to errors in measuring of mode amplitudes and variations of the waveguide parameters are considered. It is shown that error estimation of the source depth tends to the established value with increasing noise level. The qualitative and quantitative explanations of the simulation results are presented in the paper. [Work supported by grant from the Russian Science Foundation 23-61-10024.]

10:30

5aUW10. Passive acoustic localization using dictionary learning. 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)

Ray-trace simulations are used to demonstrate the use of Dictionary Learning to improve passive localization of a source monitored with a vertical linear acoustic array. The learned dictionary, generated using historical

sound velocity profile (SVP) data from a region of interest, is an over complete, sparse representation of the SVP training set. By minimizing an objective function that measures the differences between multipath reception intervals detected at a receiver for propagation through candidate and baseline SVP profiles, we show that an optimal SVP match to the "unknown" baseline can be reconstructed by an efficient dictionary search. Ray traces back-propagated through the reconstructed SVP according to beamformed receive angles computed with the baseline SVP are then found to well estimate the source position as the centroid of a cluster of multipath signal intersections. The accuracy for small unit-sparsity dictionaries of size up to 50 was evaluated on a randomly sampled, 30 SVP testing set obtained from an area close to that of 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 at source depths from 50 to 450 m and at source ranges from 2000 to 4500 m. Compared to using the average sound speed profile to estimate source position, using the learned dictionary produced a general performance increase in position accuracy.

10:45

5aUW11. Acoustic diversity, bearing estimation, and localization of sound sources in shallow water and deep ocean off the U.S. Northeast coast using a coherent hydrophone array. Sai Geetha Seri (Elec. Eng., Northeastern Univ., 360 Huntington Ave. Boston, MA 02115, seri.s@northeastern.edu), Max Radermacher (Northeastern Univ., Boston, MA), Hamed Mohebbi-Kalkhoran (Mech. Eng., Massachusetts Inst. of Technol., Cambridge, MA), and Purnima R. Makris (Elec. and Comput. Eng., Northeastern Univ., Boston, MA)

An in-house developed 160-element large-aperture hydrophone array from Northeastern University was utilized for ocean monitoring in both the shallow waters of the Great South Channel (GSC) and the deep-water area off the continental slope south of Rhode Island in September 2021. Utilizing passive ocean acoustic waveguide remote sensing (POAWRS) technology, the system enabled real-time, broad-area surveillance of the oceanic environment. A novel multistage clustering was introduced for annotating the vast volume of underwater acoustic data, drawing inspiration from the human cognitive process of categorizing sounds. Initially, sounds are separated into broad groups, akin to a first human listening pass. Subsequent stages of clustering refine these categories, utilizing different sets of features to achieve a granularity of annotation that closely mirrors expert human annotation processes. It was found that the acoustic landscape of the GSC was primarily composed of diverse marine mammal sounds, such as fin whale 20 Hz pulses, humpback whale songs, minky whale buzz sequences, sperm whale and dolphin echolocation clicks, and unidentified whale calls. Conversely, the deep-water area off the continental slope featured predominantly fish sounds, chorus-like sounds, and dolphin vocalizations. Detailed analyses of the most prevalent vocalizations are presented, including their bearing-time trajectories and localizations.

11:00

5aUW12. Evaluating the directivity of compact underwater acoustic recording devices. Robert T. Taylor (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, rtaylor119@utexas.edu), Megan Ballard, Colby W. Cushing (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Preston S. Wilson (Walker Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Kevin M. Lee, Andrew R. McNeese, Luis Acuna (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Julien Bonnel (Woods Hole Oceanographic Inst., Woods Hole, MA)

Commercially available underwater acoustic recorders have become commonplace tools in ocean-acoustics research, due to their ease of deployment, compact size, and relatively low cost. The size of these systems typically results in a configuration with the hydrophone in close proximity to the electronics housing, flotation devices, and other equipment that can degrade the generally assumed omnidirectional response of the hydrophone. The mid-frequency acoustic regime (0.5–10 kHz) is particularly affected due to the similarity between the length-scales of these structures and the corresponding acoustic wavelengths. Calibration measurements made at an open-water test facility with a calibrated source/receiver pair characterized the frequency-dependent receiver directivity for three underwater recording devices with different housings and hydrophones: the PVC air-filled Loggerhead Snap, PVC oil-filled SoundTrap ST300, and titanium air-filled SoundTrap ST600. Furthermore, directivity measurements of a TOSSIT mooring [Zitterbart *et al.*, HardwareX (2022)] were taken with the SoundTrap ST300 and SoundTrap ST600. Results suggest the frequency-dependent acoustic directivity of the recorders should not be neglected. In particular, the Loggerhead Snap had variations in receive level of over 20 dB as a function of receiver orientation angle and frequency, introducing a bias that obscures spectral levels of the *in situ* environment.

11:15

5aUW13. Optimizing parametric acoustic arrays for high-fidelity low-frequency signals. Ernst Uzhansky (Mech. Eng., Technion — Israel Inst. of Technol., Izhaq Greenboim str, Apt. 12 (Koren Family), Haifa 3498793, Israel, ernstuzhansky@gmail.com), Maya Friedlender, and Izhak Bucher (Mech. Eng., Technion — Israel Inst. of Technol., Haifa, Israel)

Parametric acoustic array (PAA) can leverage the nonlinear behavior of sound travelling through the nonlinear medium (e.g., air, water, human tissues) to generate low-frequency broadband high-directivity sound due to the self-demodulation of finite-amplitude modulated ultrasonic waves. However, while propagating, self-demodulated signals (hereon message) interact one with another, leading to message deformation and increase of the total harmonic distortion (THD). This makes it problematic to use powerful PAA to control complex low-frequency signals. This work explores a method for reducing THD by optimizing the input signal using various digital signal processing algorithms. The sustainability of the required spectral components with distance and the spatial boundaries of the operation of the proposed optimization algorithm are also investigated experimentally with a 520-channel hexagonal PAA and numerically. The simulations were done via k-wave acoustic toolbox considering nonlinearities and power law absorption.

Session 5pPA

Physical Acoustics: General Topics in Physical Acoustics

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Jacob C. Elliott, Cochair
The Penn. State Univer., Research West, State College, PA 16801

Contributed Papers

1:00

5pPA1. Dynamics of cylindrical crevice microbubbles. Eric Rokni (Phys., Rollins College, 511 W. Kenilworth Cir., Mequon, WI 53092, eric.rokni@gmail.com) and Julianna C. Simon (Graduate Program in Acoust., Penn State Univ., University Park, PA)

Microbubbles are theorized to form in crevices on mineralizations causing twinkling, a rapid color shift observed when imaging hard mineralizations with Doppler ultrasound. While the classic Rayleigh–Plesset equation and subsequent modifications describe bubble dynamics in a free field, they do not account for crevice boundaries. In this study, cylindrical boundary conditions were incorporated into the derivation of the Rayleigh–Plesset equation and validated experimentally. Cylindrical crevices with diameters ranging from 0.08–1.2 μm and depths of 1 μm were etched into a silicon wafer. Bubbles that formed in these crevices were photographed at high-speed through an inverted microscope while being driven with ultrasound at 0.75, 2.5, and 5 MHz. Experimentally, for all tested crevice sizes, the bubbles did not visibly grow. In contrast, computational results using standard water surface tension (72.5 mN/m) predicted the bubbles should grow $\sim$6 mm. However, when applying a higher surface tension that was measured on the silicon-water interface ($\sim$3000 mN/m), the bubbles were only predicted to grow $\sim$1 nm, supporting the experimental findings. These results offer insight into the mechanism causing twinkling and provides avenues for future investigation into the effect of crevice size and shape on twinkling. [Work supported by NSF CAREER 1943937 and GRFP DGE1255832.]

1:15

5pPA2. Cryoprotectant agent characterization via acoustical and optical analyses. Alicia Casacchia (Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, 204 E. Dean Keeton St., Austin, TX 78712-1591, acasacchia@utexas.edu), Matthew J. Uden (Dept. of Psychol., The Univ. of Texas at Austin, Austin, TX), Tanya Hutter, Preston S. Wilson, and Mark F. Hamilton (Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX)

In the field of cryopreservation, there exists a class of substances known as cryoprotectant agents (CPAs), which display an ability to vitrify. These CPAs are used to prevent cellular damage during the preservation of human tissues. In a vitrified state, cells exist in a glass-like state, meaning there is no formation of mechanically damaging ice crystals during cryogenic freezing and thawing. Although a CPA's ability to vitrify increases with concentration, its toxicity limits viable levels of use. This work seeks to find a correlation between acoustic properties of various CPAs and their concentrations via analyses of cavitation noise and cavitation jets, to accompany future studies on CPA toxicity. Four common CPAs, namely, dimethyl sulfoxide, ethylene glycol, propylene glycol, and formamide were investigated at various concentrations in aqueous mixtures and compared to pure water. The acoustic spectra of each CPA show an observable dependence on concentration levels and provide a potential way to probe the hydrogen bonding dynamics within the mixtures. [This is a SAWIAGOS project.]

1:30

5pPA3. Ultrasonic measurement of milk heat coagulation time (HCT). Agustin Harte (Dept. of Eng. Sci., Penn State Univ., 615 Meeks Ln., Port Matilda, PA 16870, afh5830@psu.edu), Lauren Katch, Federico Harte (The Penn State Univ., Port Matilda, PA), and Andrea P. Arguelles (Eng. Sci. and Mech., Penn State Univ., State College, PA)

The dairy industry relies on heat coagulation tests (HCTs) to determine high-quality milk heat stability for typical thermal processes. For the HCT, an oil bath heats 1–2 ml of milk to 140 °C. The test lasts 20–30 min and finishes with visual confirmation of milk coagulation. This time for coagulation is recorded as the HCT time. While standard in the industry, this test is time-consuming and prone to operator-dependent bias. Thus, an automated alternative for measuring HCT time is desirable. This work proposed the use of ultrasonic monitoring and employed 10 MHz contact and immersion ultrasonic transducers in combination with conventional pH and rheological sensors. Room-temperature skim milk was coagulated in standard containers with a Glucono-Delta-Lactone acid concentration of 3% while simultaneously being monitored by the ultrasound, rheological, and pH sensors, at testing rates of 1 s, ca. 6 s, and 2 min, respectively. The resulting ultrasonic wave speed measurements revealed an increasing trend throughout the coagulation process. Additionally, an inflection point within the increasing wave speed corresponded to rheological and pH parameters indicating coagulation. This work demonstrates the potential for ultrasonics to be used as sensors for coagulation within the dairy industry.

1:45

5pPA4. Measurements of nonlinear electric–acoustic interactions in liquids. Robert Lirette (Commun. Technol. Lab., National Inst. of Standards and Technol., 325 Broadway, MS67201, Boulder, CO 80305, robert.lirette@nist.gov), Tomasz Karpisz (Commun. Technol. Lab., National Inst. of Standards and Technol., Boulder, CO), Małgorzata Musiał, Jason A. Widegren (Mater. Measurement Lab., National Inst. of Standards and Technol., Boulder, CO), Aaron Hagerstrom, Nathan Orloff, and Angela C. Stelson (Commun. Technol. Lab., Natl. Inst. of Standards and Technol., Boulder, CO)

Electric and acoustic waves can interact nonlinearly inside a fluid medium. These interactions can create a heterodyned signal consisting of both sum and difference frequency products of the electric and acoustic frequencies. Here, we present direct measurements of nonlinear mixing products produced by interacting electric and acoustic waves inside of various fluids. The experiment consisted of a fluid channel that was in direct contact with an electrically conductive co-planar wave guide (CPW). An acoustic signal was applied to the channel via a transducer mounted above it, and simultaneously an electric signal was applied along the CPW from a vector network analyzer (VNA). The resulting signal was then measured with VNA set to frequency offset mode to capture the sum electric-acoustic mixing product. Results are presented for various organic solvents and salt solutions, each showing a different characteristic nonlinear response. This metrology allows for direct probing of non-linear effects in electrici–acoustic systems and

could lead to the development of new spectroscopic methods for characterizing chemicals and solutions.

2:00

5pPA5. Custom lens design for ultrasonic inspection of immersed anisotropic parts. Lauren Katch (Eng. Sci. and Mech., Penn State Univ., 212 Earth and Eng. Sci. Bldg., State College, PA 16802, Luk50@psu.edu) and Andrea P. Arguelles (Eng. Sci. and Mech., Penn State Univ., State College, PA)

Inspection of anisotropic parts using immersion ultrasonic waves poses resolution challenges caused by directionally dependent wave properties. Currently, focused lenses with spherical or cylindrical geometries are the standard method for creating convergent, small cross-section, and high amplitude beams within solids. These lenses form a conical beam that uniformly refracts at the fluid-solid interface of the immersed part when the solid is isotropic, producing the desired focusing behavior. However, for anisotropic solids, refraction becomes directionally dependent, which distorts the beam and causes non-uniform focal profiles and, in some cases, multiple foci. In this work, the forward wave propagation for an immersed anisotropic solid is modeled and exploited to develop custom lens geometries. These custom lenses utilize inverted slowness profiles to produce conical beams within the anisotropic solid. By incorporating these lenses, the focused beam behavior is improved and becomes more circular, higher in amplitude, and smaller in the cross-sectional area. These lenses have the potential to improve inspection resolution in anisotropic solids, an effort beneficial to the additive, aerospace, and electronics industries.

2:15

5pPA6. Loss factors of viscoelastic solid materials and the Poisson's ratio. Tamás Pritz (Budapest Univ. of Technol. and Economics, Műegyetem rkp. 3-9, Budapest 1111, Hungary, tampri@eik.bme.hu)

The dynamic properties of viscoelastic materials can effectively be characterized in the frequency domain by the complex moduli of elasticity and the complex Poisson's ratio. The ratio of imaginary to real part in these complex frequency functions is referred to as loss factor. In this paper, the relations between the loss factors of various complex moduli (shear, bulk, Young's, longitudinal) including the complex Poisson's ratio are investigated for homogeneous, isotropic, linear viscoelastic solids possessing positive Poisson's ratio. The novelty of the work is that the loss factor relations are expressed and investigated as a function of Poisson's ratio, ν, the magnitude of which is known for real solids, namely: $0 \leq \nu < 0.5$. It is shown that if $\nu = 0$ (e.g., cork), then all modulus loss factors are identical with the shear one, while the Poisson's loss factor is zero. In the general case when $0 < \nu < 0.5$ (e.g., hard plastics), the shear loss factor is the largest, and the others are smaller in the following order: Young's, longitudinal, bulk, and Poisson's loss factor. Finally, if $\nu \rightarrow 0.5$ (e.g., rubber), then the Young's loss factor approaches the shear one, while the others are usually much smaller than that.

2:30–2:50 Break

2:50

5pPA7. Acoustic signatures of helium abundance in hydrogen for planetary science. Rishabh Chaudhary (Mech. Eng., Tufts Univ., 200 College Ave., Medford, MA 02155, rishabh.chaudhary@tufts.edu), Robert D. White, Zarina Kosherbayeva (Mech. Eng., Tufts Univ., Medford, MA), Don Banfield, Anthony Colaprete (NASA Ames Res. Ctr., Mountain View, CA), Ian Neeson (VN Instruments, Elizabethtown, ON, Canada), Andrew Powell, and Andi Petculescu (Dept. of Phys., Univ. of Louisiana at Lafayette, Lafayette, LA)

The aim of this work is to model and measure the acoustic signatures of varying mixtures of helium, ortho-hydrogen, and para-hydrogen. In particular, we hope to demonstrate a sensor for helium abundance and hydrogen ortho:para ratio for planetary science missions to Saturn or Uranus. Gas composition as a function of depth is important for understanding atmospheric dynamics and planetary formation of the gas giants but is not well known. In the presence of molecular relaxation, sound speed and absorption in polyatomic gas mixtures become dispersive due to frequency-dependent heat capacities. This may potentially allow the identification of gas composition through relaxational acoustic spectroscopy. We compare theoretical predictions to measurements done in a pressure vessel up to a maximum pressure of 20 bar of relevance to gas giant atmospheres. Experiments for different gas mixtures at various pressures and temperatures are conducted using wide-bandwidth ultrasound between 20 kHz and 10 MHz. A technique is proposed to determine the ratio of hydrogen to helium using speed of sound and attenuation spectra. This approach may be further utilized to determine the ortho:para ratio. Challenges include providing sufficient bandwidth and adjusting for temperature, diffraction, and reflections. [Work supported from NASA Center Innovation Fund.]

3:05

5pPA8. Multi-sensor acoustic intensity analysis of the NG-19 Antares 230 launch. Carson F. Cunningham (Phys., Brigham Young Univ., Brigham Young University, Provo, UT 84602, carsonfcunningham@gmail.com), Micah Shepherd (Brigham Young Univ., Provo, UT), and Kent L. Gee (Dept. of Phys. and Astron., Brigham Young Univ., Provo, UT)

In August 2023, the Antares 230 successfully launched for the NG-19 resupply mission to the International Space Station. Acoustical data were collected from various locations surrounding the launch pad from 60-200 m away from the vehicle. Each measurement location was configured such that acoustic intensity could be determined for a wide range of frequencies. Intensity calculations are made using the phase and amplitude gradient estimate (PAGE) method for information past the spatial Nyquist frequency of the station configuration. For the analyses presented here, the PAGE method utilized the generic MATLAB phase unwrapping function. This paper will show intensity vectors as a function of frequency and time to estimate the approximate location of the sound source. A least-squares optimization routine is used for the intersection point of each stations' intensity vector.

3:20

5pPA9. Nonlinear waveform steepening in time reversal focusing of airborne, one-dimensional sound waves, and the absence of Mach stems. Michael M. Hogg (Phys. & Astron., Brigham Young Univ., N283 ESC, Provo, UT 84602, mikerhogg@gmail.com), Brian D. Patchett (Phys., Utah Valley Univ., Orem, UT), and Brian E. Anderson (Phys. & Astron., Brigham Young Univ., Provo, UT)

Time reversal (TR) is a process that can be used to generate high amplitude focusing of sound. Previous research has shown that TR in reverberant environments can generate peak levels that exceed 200 dB with airborne, audible sound [Patchett and Anderson, J. Acoust. Soc. Am. 151(6), (2022)], and 134 dB with airborne ultrasound [Wallace and Anderson, J. Acoust. Soc. Am. 150(2), (2021)]. Particularly, in the focusing of audible sound, these high amplitude focused waveforms exhibit multiple nonlinear properties including waveform steepening and a nonlinear increase in compressions due to the generation of free space Mach stems. This study attempts to remove the Mach stems by generating the focus in a 1D system, since Mach stems require higher dimensional spaces to form. The system of pipes used ensures a planar 1D reverberant environment. Results show that waveform steepening remains as expected but that the nonlinear increase in compression amplitudes disappears without the ability for Mach stems to form. This provides further confirmation that Mach stems are the cause of the nonlinear increase in compression amplitudes observed with high amplitude TR.

3:35

5pPA10. Characterization and impact of dissolved gases in a high-frequency flow reactor. Yihao Huang (Eng. Sci., Univ. of Oxford, Dept. of Eng. Sci., University of Oxford, Parks Rd., Oxford OX13PJ, United Kingdom, yihao.huang@exeter.ox.ac.uk), Pankaj Sinhmar (Eng. Sci., Univ. of Oxford, Oxford, United Kingdom), Tristan Nerson (LAUM, Le Mans Université, Le Mans, France), Cherie Wong, Lillian N. Usadi (Eng. Sci., Univ. of Oxford, Oxford, United Kingdom), Jason L. Raymond (Fralin Biomedical Res. Inst. at VTC, Oxford, United Kingdom), and James Kwan (Eng. Sci., Univ. of Oxford, Oxford, United Kingdom)

Addressing the growing demands for sustainable green technology in chemical processes, we aim to develop a novel high-frequency ultrasonic flow reactor that utilizes converting ultrasound waves, resulting in intense

localized acoustic pressure fields. Our hypothesis that our design of converging waves will establish a uniform spatial distribution of cavitation activity with high energy density, thereby augmenting the rate of radical formation and significantly enhancing overall reactor performance. To validate the hypothesis, KI and terephthalic acid dosimetry were performed. A comprehensive calorimetric study was employed for acoustic power, yielding valuable insight into the energy dynamics of the system. Furthermore, our research systematically assesses the impact of dissolved gases and varying flow rates on reactor performance. Notably, we observe that dissolved gases in the reaction medium exert a substantial influence on reactor efficiency. Additionally, an increased flow rate beyond the optimum decelerates hydroxyl radical generation due to a lower residence time. The detailed characterization, facilitated by dosimetry techniques, unveils the potential of the sonochemical reactor for industrial applications.

3:50

5pPA11. Sono-catalytic syngas production by low-temperature biomass gasification. Cherie Wong (Eng. Sci., Univ. of Oxford, Begbroke Directorate, Begbroke Hill, Woodstock Rd., Oxford OX5 1PF, United Kingdom, cherie.wong@eng.ox.ac.uk) and James Kwan (Eng. Sci., Univ. of Oxford, Oxford, United Kingdom)

Syngas holds a crucial role in diverse industrial applications, including chemical synthesis and power generation. It has been traditionally generated through high-temperature processes, such as steam reforming or biomass gasification, demanding temperatures of up to 1200 °C, which results in substantial energy consumption. Consequently, there is a growing interest in exploring alternative, low-temperature biomass gasification processes. This paper introduces an innovative method for syngas production through sonocatalytic biomass conversion at significantly lower temperatures. The approach employs a sonochemical reactor that concentrates acoustic waves to activate inertial cavitation. Furfural, a biomass-derived platform chemical, was chosen as the primary chemical for the study, showcasing the potential of sonochemical syngas production. The reaction converted furfural into various gas products, particularly a mixture of hydrogen and carbon monoxide, with the chemical reaction hypothesized to be triggered by pyrolysis during bubble collapse hot spots. The incorporation of heterogeneous metal foams into the system further enhanced furfural conversion rates. This improvement was attributed to the preferential nucleation of cavitation bubbles on the porous metal foam surface, bringing the hot spots into proximity with furfural molecules. This application of sonochemistry in furfural pyrolysis at low temperatures holds promise for a more economical approach to biomass valorization.

4:05

5pPA12. Torsional pendulum driven by sound-matter orbital angular momentum transfer involving acoustic vortices. Elena Annenkova (Laboratoire Ondes et Matière d'Aquitaine, Univ. of Bordeaux, 351 Cours de la Libération, Talence 33405, France, a-a-annenkova@yandex.ru), Benjamin Sanchez-Padilla, and Etienne Brasselet (Laboratoire Ondes et Matière d'Aquitaine, Univ. of Bordeaux, Talence, France)

The transfer of the orbital angular momentum from sound to matter using acoustic vortex beams has been experimentally revealed only recently. The physics of such a process may involve two distinct mechanisms that can add one to another: a dissipative one driven by sound absorbing targets and a nondissipative one driven by angular momentum conversion as a result of a scattering process. In both cases, the mechanical consequence is the appearance of an acoustic radiation torque exerted on the irradiated body. Here, following recent developments of torsional pendulum driven by sound-matter orbital angular momentum transfer, we report on our recent advances, which are two-fold. First, we report on drastic improvement by more than two orders of magnitude of the quality factor of the mechanical resonance, reaching values >1000. Second, we extend the detection and measurement of acoustic radiation torque from a scalar framework to situations where the vectorial nature of the angular momentum matters. These results contribute to the development of sensitive wave-matter devices for rotational metrology applications encompassing material and wave aspects. [Work received funding from the European Union's Horizon 2020 research and innovation programme under the Marie Sklodowska-Curie grant agreement No. 101027737.]

4:20

5pPA13. Effects of increasing orbital number on the field transformation in focused vortex beams. Chirag A. Gokani (Appl. Res. Labs. and Walker Dept. of Mech. Eng., Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, cgokani@arlut.utexas.edu), Michael R. Haberman, and Mark F. Hamilton (Appl. Res. Labs. and Walker Dept. of Mech. Eng., Univ. of Texas at Austin, Austin, TX)

Acoustic vortex beams possess helical wavefronts characterized by an orbital number n. In the presence of focusing, the toroidal vortex ring for $n = 1$ moves out of the focal plane $z = d$ and toward the source with increasing n, accompanied by a departure from its toroidal shape. In this talk, ray theory is developed to explain this field transformation. An expression is derived for the radius and cross-sectional area of the annular channel formed by the family of rays emanating from a thin circular ring centered at the origin in the source plane $z = 0$. This expression leads to explicit results for the field amplitude and caustic coordinates due to a focused vortex source with an arbitrary axisymmetric amplitude distribution. Comparison with field calculations in which diffraction is included demonstrate that the caustics describe the redistribution of the global maximum and its shift toward the source plane with increasing n. For moderate focusing gains (of order 10 or 20), a toroidal vortex ring for $n = 1$ transforms with increasing n into a spheroidal surface in the prefocal region $0 < z < d$ having volume $n\lambda d^2/6$, where λ is the wavelength, inside of which exists a shadow zone. [C.A.G. was supported by the ARL:UT McKinney Fellowship in Acoustics.]

Session 5pUW

Underwater Acoustics: Underwater Sources and Receivers

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Contributed Papers

1:00

5pUW1. Structure of interferogram and hologram of non-moving source in presence of shallow water internal waves. Mohsen Badiey (Elec. and Comput. Eng., Univ. of Delaware, 139 The Green, Rm. 140, Evans Hall Rm. 140, Newark, DE 19716, badiey@udel.edu), Sergey A. Pereselkov (Math. Phys. and Inform. Technol., Voronezh State Univ., Voronezh, Russian Federation), and Venedikt Kuz'kin (Sci. Ctr. for Wave Res., General Phys. Inst., Moscow, Russian Federation)

The results of holographic signal processing of the oceanographic experiment SWARM-95 on the coast of New Jersey are presented in the paper. The used acoustic signals were obtained on stationary acoustic track (non-moving airgun source) in the presence of shallow water intensive internal waves (IIW). The IIW led to significant 3D acoustic effects [Badiey et al., JASA 117(2), 613–625]. Within framework of holographic signal processing [Pereselkov and Kuz'kin, JASA 151(2), 666–676] the source interferogram (sound intensity distributions in frequency-time coordinates) is analyzed by two-dimensional Fourier transform (2D-FT). The result of the 2D-FT is called the Fourier hologram (source hologram). In result of holographic signal processing, the two separated sets of spectral spots are obtained in hologram domain. The first set of spectral spots corresponds to sound field in nonperturbed waveguide (without of IIW). The second set of spectral spots in hologram corresponds to hydrodynamic field perturbation by IIW. This structure of the hologram allows the reconstruction of interferogram of nonperturbed field for waveguide in the absence of IIW. The error in the reconstruction of the nonperturbed interferogram is estimated. [Work supported by grant from the Russian Science Foundation 23-61-10024.]

1:15

5pUW2. Quiet uncrewed surface vessels assess fish avoidance to motorized survey ships with varying noise levels in the Great Lakes. Thomas M. Evans (DNRE, Cornell, Ithaca, NY), Lars G. Rudstam (Natural Resources and the Environment, Cornell Univ., Bridgeport, NY), Suresh A. Sethi (Earth and Environ. Sci., Brooklyn College, Brooklyn, NY), Daniel L. Yule (USGS, Ashland, WI), David M. Warner (USGS, Ann Arbor, MI), S. D. Hanson (USFWS, New Franken, WI), Benjamin Turschak (Michigan Dept. of Natural Resources, Charlevoix, MI), Mark R. Dufour, Steven A. Farha (USGS, Ann Arbor, MI), Andrew Barnard (Acoust., Appl. Sci. Bldg., Penn State, 201C, University Park, PA 16802, barnard@psu.edu), Steven Senczyszyn (Mech. Engineering-Eng. Mech., Michigan Technol. Univ., Houghton, MI), Susan E. Wells (USFWS, Ithaca, NY), Scott R. Koproski (USFWS, Alpena, MI), Timothy P. O'Brien (USGS, Ann Arbor, MI), Kevin N. McDonnell (USFWS, Alpena, MI), Hannah Blair (DNRE, Cornell, Southampton, NY), James M. Watkins (DNRE, Cornell, Ithaca, NY), Patricia M. Dieter (USGS, Ann Arbor, MI), James J. Roberts (USGS, Sandusky, OH), and Peter Esselman (USGS, Ann Arbor, MI)

Acoustic surveys are a foundational component of many fisheries monitoring programs because they allow assessment of spatially extensive stocks. They are widely used to evaluate prey fish throughout the Great Lakes by numerous coordinating vessels. Traditionally, these surveys have been conducted by crewed and motorized vessels, but fish avoidance of motorized platforms has been reported in multiple studies and may bias survey estimates. Quiet uncrewed platforms are becoming increasingly available and offer the opportunity to explore bias in traditional surveys. Several identical quiet uncrewed surface vehicles (USVs) operated by Saildrone were equipped with 120 kHz Simrad EK80 transducers and deployed in Lakes Erie, Huron, Michigan, and Superior in the summers of 2021, 2022, and 2023. The USVs were overtaken by numerous motorized vessels over 2 km transects using transducers with the same frequency. Fishes showed a limited response to approaching vessels, and acoustic measures were similar between the USVs and motorized vessels. Therefore, acoustic surveys in the Great Lakes appear unbiased and are widely comparable. Findings from this work will inform interpretation of acoustic data in the Great Lakes and provide the largest scale testing of fish avoidance during acoustic surveys to date.

1:30

5pUW3. Characterization, understanding, and mitigation of underwater noise radiated by ships in the St. Lawrence Estuary. Pierre Cauchy (Institut des Sci. de la mer (ISMER), Université du Québec à Rimouski (UQAR), 310 allée des Ursulines, Rimouski, QC G5L 3A1, Canada, pierre_cauchy@uqar.ca), Pierre Mercure-Boissonnault, Cécile Perrier de la Bathie, Faniry Rabetoandro (Institut des Sci. de la mer (ISMER), Université du Québec à Rimouski (UQAR), Rimouski, QC, Canada), Cédric Gervaise (Chorus, Grenoble, France), Sylvain Lafrance (Innovation Maritime, Rimouski, QC, Canada), and Guillaume St-Onge (Institut des Sci. de la mer (ISMER), Université du Québec à Rimouski (UQAR), Rimouski, QC, Canada)

Underwater noise generated by commercial traffic is the main source of anthropogenic noise pollution at low frequencies, increasingly present at a global scale and of critical interest in the St. Lawrence Estuary (eastern Canada), where a rich biodiversity meets the shipping corridor linking the Great Lakes to the Atlantic Ocean. The Marine Acoustic Research Station (MARS, www.projet-mars.ca/en) is an applied research project dedicated to characterizing, understanding, and mitigating underwater traffic noise, contributing to the global effort of improving cohabitation between human activities and marine life. A cutting edge marine acoustic observatory has been specifically designed to collect high-quality measurements of the underwater noise radiated by ships. It is deployed yearly since 2021, operating during the ice-free season. A database of over 2000 measurements representative of the underwater noise radiated by the fleet operating in the St Lawrence Estuary has been collected, with a demonstrated repeatability of less than 1.5 dB which confirms the ability of the observatory to effectively assess the efficiency of noise reduction measures. The MARS database supports the development of a noise prediction model and provides feedback to shipowners and relevant information regarding the St. Lawrence fleet to the government for future underwater vessel noise reduction targets.

1:45

5pUW4. Dynamic scheduling for the underwater acoustic localization of multiple moving targets. Youngchol Choi (KRISO, 32 1312 beon-gil, Yuseong-daero, Yuseong-gu, Daejeon 34103, Republic of Korea, ycchoi@kriso.re.kr) and Sea-Moon Kim (Ocean System Eng. Res. Div., Korea Res. Inst. of Ships and Ocean Eng., Daejeon, Republic of Korea)

We propose a novel method for the underwater acoustic localization of multiple moving transponders with a single ping. We make the transceiver receive the response signals from transponders without packet collisions by scheduling the packet transmission time of each transponder. To this end, the transceiver computes a doubly conservative round trip time (RTT) for each transponder based on the latest RTT, packet durations, maximum relative speed between the transceiver and the transponder, and the transceiver assigns a waiting time to each transponder such that the earliest packet arrival time of the next transponder should be greater than the latest packet reception completion time of the previous transponder. This dynamic scheduling method improves the update rate of the underwater acoustic localization system for multiple moving targets by ensuring the reception of packets of transponders in a packet train manner. In addition, even in scenarios where positioning and communication occur simultaneously, the proposed method guarantees robust and stable update rate as response signals from multiple moving transponders never collide at the transceiver, regardless of packet length variations due to changes in communication payload. [Work supported by KIMST funded by the Agency of Korea Coast Guard (KIMST-20210547).]

2:00

5pUW5. The underwater wind-generated noise characteristics and wind speed prediction of typhoon "Ma-on" by single vector hydrophone. Xiaoming Cui (Southern Marine Sci. and Eng. Guangdong Lab. (Guangzhou), 1119 Haibin Rd., Nansha District, Guangzhou 511458, China, nwpucxm@163.com), Huayong Yang (Southern Marine Sci. and Eng. Guangdong Lab. (Guangzhou), Guangzhou, China), Qing Hu (School of Ocean Eng. and Technol., Sun Yat-sen Univ., Zhuhai, China), Chao Li, and Siyuan Cang (Southern Marine Sci. and Eng. Guangdong Lab. (Guangzhou), Guangzhou, Guangdong, China)

Underwater ambient noise is greatly affected by sea surface wind speed, especially during typhoons when the wind speed over the sea surface rapidly increases within a short period of time. The relationship between noise and sea surface wind speed can be used to predict typhoon intensity. Based on the normal mode theory as well as the wind-generated sea noise propagation model and the wind-generated noise source level model, this paper uses the underwater noise vector sound field data measured by a single vector hydrophone during the passage of typhoon "Ma-on" to examine the relationship between underwater ambient noise intensity and the changes in wind speed and frequency during typhoon conditions. It also calculates the sea surface wind speed during a typhoon through inversion. The inversion result is essentially consistent with the forecasted typhoon wind speed provided by the meteorological observatory.

2:15

5pUW6. Derivations of transfer functions for estimating ship underwater radiated noise from onboard vibrations. Esen Cintosun (DRDC Atlantic, 9 Grove St., Dartmouth, NS B3A 3C5, Canada, Esen.Cintosun@ecn.forces.gc.ca)

The adverse impacts of underwater radiated noise (URN) from marine vessels on marine life are increasingly recognized. URN estimation and subsequent monitoring could be used to track URN and take measures to reduce it in sensitive environments (e.g., a marine life protected area). Two common analytical radiated noise power approximations and two empirical methods are assessed for URN estimation. The assessments were carried out by testing the methods on data from acoustic range measurements of an ORCA-class training vessel of the Royal Canadian Navy, named Patrol Craft Training (PCT) Moose and a DAMEN Combi Freighter 3850, named Heinz G. The analytical approximations are called equivalent radiated power and power from volume velocity; both are used to estimate URN directly from the onboard ship vibration measurements. The empirical methods are based on the correlations of onboard vibrations with measured URN, from which the derived transfer functions. The analytical and empirical transfer functions are compared. A statistical energy analysis model of PCT Moose is also used to estimate onboard vibrations from the ship's engine, generator, and propeller specifications for URN estimation at the design stage.

2:30

5pUW7. Bistatic spatial coherence for micronavigation of a volumetric synthetic aperture sonar. Kyle S. Dalton (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, ksd5377@psu.edu), Thomas E. Blanford (Univ. of New Hampshire, Durham, NH), and Daniel C. Brown (Graduate Program in Acoust., Penn State Univ., State College, PA)

Combining multiple passes of a volumetric synthetic aperture sonar (SAS) may open new possibilities for sonar imaging, if the passes can be precisely aligned. Previous work suggests that coherence-based micronavigation is a potential approach to achieve this alignment. However, existing coherence-based techniques require assumptions that break down in many volumetric imaging geometries. This breakdown leads to a loss of coherence that has not been widely studied, and consequently, has hindered the application of micronavigation to volumetric SAS systems. This work will explore the impact of sensing range and bistatic separation on spatial coherence in downward-looking, volumetric sonar applications where the far-field assumption no longer holds. As signals transmitted from downward-looking sonars may penetrate into the sediment floor, this analysis will consider both interface and volumetric scattering. First, using both simulated and experimental sonar data, we will quantify the loss in coherence that occurs in these near-field geometries. Then, methods to regain lost coherence will be examined in an effort to enable spatial coherence-based navigation for a volumetric SAS. Finally, the work's relevancy for ongoing efforts in acoustic navigation and repeat-pass imaging will be discussed.

2:45

5pUW8. Directional soundscapes in the Norwegian Seas observed with a coherent hydrophone array. Arpita Ghosh (Northeastern Univ., 360 Huntington Ave., Boston, MA 02115, ghosh.arp@northeastern.edu), Sai Geetha Seri (Northeastern Univ., Boston, MA), Hamed Mohebbi-Kalkhoran (Mech. Eng., Massachusetts Inst. of Technol., Cambridge, MA), Olav Rune Godø (Inst. of Marine Res., Bergen, Norway), Heidi Ahonen (Norwegian Polar Inst., Tromsø, Norway), Nicholas C. Makris (Mech. Eng., Massachusetts Inst. of Technol., Cambridge, MA), and Purnima R. Makris (Northeastern Univ., Boston, MA)

Directional soundscaping is an efficient approach for examining marine ecosystems since it allows the study of living organisms, their behavior, and temporo-spatial interactions with other natural and man-made objects underwater, useful for ecosystem monitoring during offshore energy activities, maritime surveillance, and defense. A large aperture coherent hydrophone array was employed for remote sensing in the Norwegian and Barents Seas in Spring 2014. Extensive analysis have been conducted in post-processing of recorded hydrophone array data for automatic detection, bearing estimation, and classification of signals by different sound producers, such as whale calls, ship radiated noise, and fish sounds. Directional soundscaping through passive ocean acoustic waveguide remote sensing (POAWRS) provides bearing-time trajectories of signal detections that can be applied for geographical mapping. The received marine mammal sounds include vocalizations from baleen whales such as humpback, fin, and minke, and toothed whales, such as beluga, pilot, and sperm. We provide an insight to the vocalizations and behaviorial patterns of these whales in the Lofoten and Northern Finmark regions. The relative contributions of distinct sound sources, including whales, fish, and ships, to the directional soundscapes are quantified. We also examine their temporo-spatial dependences via geographic mapping of acoustic signals from distinct sound producers.

3:00–3:15 Break

3:15

5pUW9. Deep-water ambient sound on near-bottom receivers in the New England Seamount Chain. Oleg A. Godin (Phys. Dept., Naval Postgrad. School, Monterey, CA 93943, oagodin@nps.edu), Matthew W. Walters (Phys. Dept., Naval Postgrad. School, Monterey, CA), Tsu Wei Tan (Marine Sci., ROC Naval Acad., Kaohsiung, Taiwan), and John E. Joseph (Oceanogr. Dept., Naval Postgrad. School, Monterey, CA)

Ambient sound was recorded for about two months by three synchronized, single-hydrophone receivers that were moored at depths of 2573, 2994, and 4443 m on seamount flanks. Hydrophones were located within 5 m from the seafloor. The data reveal the power spectra and intermittency of the ambient sound intensity in a 13-octave frequency band from 0.5 to 4000 Hz. Statistical distribution of pressure amplitude exhibits much heavier tails than the expected Rayleigh distribution throughout the frequency band of observations. Spatial variability of the observed ambient sound is controlled by the seafloor properties, bathymetric shadowing, and nonuniform distribution of the noise sources on the sea surface due to the Gulf Stream and its meanders. Interferometry of the ambient sound recorded by the receivers with the horizontal separations of 7.0 km shows strong variations of the acoustic travel time between the receivers along a surface-reflected path due to the evolution of the Gulf Stream position. The magnitude of the variations of the passively measured acoustic travel time is consistent with the available contact measurements of the sound speed profiles. Environmental inferences derived from the ambient, shipping, and flow noise data will be discussed. [Work supported by the ONR TFO DRI.]

3:30

5pUW10. Evaluation of the performance of technologies for reducing ships' machinery noise using a small-scale ship-like structure in a Water Basin. Marc-André Guy (Université de Sherbrooke, 4682, rue Mary-Cormier, Québec, QC G1Y3M8, Canada, guym3301@usherbrooke.ca), Kamal Kesour (Innovation Maritime, Rimouski, QC, Canada), Olivier Robin (Université de Sherbrooke, Sherbrooke, QC, Canada), Julien St-Jacques (Innovation Maritime, Rimouski, QC, Canada), and Mathis Vulliez (Université de Sherbrooke, Sherbrooke, QC, Canada)

Ocean ambient levels have increased in the last decades, especially in the low-frequency domain (under 500 Hz). This increase is partly due to underwater radiated noise (URN) from commercial ships. Excessive URN harms marine life and is, therefore, considered a pollution that needs to be reduced. At low speeds, machinery is the primary noise source on ships. Mitigation technologies exist to limit machinery's contribution to URN. While implementing these technologies is costly, a lack of quantitative data regarding their exact performances usually results in limited concrete ship applications since the cost-to-benefit ratio is imprecise. This study aims to quantify better the performance of standard noise mitigation technologies using a small-scale ship-like structure in a water basin. The basin's acoustic field is first characterized with and without the structure. The structure is then equipped with different mitigation technologies. A loudspeaker and a vibration shaker are fed with pink noise or measured signals on actual machinery. Hydrophones, microphones, accelerometers, and force sensors measure the response in the basin and on the structure. The performance of each tested technology is evaluated and ranked in terms of URN reduction. The relative contributions of airborne and structure-borne transmission paths on URN are also examined.

3:45

5pUW11. Comparative field trials of distributed acoustic sensing using helically wound fibers versus straight fibers in cables deployed on lakebed of the Xinfengjiang Reservoir. Chao Li (Southern Marine Sci. and Eng., Guangdong Lab., No. 1119, Haibin Rd., Nansha District, Guangzhou, Guangdong 511458, China, lichao@gmlab.ac.cn), Kan Gao (Shanghai Inst. of Optics and Fine Mech., Chinese Acad. of Sci., Shanghai, China), Haocai Huang (Ocean College, Zhejiang Univ., Zhoushan, Zhejiang, China), Siyuan Cang, Xiaoming Cui, and Huayong Yang (Southern Marine Sci. and Eng., Guangdong Lab., Guangzhou, China)

The application of distributed acoustic sensing (DAS) in underwater acoustics and marine geophysics has faced challenges due to axial and relatively low sensitivity issues associated with ordinary fibers. Helically wound fibers (HWFs), employing an elastic plastic mandrel, represent a primary approach to overcoming the sensitivity shortcomings of straight fibers. This study aims to qualitatively validate the theoretically predicted enhancements and deterioration offered by two types of HWF compared to straight submarine optical cables in underwater applications using one DAS host. Our findings confirm a significant enhancement in the broadband sensitivity of HWF for earthquake, ship noise, and active sources, reaching a maximum improvement of approximately 20 dB at 1000 Hz. In our field data, we observe that all types of cables captured signals ranging from several to 1700 Hz, encompassing the crucial frequency band for underwater acoustic and seismological studies. However, the use of HWF introduced higher background noise in the some frequency bands. The HWF cable, with a 2-cm diameter, is easily manufacturable for kilometer range, enhancing DAS sensitivity comparable to hydrophones. Consequently, HWF emerges as a promising and cost-effective solution for achieving high sensitivity in acoustic and seismic monitoring.

4:00

5pUW12. Efficiency of a cluster of underwater low-frequency sources. Andrey K. Morozov (Marine Syst., Teledyne, 49 Edgerton Dr., North Falmouth, MA 02556, amorozov@teledyne.com)

The efficiency of underwater low-frequency sound sources can be improved by using tunable high-quality resonators, increasing the emission aperture, or using source clusters. A very low frequency tunable resonator is hard to build. Large aperture sources are difficult to deploy. This study shows that the efficiency of a cluster of sources can be much higher than that of each source. Low frequency sources separated by distance operate independently, the sources add up the pressure, and the emitted pressure is proportional to the number of elements in the cluster. As a result, the radiated power of the cluster increases according to the quadratic law of the number of its elements, and the efficiency of the cluster increases in proportion to this number. To prove this, finite element modeling of an array of underwater sources was carried out. The simulation included the internal structure of each sound source with Helmholtz bubble resonators driven by standard audio 2 kW subwoofers. Indeed, the cluster efficiency increases when the distance between the sources exceeds several of their diameters. Simulations showed that a cluster of 32 broadband sources with a frequency of 10–100 Hz could produce sound pressure levels equal to a standard air gun.

4:15

5pUW13. The use of Navy range bottom-mounted, bi-directional transducers for long-term, deep-ocean prey mapping. Alexander Muniz (NUWC, 1178 Howell St., Newport, RI 02841, alexander.p.muniz.civ@us.navy.mil), Stephanie Watwood, Ronald Morrissey (NUWC, Newport, RI), and Kelly Benoit-Bird (Monterey Bay Aquarium Res. Inst. (MBARI), Moss Landing, CA)

Beaked whales have been shown to be particularly sensitive to mid-frequency active sonar but continue to spend considerable time on Navy training ranges, where exposure to sonar is frequent. Understanding the underlying movements of their prey could help to explain the distribution and abundance patterns seen for beaked whales. Beaked whales are among the deepest diving marine mammals and have been shown to feed near the ocean bottom, at depths up to 3000 m. Typical surface-deployed prey mapping technology is not suitable for probing these deep foraging depths. This pilot study aims to us the bi-directional nodes on Navy hydrophone ranges that were developed for underwater communications as a rudimentary prey mapping system. A 1 ms 2 kHz signal was broadcast on an 8 kHz carrier signal and resulting returns were processed to estimate depths of the reflectors. Data are presented from three different time periods on the Southern California Anti-Submarine Warfare Range.

4:30

5pUW14. An overview of the calibration process for a digital hydrophone. Nicolas Wood (Ocean Sonics, 110 Parkway Dr., Truro Heights, NS B6L1N8, Canada, nicolas.wood@oceansonics.com)

Digital hydrophones are instruments for measuring underwater sound that incorporate the entire data chain into a single unit. Calibration of analog

and digital hydrophones is largely similar but differs in measurement signal type. Digital hydrophone calibration is not currently captured in the standards outlining hydrophone calibration (such as IEC 60565), so they currently cannot be calibrated to a recognized standard. The testing setup and calibration methods are the same as for an analog hydrophone but without needing external digitization or analog signal conditioning. Potential issues regarding digital hydrophone calibration: careful calculation of signal amplitudes is required at upper frequency band; the lack of analog output requires different units of measure or information on counts/volt conversion. The first point here is also a current issue for analog hydrophones where the digitization and the processing of data during the calibration are not prescribed. Ideally, both of these issues would be addressed in future standards updates, which would improve the operation of both analog and digital hydrophones. This paper will expand on the issues highlighted here involving digital hydrophone calibration and offer recommendations for best practice. It will also provide an outline of the current calibration procedure as used at Ocean Sonics.

4:45

5pUW15. The ONR three octave research array at Penn State. Chad M. Smith (Appl. Res. Lab., The Penn State Univ., State College, PA 16804, chad.smith@psu.edu), Luigi Troiano (KFM Technol. Resources Inc., La Spezia, Italy), Milan Markovic, and Marco Bernardini (Ctr. for Maritime Res. and Experimentation, La Spezia, Italy)

A new, large diameter towed research array has been built to support the experimental efforts of the Office of Naval Research (ONR) Ocean Acoustics Program. Array development was a collaborative effort between the Centre for Maritime Research and Experimentation (CMRE) in La Spezia, Italy, and the Penn State Applied Research Laboratory (PSU). PSU chose the array specifications based on discussions within the underwater acoustics community, prior measurement experience, cost efficiency, and flexibility in scientific measurements, while CMRE designed and assembled the array. The design includes three, forward-nested acoustic apertures cut for 1, 2, and 4 kHz, and hence, has been dubbed the Three Octave Research Array (THORA) in deference to its predecessor, the Five Octave Research Array (FORA). However, this system has a modular design to allow addition of acoustic apertures if scientific interest and sponsor support warrant. The array currently consists of a 50 m acoustic module and a 25 m vibration isolation module to help isolate the acoustic module from cable strum. This ship towed system has nearly 1 km of tow cable and a maximum measurement depth of 500 m. The array's design, its use in the ONR NESMA23 Pilot experiment, and data quality will be discussed.

5:00

5pUW16. Simultaneous in-air versus in-water measurements of humpback whale breathing sounds. Max Radermacher (Northeastern Univ., 360 Huntington Ave. Boston, MA 02115, radermacher.m@northeastern.edu), Matthew E. Schinault, Sai Geetha Seri, and Purnima Ratilal (Northeastern Univ., Boston, MA)

Humpback whales produce an assortment of sounds ranging from moans and songs to surface-breaching, breathing, and foreflipper flaps. During a sea trial of Northeastern University's coherent hydrophone array at the Great South Channel in US Northeastern coastal waters, the HF subaperture consisting of 32 hydrophone elements at 0.375 m spacings was deployed vertically with half of the hydrophones in air and the other half submerged underwater on September 6th, 2021. Many instances of humpback whale breathing sounds were recorded over several hours of observation. Visual sightings and video recordings were used to coregister sounds recorded on individual hydrophones of the HF subaperture. Here, we examine and compare the simultaneously recorded humpback whale breathing sound spectra, bandwidth, and duration for in-air versus underwater hydrophones. Whale distances to the vertical HF subaperture could be calculated from curved time of arrival differences due to close proximity of the array. These distances are applied to correct the received underwater sound pressure level for transmission loss and estimate underwater recorded whale breathing sound source level. A consistent broadband dip in the measured underwater sound spectra with null centered at 15 kHz is investigated by propagation modelling, considering both modal interference and attenuation from loud bubbles.

5:15

5pUW17. Optimizing hydrophone-preamplifier systems: Theoretical design, computational analysis, and experimental validation for towed array applications. Matthew E. Schinault (Northeastern Univ., 360 Huntington Ave., 409 Dana, Boston, MA 02115, schinault.m@northeastern.edu), Max Radermacher, Jesse Segel, and Purnima Ratilal (Northeastern Univ., Boston, MA)

Hydrophone response parameters greatly influence the design of pre-amplifier and analog to digital conversion (ADC) components. Here, we show a comprehensive methodology for the modeling of hydrophone response, focusing on the theoretical, computational, and experimental approach for low and high frequency SONAR applications. Theoretical calculation of hydrophone response based on electromechanical properties is compared with Finite Element Analysis (FEA) and experimental measurement. An electrical equivalent hydrophone simulator Butterworth-Van Dyke (BVD) circuit is constructed based on admittance and susceptance measurements to represent the transducer's electrical behavior for testing pre-amplifier and ADC response without the presence of environmental or hydrophone noise. This model facilitates a systematic approach to towed array front-end design with hardware in the loop to update models to improve performance prediction. Measurements of the constructed transducer system validate the theoretical and computational models, providing insights into real-world performance.

5:30

5pUW18. Great Lakes fisheries vessels and uncrewed surface vessels: A publicly available dataset. Steven Senczyszyn (Great Lakes Res. Ctr., Michigan Technol. Univ., 1400 Townsend Dr., Houghton, MI 49931, sasenczy@mtu.edu), Andrew Barnard (Acoust., Penn State, University Park, PA), Erik Kocher (Great Lakes Res. Ctr., Michigan Technol. Univ., Houghton, MI), Thomas M. Evans (Natural Resources and the Environment, Cornell Univ., Ithaca, NY), Lars G. Rudstam (Natural Resources and the Environment, Cornell Univ., Bridgeport, NY), Suresh A. Sethi (Natural Resources and the Environment, Cornell Univ., Ithaca, NY), Daniel L. Yule (Great Lakes Sci. Ctr., U.S. Geological Survey, Ashland, WI), David M. Warner, Steven A. Farha, Mark R. Dufour, Timothy P. O'Brien, Patricia M. Dieter (Great Lakes Sci. Ctr., U.S. Geological Survey, Ann Arbor, MI), James J. Roberts (Great Lakes Sci. Ctr., U.S. Geological Survey, Sandusky, OH), James M. Watkins (Natural Resources and the Environment, Cornell Univ., Ithaca, NY), and Peter Esselman (Great Lakes Sci. Ctr., U.S. Geological Survey, Ann Arbor, MI)

For decades, fish abundance surveys have been performed by large, crewed fisheries vessels using echosounders or trawling. However, the estimates obtained by these traditional methods may be biased due to the propagated noise generated by these vessels. A three-year collaborative effort has been conducted to measure the radiated acoustic signature of 17 fisheries vessels operating across all five of the Great Lakes under several operating conditions. This was done using a mobile ship noise measurement system developed specifically for this purpose. Additionally, with the recent advent of uncrewed surface vessels (USVs, "Saildrone"), an alternative "quiet" method of performing these abundance surveys is now possible. A detailed acoustic characterization was conducted for a USV to provide a comparison between traditional "loud" fisheries vessels and these "quiet" surface vessels. This talk will detail the acoustic signatures of all 18 vessels, compare the "loud" and "quiet" vessels, and provide a detailed description of the vessel measurement profiles which are now available as a public dataset. This dataset includes the time domain acoustic measurements, sound speed profiles, and GPS tracks of the measured vessels.