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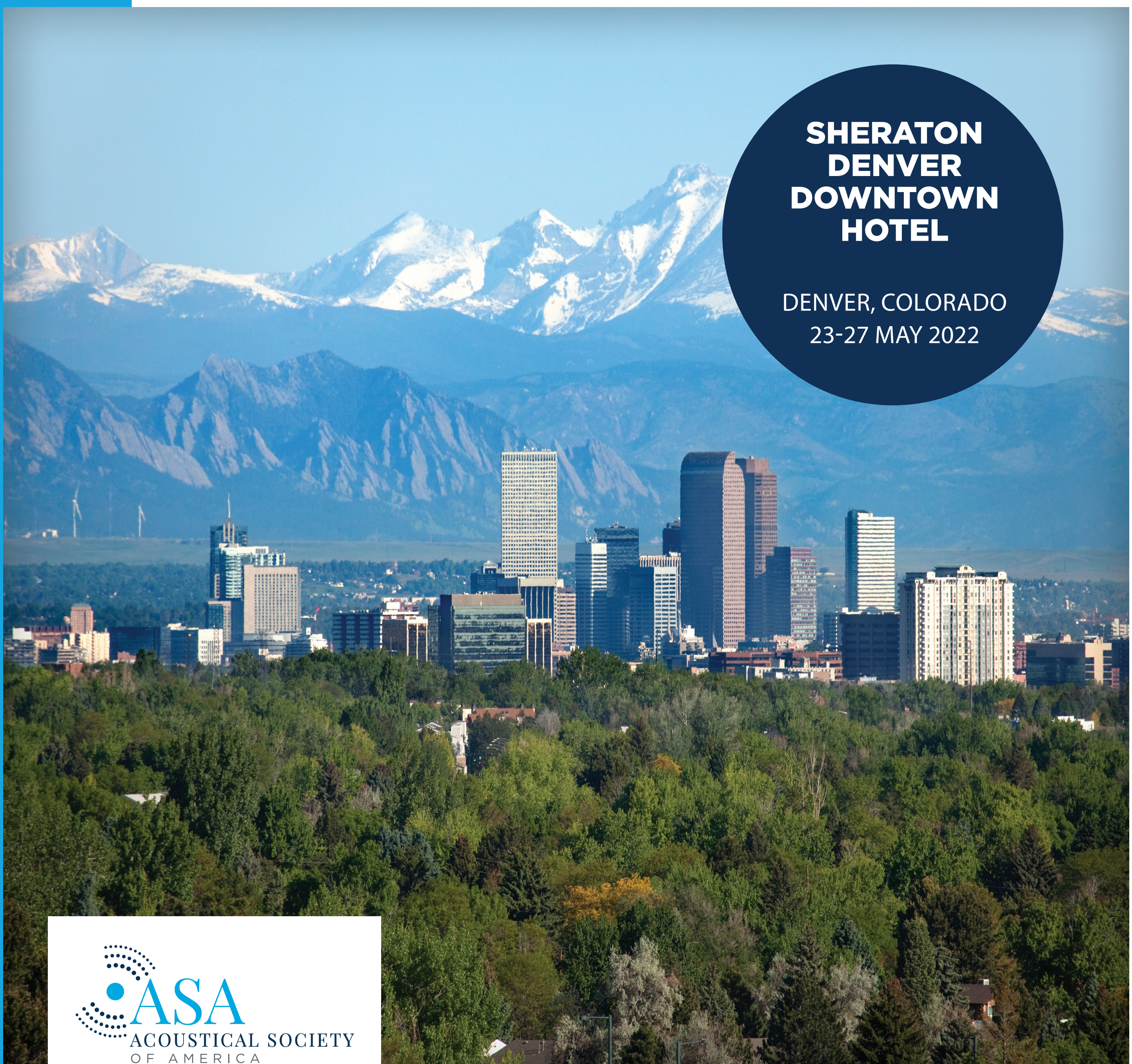

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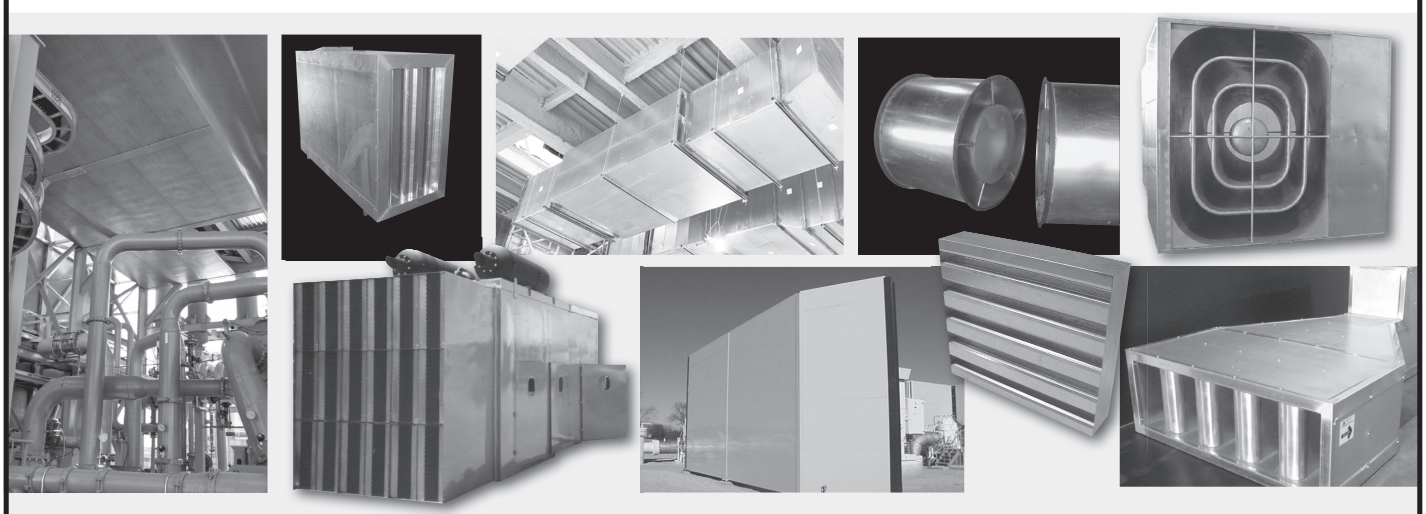

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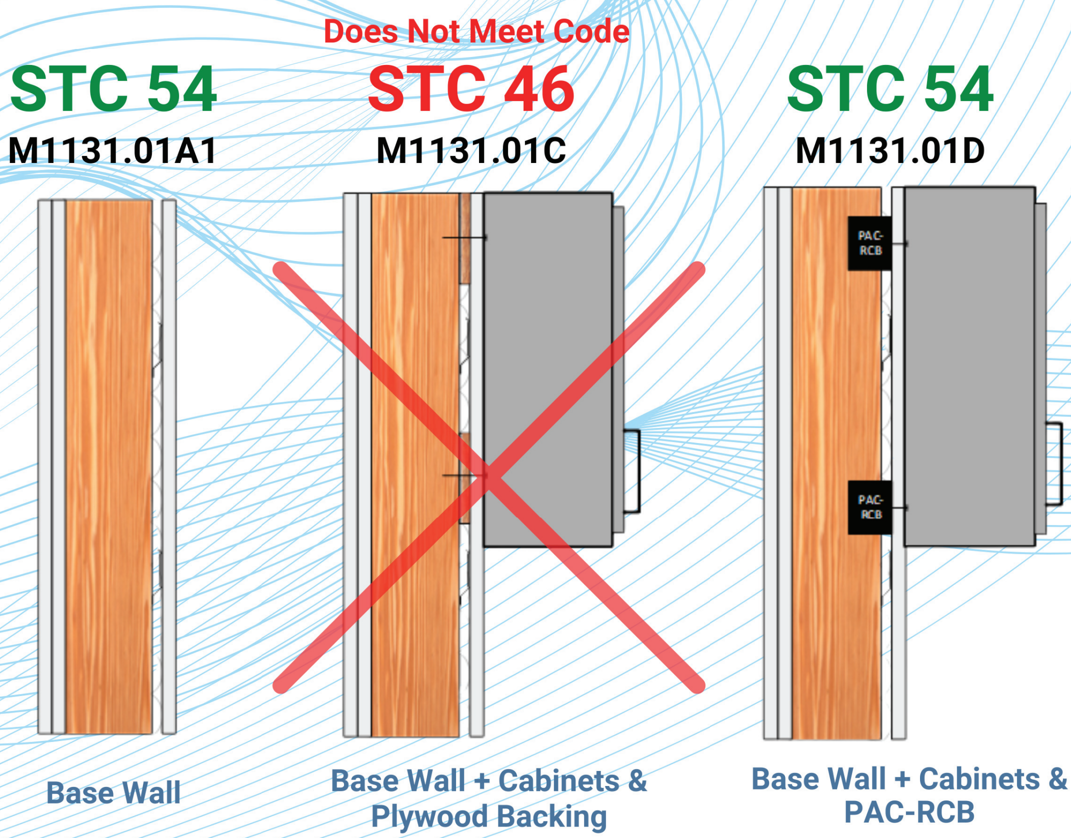

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Tue, 24 May, 2:00 p.m.	Meetings ReImagined	Plaza Court 5
Tue, 24 May, 5:00 p.m.	Women in Acoustics	Directors Row I
Wed, 25 May, 7:30 a.m.	Finance	Plaza Court 3
Wed, 25 May, 7:30 a.m.	Public Relations	Plaza Court 3
Wed, 25 May, 5:00 p.m.	TCAA Speech Privacy	Plaza Court 3
Wed, 25 May, 5:30 p.m.	Improve Racial Diversity and Inclusivity	Plaza Court 5
Thu, 26 May, 7:00 a.m.	Member Engagement	Plaza Court 3
Thu, 26 May, 7:30 a.m.	Editorial Board	Directors Row E
Thu, 26 May, 9:00 a.m.	Revenue Reimagined	Plaza Court 5
Thu, 26 May, 1:00 p.m	Investments	Plaza Court 3
Thu, 26 May, 2:00 p.m.	Strategic Plan Champions	Directors Row E
Fri, 27 May, 8:00 a.m.	Technical Council	Directors Row E
Fri, 27 May, 12:00 p.m.	Executive Council	Directors Row E

TECHNICAL COMMITTEEE OPEN MEETINGS

Tue, 24 May, 4:45 p.m.	Engineering Acoustics	Governors Square 10
Tue, 24 May, 7:30 p.m.	Acoustical Oceanography	Governors Square 14
Tue, 24 May, 7:30 p.m.	Animal Bioacoustics	Governors Square 17
Tue, 24 May, 7:30 p.m.	Architectural Acoustics	Plaza Ballroom A
Tue, 24 May, 7:30 p.m.	Physical Acoustics	Governors Square 11
Tue, 24 May, 7:30 p.m.	Psychological and Physiological Acoustics	Plaza Ballroom D
Tue, 24 May, 7:30 p.m.	Signal Processing in Acoustics	Governors Square 16
Tue, 24 May, 7:30 p.m.	Structural Acoustics and Vibration	Governors Square 12
Wed, 25 May, 7:30 p.m.	Biomedical Acoustics	Governors Square 15
Thu, 26 May, 4:30 p.m.	Computational Acoustics	Governors Square 12
Thu, 26 May, 7:30 p.m.	Musical Acoustics	Directors Row H
Thu, 26 May, 7:30 p.m.	Noise	Plaza Ballroom D
Thu, 26 May, 7:30 p.m.	Speech Communication	Plaza Ballroom E
Thu, 26 May, 7:30 p.m.	Underwater Acoustics	Governors Square 14

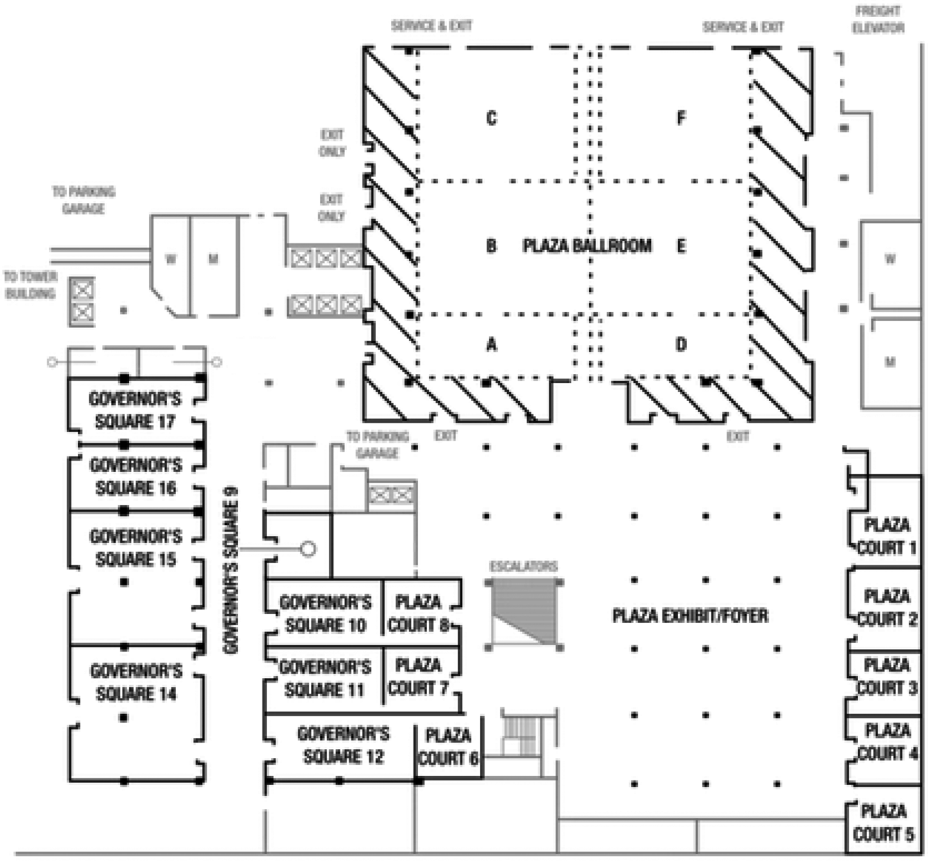
STANDARDS COMMITTEES AND WORKING GROUPS

Mon, 23 May, 5:00 p.m.	ASC SC2-Mechanical Vibration and Shock	Plaza Court 5
Mon, 23 May, 7:00 p.m.	ASACOS Steering	Plaza Court 5
Tue, 24 May, 7:30 a.m.	ASACOS	Directors Row I
Tue, 24 May, 9:15 a.m.	Standards Plenary Including TAGs	Directors Row E
Tue, 24 May, 11:00 a.m.	ASC S1 Acoustics	Directors Row E
Tue, 24 May, 2:00 p.m.	ASC S3 Bioacoustics	Directors Row E
Tue, 24 May, 3:30 p.m.	ASC S3/SC1 Bioacoustics	Directors Row E
Tue, 24 May, 5:00 p.m.	ASC S12 Noise	Directors Row E
Tue, 24 May, 5:00 p.m.	WG44/S12-70	Plaza Court 3

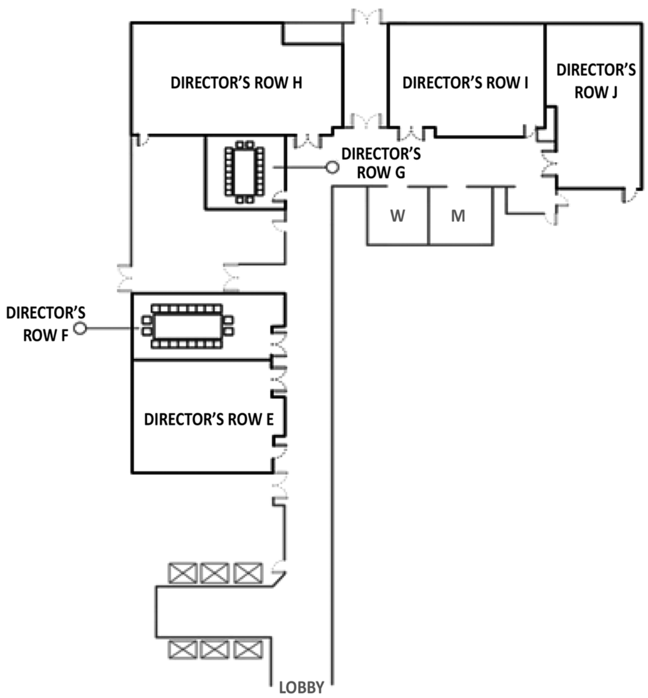
MEEETING SERVICES, SPECIAL EVENTS, SOCIAL EVENTS

Mon-Thu, 23-26 May 7:30 a.m. - 5:00 p.m. Fri, 27 May 7:30 a.m. - 12:00 noon	Registration	Plaza Exhibit Hall
Mon-Thu, 23-26 May 7:00 a.m. - 5:00 p.m. Fri, 27 May, 7:00 a.m. - 12:00 noon	Internet Zone	Plaza Exhibit Hall
Mon-Thu, 23-26 May 7:00 a.m. - 5:00 p.m. Fri, 27 May, 7:00 a.m. - 12:00 noon	A/V Preview	Plaza Court 7
Mon-Thu, 23-26 May, 8:00 a.m. - 5:00 p.m. Fri, 27 May, 8:00 a.m. - 12 noon	Lactation Room	Plaza Court 6
Mon-Fri, 23-27 May 8:00 a.m. - 10:00 a.m.	Accompanying Persons	Plaza Court 4
Mon-Fri, 23-27 May, 9:00 a.m. - 11:00 a.m.	Coffee Breaks	Plaza Exhibit Hall
Mon, 23 May 5:00 p.m. - 5:30 p.m.	First-Time Attendee Orientation	Governors Square 14
Mon, 23 May 5:30 p.m. - 6:45 p.m.	Student Meet and Greet	Windows Room (Tower Building)
Tue, 24 May 6:00 p.m. - 7:30 p.m.	Social Hour	Plaza ABC
Wed, 25 May, 11:45 a.m. - 1:30 p.m.	Women in Acoustics Luncheon	Windows Room (Tower Building)
Wed, 25 May, 4:15 p.m. - 5:30 p.m.	Plenary Session/Awards Ceremony	Plaza ABC
Wed, 25 May, 5:30 p.m.	TCAA Subcommittee on Speech Privacy	Plaza Court 3
Wed, 25 May, 6:00 p.m. - 8:00 p.m.	Student Reception	Windows Room (Tower Building)
Wed, 25 May, 8:00 p.m. - 12:00 midnight	ASA Jam	Plaza Ballroom
Thu, 26 May, 10:00 a.m.	Building Equitable Professional Networks Through Mentored Relationships	Directors Row I
Thu, 26 May, 12:00 noon-2:00 p.m.	Society Luncheon and Lecture	Windows Room (Tower Building)
Thu, 26 May, 5:00 p.m. - 6:00 p.m.	Panel on Careers Beyond Academia	Directors Row I
Thu, 26 May, 2:00 p.m. - 4:00 p.m.	Strategic Champions	Directors Row E
Thu, 26 May, 6:00 p.m. - 7:30 p.m.	Social Hour	Plaza Ballroom ABC

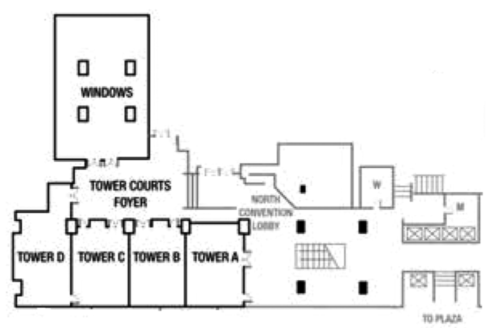
PLAZA BUILDING CONCOURSE LEVEL



PLAZA BUILDING LOBBY LEVEL



I.M. Pei Tower Second Level



TECHNICAL PROGRAM CALENDAR
182nd Meeting of the Acoustical Society of America
23—27 May 2022

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

Monday Morning

9:00 1aAA **Architectural Acoustics and Noise:** Outdoor Performance and Sports Facilities. Plaza Ballroom F

8:20 1aAB **Animal Bioacoustics, Acoustical Oceanography, Signal Processing in Acoustics, and Underwater Acoustics:** Open-Source and Free Tools for Bioacoustics. Governors Square 17

8:45 1aBA **Biomedical Acoustics: General Topics in Biomedical Acoustics:** Bubbles and Cavitation. Governors Square 15

9:00 1aNS **Noise, Architectural Acoustics, and ASA Committee on Standards:** Standards, Codes, and Criteria Applications in the Real World I. Plaza Ballroom E

8:00 1aPAa **Physical Acoustics, Computational Acoustics, Signal Processing in Acoustics, Animal Bioacoustics, Engineering Acoustics, Noise, Architectural Acoustics:** Acoustical Remote Sensing in Urban Environments. Governors Square 11

9:55 1aPAb **Physical Acoustics and Biomedical Acoustics:** Advances in Sonochemistry I. Governors Square 16

8:15 1aSA **Structural Acoustics and Vibration and Physical Acoustics:** Nonlinear Metamaterials and Phononics. Governors Square 12

9:00 1aSC **Speech Communication:** Speech Production and Acoustics I (Poster Session). Plaza Exhibit Hall

8:35 1aUW **Underwater Acoustics and Acoustical Oceanography:** Understanding and Representing Uncertainty in Underwater Acoustic Models I. Governors Square 14

Monday Afternoon

1:30 1pAA **Architectural Acoustics, Noise, Psychological and Physiological Acoustics, and Speech Communication:** Balancing Speech Intelligibility with Privacy for Indoor Spaces. Plaza Ballroom F

1:00 1pBA **Biomedical Acoustics, Physical Acoustics, and Signal Processing in Acoustics:** Emerging Techniques Involving Super-Resolution Imaging Including Image Processing Methods, Applications, and Instrumentation. Governors Square 15

1:00 1pCAa **Computational Acoustics, Biomedical Acoustics, Physical Acoustics, Underwater Acoustics, Structural Acoustics and Vibration, Noise, Musical Acoustics, and Architectural Acoustics:** Finite and Boundary Element Methods Across Acoustics. Governors Square 12

2:25 1pCAb **Computational Acoustics, Signal Processing in Acoustics, Biomedical Acoustics, Physical Acoustics, and Structural Acoustics and Vibration:** Contributions in Emerging Methods for Design and Optimization in Computational Acoustics. Governors Square 12

1:00 1pEA **Engineering Acoustics:** General Topics in Engineering Acoustics. Plaza Ballroom A

1:00 1pMU **Musical Acoustics:** General Topics in Musical Acoustics I. Directors Row H

1:00 1pPA **Physical Acoustics and Biomedical Acoustics:** Advances in Sonochemistry II. Governors Square 16

1:00 1pSC **Speech Communication:** Speech Production and Acoustics II (Poster Session). Plaza Exhibit Hall

1:00 1pUW **Underwater Acoustics and Acoustical Oceanography:** Understanding and Representing Uncertainty in Underwater Acoustic Models II. Governors Square 14

Monday Afternoon

4:00 1eID **Interdisciplinary:** Keynote Lecture. Plaza Ballroom D

Tuesday Morning

8:25 2aAA **Architectural Acoustics:** Acoustic Comfort in Healthcare Facilities I. Plaza Ballroom A

8:20 2aAB **Animal Bioacoustics, Acoustical Oceanography, and Underwater Acoustics:** Whitlow Au Memorial Session I. Governors Square 17

8:55 2aBAa **Biomedical Acoustics:** New Developments in Lung Ultrasound I. Governors Square 14

9:00 2aBAb **Biomedical Acoustics:** For a Few Bubbles More: Recent Developments in Medical Ultrasound I. Governors Square 15

9:00 2aMU **Musical Acoustics and Education in Acoustics:** Strategies for Online and Hybrid Teaching of Musical Acoustics. Directors Row H

9:00 2aNS **Noise:** Rockets, Wind Turbines, and Other Topics on Noise. Plaza Ballroom F

7:55 2aPAa **Physical Acoustics, Computational Acoustics, Noise, and Signal Processing in Acoustics:** Progress in Sonic Boom Modeling, Processing, and Analysis on Community Response. Governors Square 10

7:55 2aPAb **Physical Acoustics, Biomedical Acoustics, and Structural Acoustics and Vibration:** Vortex Beams and Radiation Torques I. Governors Square 11

8:30 2aPP **Psychological and Physiological Acoustics and Speech Communication:** Age-Related Changes in Mechanisms of Speech Perception. Plaza Ballroom D

8:15 2aSAa **Structural Acoustics and Vibration and Physical Acoustics:** Willis Coupling in Acoustic and Elastic Media. Governors Square 12

10:00 2aSAb **Structural Acoustics and Vibration, Engineering Acoustics, and Physical Acoustics:** Acoustic Metamaterials I. Governors Square 12

9:00 2aSC **Speech Communication:** Race, Racialization, and Racism in Speech Perception. Plaza Ballroom E

8:30 2aSP **Signal Processing in Acoustics, Underwater Acoustics, Engineering Acoustics, Physical Acoustics, and Animal Bioacoustics:** Cognitive SONAR. Governors Square 16

Tuesday Afternoon

1:05 2pAA **Architectural Acoustics:** Acoustic Comfort in Healthcare Facilities II. Plaza Ballroom A

1:00 2pAB **Animal Bioacoustics:** Whitlow Au Memorial Session II and Underwater Acoustics. Governors Square 17

1:00 2pBAa **Biomedical Acoustics:** For a Few Bubbles More: Recent Developments in Medical Ultrasound II. Governors Square 15

1:00 2pBAb **Biomedical Acoustics:** New Developments in Lung Ultrasound II. Governors Square 14

1:00 2pED **Education in Acoustics, Engineering Acoustics, and Architectural Acoustics:** Connecting Industry and Education. Directors Row H

2:00 2pNS **Noise, Architectural Acoustics, and ASA Committee on Standards:** Standards, Codes, and Criteria Applications in the Real World II. Plaza Ballroom E

1:00 2pPAa **Physical Acoustics, Noise, Biomedical Acoustics, and Computational Acoustics:** Sonic Boom Focusing Predictions. Governors Square 10

1:00 2pPAb **Physical Acoustics, Biomedical Acoustics, and Structural Acoustics and Vibration:** Vortex Beams and Radiation Torques II. Governors Square 11

1:00 2pPPa **Psychological and Physiological Acoustics:** Animal Physiology. Plaza Ballroom D

2:00 2pPPb **Psychological and Physiological Acoustics:** Clinical Populations and Devices I (Poster Session). Plaza Exhibit Hall

1:15 2pSA **Structural Acoustics and Vibration, Engineering Acoustics, and Physical Acoustics:** Acoustic Metamaterials II. Governors Square 12

1:00 2pSC **Speech Communication:** Instrumentation and Method for Speech Analysis (Poster Session). Plaza Exhibit Hall

1:15 2pSP **Signal Processing in Acoustics:** General Topics in Signal Processing I. Governors Square 16

1:00 2pUW **Underwater Acoustics and Acoustical Oceanography:** Collaborative Measurements in Underwater Acoustics: A Memorial Session for John R. Preston. Plaza Ballroom F

Wednesday Morning

8:15 3aAA **Architectural Acoustics, ASA Committee on Standards, Noise, Physical Acoustics, and Biomedical Acoustics:** Advanced Measurement and Modeling of Sound Absorption and Scattering I. Plaza Ballroom F

8:20 3aAB **Animal Bioacoustics:** Animal Bioacoustics: Behavior and Physiology. Governors Square 17

8:30 3aAO **Acoustical Oceanography:** Acoustic Sensing of Biological and Physical Processes in Littoral Environments. Governors Square 14

8:00 3aBA **Biomedical Acoustics and Physical Acoustics:** Bubble-Cell Interaction I. Governors Square 15

8:00 3aEA **Engineering Acoustics, Structural Acoustics and Vibration, and Physical Acoustics:** Smart Metamaterials and Metastructures I. Governors Square 10

9:00 3aMU **Musical Acoustics:** General Topics in Musical Acoustics II. Directors Row H

8:05 3aPA **Physical Acoustics and Noise:** Meteorological Acoustics I. Governors Square 11

7:50 3aPPa **Psychological and Physiological Acoustics:** Binaural Hearing. Plaza Ballroom D

9:55 3aPPb **Psychological and Physiological Acoustics:** Clinical Populations and Devices II. Plaza Ballroom D

10:00 3aSA **Structural Acoustics and Vibration, Computational Acoustics, and Engineering Acoustics:** Dynamic Substructuring Techniques and Their Application to Structural Acoustics. Governors Square 12

8:30 3aSC **Speech Communication:** Children's Speech Intelligibility I. Plaza Ballroom E

Wednesday Afternoon

1:30 3pAA **Architectural Acoustics, ASA Committee on Standards, Noise, Physical Acoustics, and Biomedical Acoustics:** Advanced Measurement and Modeling of Sound Absorption and Scattering II. Plaza Ballroom F

1:00 3pBAa **Biomedical Acoustics and Physical Acoustics:** Bubble-Cell Interaction II. Governors Square 15

1:00 3pBAb **Biomedical Acoustics:** Biomedical Acoustics Best Student Paper Poster Session. Plaza Exhibit Hall

1:00 3pCA **Computational Acoustics, Signal Processing in Acoustics, and Structural Acoustics and Vibration:** Application of Model Reduction Across Acoustics. Governors Square 12

1:00 3pEA **Engineering Acoustics, Structural Acoustics and Vibration, and Physical Acoustics:** Smart Metamaterials and Metastructures II. Governors Square 10

2:40 3pID **Interdisciplinary:** Hot Topics in Acoustics. Plaza Ballroom D

1:00 3pMU **Musical Acoustics:** Evolution and Maturation of Musical Acoustics. Directors Row H

1:00 3pPA **Physical Acoustics and Noise:** Meteorological Acoustics II. Governors Square 11

1:20 3pPP **Psychological and Physiological Acoustics:** Auditory Neuroscience Prize Lecture. Plaza Ballroom D

1:00 3pSC **Speech Communication:** Children's Speech Intelligibility II. Plaza Ballroom E

1:00 3pSP **Signal Processing in Acoustics:** General Topics in Signal Processing II. Governors Square 16

1:00 3pUW **Underwater Acoustics:** General Topics in Underwater Acoustics I: Modeling and Measurements. Governors Square 14

Thursday Morning

9:00 4aAA **Architectural Acoustics, Musical Acoustics, Education in Acoustics, and Noise:** Music Education Facilities I. Plaza Ballroom F

8:00 4aBA **Biomedical Acoustics:** General Topics in Biomedical Acoustics: Elastography and Therapeutics. Governors Square 15

8:00 4aNS **Noise and Psychological and Physiological Acoustics:** Health Effects of Noise. Plaza Ballroom D

8:30 4aPA **Physical Acoustics:** General Topics in Physical Acoustics I. Governors Square 11

9:00 4aPP **Psychological and Physiological Acoustics:** Psychophysics, Methods, Models (Poster Session). Plaza Exhibit Hall

8:30 4aSAa **Structural Acoustics and Vibration:** Absorption, Isolation, and Noise Mitigation. Governors Square 12

10:30 4aSAb **Structural Acoustics and Vibration:** Acoustic Radiation from Structures. Governors Square 12

9:00 4aSC **Speech Communication:** Perspectives on Long-Distance Coarticulation. Plaza Ballroom E

8:30 4aSP **Signal Processing in Acoustics, Computational Acoustics, Underwater Acoustics, and Physical Acoustics:** Model Based Signal Processing, Bayesian Learning, and Machine Learning I. Governors Square 16

8:30 4aUW **Underwater Acoustics:** General Topics in Underwater Acoustics II. Governors Square 14

Thursday Afternoon

1:30 4pAA **Architectural Acoustics, Musical Acoustics, Education in Acoustics, and Noise:** Music Education Facilities II. Plaza Ballroom F

1:00 4pAB **Animal Bioacoustics, Noise, Acoustical Oceanography, and Underwater Acoustics:** Acoustical Impacts and Monitoring Protocols Associated with Offshore Windfarms. Governors Square 17

1:20 4pAO **Acoustical Oceanography:** Topics in Acoustical Oceanography. Governors Square 14

1:30 4pBA **Biomedical Acoustics, Education in Acoustics, and Physical Acoustics:** Ultrasound and Its Role in Powering, Sensing and Communicating for Medical and Non-Medical Applications. Governors Square 15

1:00 4pEA **Engineering Acoustics, Education in Acoustics Architectural Acoustics, Noise, and Physical Acoustics:** Low Cost Acoustical Measurement Systems. Governors Square 10

1:00 4pNS **Noise, Architectural Acoustics, Signal Processing in Acoustics, Computational Acoustics, and Psychological and Physiological Acoustics:** Soundscape and Virtual Reality. Plaza Ballroom A

1:00 4pPA **Physical Acoustics, Structural Acoustics and Vibration, Signal Processing in Acoustics, Engineering Acoustics, and Computational Acoustics:** Machine Learning in Acoustic Metamaterials. Governors Square 11

1:30 4pPP **Psychological and Physiological Acoustics:** Human and Animal Physiology (Poster Session). Plaza Exhibit Hall

1:00 4pSC **Speech Communication:** Speech Perception I (Poster Session). Plaza Exhibit Hall

1:00 4pSP **Signal Processing in Acoustics, Computational Acoustics, Underwater Acoustics, and Physical Acoustics:** Model Based Signal Processing, Bayesian Learning, and Machine Learning II. Governors Square 16

Friday Morning

8:30 5aAA **Architectural Acoustics, ASA Committee on Standards, and Noise:** Courts and Municipal Buildings. Plaza Ballroom F

8:00 5aPA **Physical Acoustics:** General Topics in Physical Acoustics II. Governors Square 11

9:00 5aSC **Speech Communication:** Speech Perception II (Poster Session). Plaza Exhibit Hall

9:00 5aUW **Underwater Acoustics:** General Topics in Underwater Acoustics III. Governors Square 14

182nd Meeting of the Acoustical Society of America

The 182nd meeting of the Acoustical Society of America will be held Monday through Friday, 23-27 May 2022 at the Sheraton Denver Downtown Hotel, Denver Colorado, USA.

SECTION HEADINGS

1. COVID-19 VACCINATION REQUIREMENT FOR ATTENDANCE

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

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

2. HOTEL INFORMATION

The Sheraton Denver Downtown is the headquarters hotel where all meeting events will be held.

The cut-off date for reserving rooms at special rates has passed. Please contact the Sheraton (1550 Court Pl, Denver, CO 80202, Phone: (303) 893-3333) for information about room availability.

3. TRANSPORTATION AND TRAVEL

Visit https://www.denver.org/listing/denver-international-airport/4065/ for ground transportation options between the Denver airport and the Sheraton Denver Downtown Hotel.

4. REGISTRATION

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

Registration will open on Monday, 23 May, at 7:00 a.m. in the Plaza Exhibit Hall (see floor plan on page A10).

Checks in U.S. funds drawn on U.S. banks and Visa, MasterCard and American Express credit cards will be accepted for payment of registration. Meeting attendees who have pre-registered may pick up their badges and registration materials at the pre-registration desk.

The registration fees (in USD) are $725 for members of the Acoustical Society of America; $925 for non-members, $250 for Emeritus members (Emeritus status pre-approved by ASA), $425 for ASA Early Career members (for ASA members within three years of their most recent degrees – proof of date of degree required), $200 for ASA Student members, $300 for students who are not members of ASA, $25 for Undergraduate Students, and $250 for accompanying persons.

One-day registration is available at $425 for members and $500 for nonmembers (one-day means attending the meeting on only one day either to present a paper and/or to attend sessions). A nonmember who pays the $925 nonmember registration fee and simultaneously applies for Associate Membership in the Acoustical Society of America will be given a $50 discount off their dues payment for 2022 dues.

Invited speakers who are members of the Acoustical Society of America are expected to pay the Member full-week or one-day registration fees. Nonmember invited speakers who participate in the meeting only on the day of their presentation may register without charge. The registration fee for nonmember invited speakers who wish to participate for more than one day is $400 and includes a one-year Associate Membership in the ASA upon completion of an application form.

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

5. ACCESSIBLITY

If you have special accessibility requirements, please indicate this by informing ASA (1305 Walt Whitman Road, Suite 110, Melville, NY 11747-4300; asa@acousticalsociety.org) at a minimum of thirty days in advance of the meeting. Please provide a cell phone number, email address, and detailed information including the nature of the special accessibility so that we may contact you directly.

6. TECHNICAL SESSIONS

The technical program includes 93 sessions with over 800 abstracts scheduled for presentation during the meeting.

A floor plan of the Sheraton Denver Downtown Hotel appears on page A10. Session Chairs have been instructed to adhere strictly to the printed time schedule, both to be fair to all speakers and to permit attendees to schedule moving from one session to another to hear specific papers. If an author is not present to deliver a lecture-style paper, the Session Chairs have been instructed either to call for additional discussion of papers already given or to declare a short recess so that subsequent papers are not given ahead of the designated times.

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

7. TECHNICAL SESSION DESIGNATIONS

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

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

1-Monday, 23 May
2-Tuesday, 24 May
3-Wednesday, 25 May
4-Thursday, 26 May
5-Friday, 27 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.

8. HOT TOPICS SESSION

The Hot Topics session (3pID) will be held on Wednesday, 25 May, at 2:40 p.m. in Plaza Ballroom D.

9. OTHER SPECIAL EVENTS

The Member Engagement Committee has organized Careers Beyond Academia Panel oriented towards those in academia who want to expand their careers into industry, form a small business, or become a consultant. Panelists have a mixture of academic and industry experience. Panelists will discuss the motivation for moving beyond academia, the difficulties making the transition, and challenges in having a dual career in academia and private industry. Panelists will be available for one-on-one discussion at the end of the panel. Panel sponsored by the Membership Engagement Committee and Task Force for Industry and Practitioners. Organizers: Derek Olson, Diane Kewley-Port, and Derrick Knight.

The panel will be held on 26 May, 4:00 p.m. to 6:00 p.m. in Directors Row I.

A workshop on Building Equitable Professional Networks Through Mentored Relationship has been organized by the Member Engagement Committee and the Committee to Improve Racial Diversity and Inclusivity.

ASA has made a commitment to fostering inclusion in all that we do. A first step is for the society to provide access to professional growth and development opportunities for members of historically marginalized groups. ASA recognizes that our society is better when we work together. Access is not enough. Our success in this initiative depends on developing formal and informal networks of mentors, fellows, and sponsors. This workshop will provide an overview on developing networking and mentoring relationships with peers and students including members of historically marginalized groups. Our goal is to ensure members of historically marginalized groups are welcomed into our society and receive opportunities to advance in the field of acoustics in an engaged and an inclusive environment. The workshop will focus on the process of building a professional network, developing effective mentoring relationships, and connecting as peer-to-peer mentors. This workshop is intended for professionals and students from all career levels from both academic institutions and industry. Organizers are Yolanda Holt, Susannah Levi, and Kimberly Riegel.

The workshop will be held on Thursday, 26 May, 10:00 a.m. in Directors Row I.

10. HARTMANN PRIZE IN AUDITORY NEUROSCIENCE AND AUDITORY NEUROSCIENCE PRIZE LECTURE

The Hartmann Prize in Auditory Neuroscience will be presented at the Denver meeting during the Plenary Session on Wednesday, 25 May.

Dr. Philip Joris will present the Auditory Neuroscience Prize Lecture titled “Coincidences and Delays in Disguise” in session 3pPP 1:20 p.m. to 2:25 p.m. in Plaza Ballroom D.

11. TECHNICAL COMMITTEE OPEN MEETINGS

Technical Committees will hold open meetings on Tuesday, Wednesday, and Thursday. The schedule and rooms for each Committee meeting are given on page A9.

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.

12. PLENARY SESSION AND AWARDS CEREMONY

A plenary session will be held Wednesday, 25 May, at 4:30 p.m. in Plaza Ballroom ABC.

ASA scholarship recipients will be introduced. The Hartmann Prize in Auditory Neuroscience, the Silver Medal in Noise, the R. Bruce Lindsay Award, the Helmholtz-Rayleigh Interdisciplinary Silver Medal in Architectural Acoustics and Engineering Acoustics, and the Gold Medal will be presented, Certificates will be presented to Fellows elected at the Seattle meeting. See page A192 for a list of fellows and award recipients.

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

13. ANSI STANDARDS COMMITTEES

Meetings of ANSI Accredited Standards Committees will be held at the Denver meeting. Please see the Schedule of Committee Meetings and Other Events on page A9 for the schedule of Standards Committee meetings and Standards working group meetings.

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

14. COFFEE BREAKS

Morning coffee breaks will be held from 9:45 a.m. to 11:00 a.m. in the Plaza Exhibit Hall.

15. A/V PREVIEW ROOM

Plaza Court 7 will be set up as an A/V preview room for authors’ convenience and will be available on Monday through Thursday from 7:00 a.m. to 5:00 p.m. and Friday from 7:00 a.m. to 12:00 noon.

16. INTERNET ZONE

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

Tables with power cords will be set up in the Plaza Exhibit Hall for attendees to gather and to power-up their electronic devices.

17. LACTATION ROOM

A lactation room for ASA meeting attendees will be available Monday to Friday, 23-27 May in Plaza Court 6. The hours are Monday to Thursday, 8:00 a.m. to 5:00 p.m. and Friday, 8:00 a.m. to 12:00 noon.

18. PROCEEDINGS OF MEETINGS ON ACOUSTICS (POMA)

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

19. SOCIALS

Socials will be held on Tuesday and Thursday evenings, 6:00 p.m. to 7:30 p.m. in the Plaza Ballroom.

The ASA hosts these social hours to provide a relaxing setting for meeting attendees to meet and mingle with their friends and colleagues as well as an opportunity for new members and first-time attendees to meet and introduce themselves to others in the field. A second goal of the socials is to provide a sufficient meal so that meeting attendees can attend the open meetings of Technical Committees that begin immediately after the socials

20. SOCIETY LUNCHEON AND LECTURE

The Society Luncheon and Lecture, sponsored by the College of Fellows, will be held Thursday, 26 May, at 12:00 noon in the Windows Room (Tower Building).

This luncheon is open to all attendees and their guests Purchase your tickets at the Registration Desk before 10:00 a.m. on Wednesday, 25 May. The cost is USD $30.00 per ticket.

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

Follow the student twitter throughout the meeting @ ASAStudents,

A New Students/First-Time Attendee Orientation will be held on Monday, 23 May, from 5:00 p.m. to 5:30 p.m in Governors Square 14. This will be followed by the Student Meet and Greet from 5:30 p.m. to 6:45 p.m. in the Windows Room (Tower Building) where refreshments and a cash bar will be available.

The Students’ Reception will be held on Wednesday, 25 May, from 6:00 p.m. to 8:00 p.m. in the Windows Room (Tower Building). 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.

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

22. WOMEN IN ACOUSTICS LUNCHEON

The Women in Acoustics luncheon will be held at 11:45 a.m. on Wednesday, 25 May, in the Windows Room (Tower Building). Those who wish to attend must purchase their tickets in advance by 10:00 a.m. on Tuesday, 24 May. The fee is USD $30 for non-students and USD $15 for students.

23. JAM SESSION

You are invited to the JAM on Wednesday night, 25 May, from 8:00 p.m. to midnight to be held at the Sheraton Hotel in the Plaza Ballroom. 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.

24. ACCOMPANYING PERSONS PROGRAM

Spouses and other visitors are welcome at the Denver meeting. The on-site registration fee for accompanying persons is USD $250. A hospitality room for accompanying persons will be open in Plaza Court 4 8:00 a.m. to 10:00 a.m. Monday through Friday. This entitles you access to the accompanying persons room, social events on Tuesday and Thursday, the Jam Session, and the Plenary Session on Wednesday afternoon.

25. WEATHER

Spring brings showers and snow but also sunshine. Temperatures are pleasant for most of the spring, and many outdoor attractions begin opening mid- to late-season. The increasing temperature and sunshine bring a spring bloom to the city which attracts birds and butterflies, especially in April and May. It can get cool at night, but freezing temperatures are generally uncommon after mid-spring, which makes it an excellent time to experience the city. Average temperatures: High: 71 degrees F; Low: 44 degrees F.

26. TECHNICAL PROGRAM ORGANIZING COMMITTEE

Carrie Wall, Technical Program Chair; Christopher Bassett, Acoustical Oceanography; Michael Smotherman, Animal Bioacoustics; David Manley, Brandon Cudequest, Architectural Acoustics; Kang Kim, Libertario Demi, Biomedical Acoustics; Amanda Hanford, Computational Acoustics; Daniel Russell, Education in Acoustics; Michael Haberman, Thomas Blanford, Engineering Acoustics; Kurt Hoffman, Taffeta Elliott, Musical Acoustics; Aaron Vaughn, James Phillips, Hales Swift; Noise; Kevin Lee, Samuel Wallen, Physical Acoustics; Ellen Peng, Gregory Ellis, Psychological and Physiological Acoustics; Kai Gemba, Trevor Jerome, Signal Processing in Acoustics; Christina Zhao, Rajka Smiljanic, Matthew Masapollo, Matthew Winn, Speech Communication; Anthony Bonomo, Stephanie Konarski, Structural Acoustics and Vibration; Arjun Song, Underwater Acoustics.

27. MEETING ORGANIZING COMMITTEE

Dana S. Hougland, Chair; Carrie Wall, Technical Program Chair; Michael Calvisi, Stuart McGregor, *Signs* Monique Alexander, Accompanying Persons, Room Monitors.

28. PHOTOGRAPHING AND RECORDING

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

29. ABSTRACT ERRATA

This meeting program is Part 2 of the April 2022 issue of *The Journal of the Acoustical Society of America*. Corrections, for printer's errors only, may be submitted for publication in the Errata section of the *Journal*.

30. GUIDELINES FOR ORAL PRESENTATIONS, Preparation of Visual Aids

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

- If using light backgrounds (white, off-white), use dark blue, dark brown or black lettering.
- 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 very important that you check that all multi-media elements (e.g., sounds or videos) play accurately prior to the day of your session.
- Each time you display a visual aid the audience needs time to interpret it. Describe the abscissa, ordinate, units, and the legend for each figure. If the shape of a curve or some other feature is important, tell the audience what they should observe to grasp the point. They won't have time to figure it out for themselves. A popular myth is that a technical audience requires a lot of technical details. Less can be more.
- Turn off your cell phone prior to your talk and put it away from your body. Cell phones can interfere with the speakers and the wireless microphone.

31. SUGGESTIONS FOR EFFECTIVE POSTER PRESENTATIONS

Content

- The poster should be centered around two or three key points supported by the title, figures, and text. The poster should be able to "stand alone." That is, it should be understandable even when you are not present to explain, discuss, and answer questions. This quality is highly desirable since you may not be present the entire time posters are on display, and when you are engaged in discussion with one person, others may want to study the poster without interrupting an ongoing dialogue.
- To meet the "stand alone" criteria, it is suggested that the poster include the following elements, as appropriate:
- Background
- Objective, purpose, or goal
- Hypotheses
- Methodology
- Results (including data, figures, or tables)
- Discussion
- Implications and future research
- References and Acknowledgment

Design and layout

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

Lettering and text

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

Visuals

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

Presentation

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

Other suggestions

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

32. GUIDELINES FOR USE OF COMPUTER PROJECTION

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

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

Introduction

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

Guidelines

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

Specific Hardware Configurations
Macintosh

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

- Hook everything up before powering anything on. (Connect the computer to the RGB input on the projector).
- Turn the projector on and boot up the Macintosh. If this doesn't work immediately, you should make sure that your monitor resolution is set to 1024x768 for an XGA projector or at least 640x480 for an older VGA projector. (1024x768 will most always work.). You should also make sure that your monitor controls are set to mirroring.
- If it's an older PowerBook, it may not have video mirroring, but something called simulscan, which is essentially the same.
- Depending upon the vintage of your Mac, you may have to reboot once it is connected to the computer projector or switcher. Hint: you can reboot while connected to the computer projector in the A/V preview room in advance of your presentation, then put your computer to sleep. Macs thus booted will retain the memory of this connection when awakened from sleep.
- Depending upon the vintage of your system software, you may find that the default video mode is a side-by-side configuration of monitor windows (the test for this will be that you see no menus or cursor on your desktop; the cursor will slide from the projected image onto your laptop's screen as it is moved). Go to Control Panels, Monitors, configuration, and drag the larger window onto the smaller one. This produces a mirror-image of the projected image on your laptop's screen.
- Also depending upon your system software, either the Control Panels will automatically detect the video projector's resolution and frame rate, or you will have to set it manually. If it is not set at a commensurable resolution, the projector may not show an image. Experiment ahead of time with resolution and color depth settings in the A/V preview room (please don't waste valuable time adjusting the Control Panel settings during your allotted session time).

PC

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

Linux

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

33. DATES OF FUTURE ASA MEETINGS

For further information on any ASA meeting, or to obtain instructions for the preparation and submission of meeting abstracts, contact the Acoustical Society of America, 1305 Walt Whitman Road, Suite 300, Melville, NY 11747-4300; Telephone: 516-576-2360; Fax: 631-923-2875; E-mail: asa@acousticalsociety.org

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

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

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

189th Meeting, joint with the International Commission for Acoustics, New Orleans, Louisiana, May 2025

ANNUAL GIVING TO THE ACOUSTICAL SOCIETY FOUNDATION FUND – 2021

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Session 1aAA

Architectural Acoustics and Noise: Outdoor Performance and Sports Facilities

Bonnie Schnitta, Cochair
SoundSense, LLC, 39 Industrial Rd., Unit 6, PO Box 1360, Wainscott, NY 11975

Ted Pyper, Cochair
K2, 5777 Central Ave., Suite 225, Boulder, CO 80301

Chair's Introduction—9:00

Invited Papers

9:05

1aAA1. Actions and mathematical modeling that will bring noise levels from a racetrack or raceway to a level the community will accept. Bonnie Schnitta (SoundSense, LLC, 39 Industrial Rd., Unit 6, PO Box1360, Wainscott, NY 11975, bonnie@soundsense.com), Margot Criscitiello, Sean Harkin, Patrick Murray, and Collin G. Champagne (SoundSense, LLC, Wainscott, NY)

Historically, new and existing racetracks and raceways encounter conflict between owners, racecar drivers, and the surrounding community. Racecar drivers enjoy the thrill of a raceway, but neighboring residents often complain about the noise negatively impacting the quiet enjoyment of their homes. This is true even when the homes are near a major highway or road. Raceways and neighboring communities are attempting to find workable solutions without compromise to the safety and enjoyment of the raceway. The presentation discusses objective information used to assist communities or town boards, nearby neighbors, and track owners to engage in productive dialogue of the outcome of the possible solution sets. Multiple solution sets are discussed which are typically acceptable to all parties, including various barriers and other innovative noise mitigation plans. The mathematical modeling and analysis of the topography around the track is presented to show how the local terrain can be used to help to achieve the required level of track noise reduction. The information will be presented through the lenses of three case studies. Two studies demonstrate solutions for specific raceways. The other case study is used to further emphasize the importance of incorporating the local terrain into the solution set.

9:25

1aAA2. Threading the needle between acoustic performance and fabric tensile structures. Shane J. Kanter (Threshold Acoust., 141 W Jackson Blvd, Ste. 2080, Chicago, IL 60604, skanter@thresholdacoustics.com), Jennifer Nelson Smid (Threshold Acoust., Chicago, IL), Carl P. Giegold (Threshold Acoust., Evanston, IL), Dawn Schuette, Robin Glosemeyer Petrone, and Chris Springthorpe (Threshold Acoust., Chicago, IL)

Fabric tensile structures are a common architectural response to the need for an outdoor performance venue because they provide a lightweight, durable shelter, capable of spanning large distances. Particular care is necessary when threading the needle between efficient structural form and functional acoustic performance. This paper will discuss the acoustic design considerations necessary to yield the best acoustic results when designing outdoor fabric tensile structures. Concepts such as geometry, materiality, and amplified audio will be explored through the story of a few project examples.

9:45

1aAA3. The Dr. Phillips Center's Front Yard Festival—A pandemic response that just might outlive the pandemic. Scott A. Crossfield (Theatre Projects, 47 Water St., South Norwalk, CT 06854, scrossfield@theatreprojects.com)

When the doors of a Performing Arts Center are closed due to Covid, how can they carry on their mission to engage their community in a safe, socially distanced way? Join Scott Crossfield ASTC from Theatre Projects as he tells the story of how the Dr. Phillips Center in Orlando, Florida created the Front Yard Festival. Conceived as an innovative, yet temporary, 2000-seat outdoor venue intended to bring much needed live entertainment, culture, and wellness to the community during the pandemic, it has been so successful that the festival will become a permanent fixture for years to come.

10:05

1aAA4. Acoustics improvements to an outdoor music venue in an urban setting: A case study. Tim Gulsrud (Soundpost Acoust., LLC, 229 Terry St., Ste. #4, Longmont, CO 80501, tgulsrud@soundpostacoustics.com)

This paper presents a case study of acoustics improvements made to the Number 38 Social Hall located in Denver's RiNo Arts District. The venue includes an outdoor music stage which is close to adjacent residential buildings, resulting in concerns about noise impact on the residences. During the summer of 2021, improvements were made to the stage area to reduce these noise impacts. We will review the changes made to the construction of the stage area, discuss sound level measurements made both before and after the improvements, and comment on the overall acoustical results of the improvements.

10:25–10:40 Break

10:40

1aAA5. Noise abatement study for an amplified music outdoor amphitheater. David s. Woolworth (Roland, Woolworth & Assoc., 356 County Rd. 102, Oxford, MS 38655-8604, dwoolworth@rwaconsultants.net)

Amplified music outdoors poses many challenges in regard to sound abatement and requires a multi-disciplinary approach that includes sound system design and monitoring, programming considerations, architectural and landscaping sound control elements, and conditions for outdoor sound propagation including atmospheric, ground impedance, barrier and topographic effects. An interdisciplinary study was performed on the Brushy Creek Amphitheater in Hutto, Texas involving a local sound reinforcement company, audio equipment manufacturer, a local acoustical consulting firm, and a national atmospheric research group. This paper will provide the elements and findings of a multi-disciplinary investigation to minimize impact of a newly located amphitheater on the local community.

Contributed Papers

11:00

1aAA6. Real-time atmospheric effects on sound propagation for a simulated outdoor concert. David s. Woolworth (Roland, Woolworth & Assoc., 356 County Rd. 102, Oxford, MS 38655-8604, dwoolworth@rwaconsultants.net)

Outdoor live sound is best controlled at the sound source; computer simulation tools are available to estimate impact of concerts on nearby communities using methods such as NORD 2000, ISO 9613-2, and CNOSSOS. These methods utilize generalized temperature gradients and wind speed profiles up to 100 m and can be useful and somewhat accurate up to 3 miles, after which higher level atmospheric effects takeover. This paper will provide some examples of long rang propagation due to higher level atmospheric effects, compare 3 mile radius modeling to real-time measurements and examine some of the fluctuations over a 2-h simulated concert with changing weather conditions and programming.

11:15

1aAA7. Noise control of a seasonal outdoor ice rink. John Baldassano (Ostergaard Acoust. Assoc., 1480 US 9, Woodbridge, NJ 07095, jbaldassano@acousticalconsultant.com) and Joseph Keefe (Ostergaard Acoust. Assoc., Woodbridge, NJ)

This presentation discusses our involvement with community noise complaints, measurements, and noise control related to an outdoor ice rink. This seasonal rink operates November—March each winter season and is erected over municipal basketball courts. Residences in multiple municipalities border the rink and are potentially affected by rink noise, including amplified music, Zamboni operation, whistles, hockey sticks on ice, and hockey pucks impacting boards. Pre- and post-noise control data are presented that show the effect of temporary measures applied in this challenging environment.

Session 1aAB

Animal Bioacoustics, Acoustical Oceanography, Underwater Acoustics, and Signal Processing in Acoustics: Open-Source and Free Tools for Bioacoustics

Xavier Mouy, Cochair
Passive Acoustics Research Group, NOAA, 3377 SW 28th Terrace, Miami, FL 33133

Julie Oswald, Cochair
University of St Andrews, Scottish Oceans Institute, East Sands, St Andrews KY16 8LB, United Kingdom

Chair's Introduction—8:20

Invited Papers

8:25

1aAB1. Identifying and building on the current state of bioacoustics software. Tessa Rhinehart (Biological Sci., Univ. of Pittsburgh, Clapp Hall, Fifth and Ruskin Aves, Pittsburgh, PA 15260, tessa.rhinehart@pitt.edu), Samuel Lapp, and Justin Kitzes (Biological Sci., Univ. of Pittsburgh, Pittsburgh, PA)

Bioacoustics is a powerful and increasingly commonly used tool for terrestrial and marine biological assessments. As the scale of bioacoustic data collection has increased, techniques for processing these data have diversified. However, with analysis methods rapidly evolving and dozens of analysis software packages already available, it is challenging to identify which software, if any, meets a particular researcher's needs. We reviewed bioacoustics software to identify packages aimed at or used by bioacoustics researchers in ecology. We compiled descriptions of the function of 65 stable or actively developed software packages used for bioacoustics analyses. Of these, 59 were free or open-source packages. In addition, we developed free, open-source Python software, OpenSoundscape, that addresses gaps in available software. OpenSoundscape simplifies the process of creating flexible, scalable deep learning algorithms for bioacoustic analysis. It can be used to train binary or multiclass convolutional neural networks with any PyTorch-implemented model structure (e.g., ResNet50, Inception v3). Researchers can easily customize its spectrogram preprocessing and data augmentation routines to improve model performance. OpenSoundscape also includes modules to work with annotated acoustic data, apply additional signal processing algorithms, perform acoustic localization, and "open the black box" of deep learning using Grad-CAM.

8:45

1aAB2. MERIDIAN open-source software for deep learning-based acoustic data analysis. Oliver S. Kirsebom (Comput. Sci., Dalhousie Univ., 6050 University Ave. Halifax, NS B3H 4R2, Canada, oliver.kirsebom@dal.ca), Fabio Frazao, Bruno Padovese, Sadman Sakib, Yue Su, and Stan Matwin (Comput. Sci., Dalhousie Univ., Halifax, NS, Canada)

Deep neural networks have the potential to transform our approach to developing acoustic detection and classification models, enabling acousticians to develop or re-purpose such models through a fully data-driven approach requiring minimal knowledge of signal processing, algorithm design, and programming. However, open-source software to facilitate this data-driven workflow is currently lacking. MERIDIAN is working towards filling this gap through the development of several open-source software products, including the Python package Ketos and the MAIPL (Marine AI PLatform) suite of web applications. While Ketos provides a high-level programming interface for training deep neural networks at detecting and classifying sounds, MAIPL is a modular cloud computing service that supports the full model-development workflow. In this contribution, an overview of Ketos and MAIPL will be given and their functionalities will be demonstrated through their application to the HALLO (Humans and ALgorithms Listening for Orcas) project. We highlight one of the MAIPL tools, the MAIPL-Annotator, which provides a user-friendly interface for collaboratively annotating sound samples and validating model predictions. Future developments will also be described, highlighting new MAIPL applications under development such as the MAIPL-Adapter, a tool for adapting acoustic deep learning models to new acoustic environments.

9:05

1aAB3. PAMGuard: Open-source detection, classification, and Localization software. Jamie D. Macaulay (Sea Mammal Res. Unit, Univ. of St Andrews, Sea Mammal Res. Unit, St Andrews, Fife KY16 8LB, United Kingdom, jdjm@st-andrews.ac.uk) and Douglas Gillespie (Sea Mammal Res. Unit, Univ. of St Andrews, St Andrews, Scotland, United Kingdom)

The PAMGuard open-source software project was established in 2006 to provide a toolbox and app for the detection, classification, and localization (DCL) of bioacoustic signals. Releases typically attract between 1500 and 3500 downloads. Primarily developed for marine mammals, users can configure PAMGuard to work with a wide variety of hardware configurations (single hydrophones, to complex 3D arrays) and DCL algorithms, depending on the species of interest. A Java code base and integrated multi-threading allow multiple

DCL algorithms to be run in parallel enabling rapid analysis of large acoustic datasets. A comprehensive data management system and a suite of interactive displays then allow users to visualize and annotate resulting data at temporal scales from milliseconds to years. This combination of advanced analysis algorithms and powerful data visualization in both real-time, or reviewing data post-hoc, means PAMGuard provides both automatic and a proven and reliable human-in-the-loop approach to analysis. Developers can use the program's infrastructure to rapidly develop and deploy new algorithms; recently, an international collaboration has improved support for terrestrial species and developed a new module to import acoustic deep learning models, enabling users to take advantage of PAMGuard's data management and visualization capabilities alongside the latest automated acoustic classifiers.

9:25

1aAB4. Development of deep neural networks for marine mammal call detection using an open-source, user friendly tool. Elizabeth L. Ferguson (Ocean Sci. Analytics, San Diego, CA, eferguson@oceanscienceanalytics.com), Peter Sugarman (Humans & Dolphins Talking, LLC, Bellevue, WA), Kevin R. Coffey (Univ. of Washington School of Medicine, Seattle, WA), Jennifer Pettis Schallert (BioSci, LLC, Satellite Beach, FL), and Gabriela C. Alongi (Ocean Sci. Analytics, San Diego, CA)

As the collection of large acoustic datasets used to monitor marine mammals increases, so too does the need for expedited and reliable detection of accurately classified bioacoustic signals. Deep learning methods of detection and classification are increasingly proposed as a means of addressing this processing need. These image recognition and classification methods include the use of a neural networks that independently determine important features of bioacoustic signals from spectrograms. Recent marine mammal call detection studies report consistent performance even when used with datasets that were not included in the network training. We present here the use of DeepSqueak, a novel open-source tool originally developed to detect and classify ultrasonic vocalizations from rodents in a low-noise, laboratory setting. We have trained networks in DeepSqueak to detect marine mammal vocalizations in comparatively noisy, natural acoustic environments. DeepSqueak utilizes a regional convolutional neural network architecture within an intuitive graphical user interface that provides automated detection results independent of acoustician expertise. Using passive acoustic data from two hydrophones on the Ocean Observatories Initiative's Coastal Endurance Array, we developed networks for humpback whales, delphinids, and fin whales. We report performance and limitations for use of this detection method for each species.

9:45

1aAB5. INSTINCT: The infrastructure for noise and soundscape tolerant investigation of nonspecific call types. Daniel F. Woodrich (UW Cooperative Inst. for Climate, Ocean, & Ecosystem Studies, Univ. of Washington, 7600 Sand Point Way, Seattle, WA 98115, daniel.woodrich@noaa.gov)

INSTINCT is open-source, command line software for custom data pipelines, developed by the NOAA's Alaska Fisheries Science Center to formalize AI workflows for passive acoustic monitoring (PAM) of sounds produced by marine mammals and to provide a framework for new algorithm development. INSTINCT is built in Python over the Luigi framework and was designed to be lightweight and Window/Linux compatible. To promote collaboration, the software is partitioned into a core module and organization-governed submodules. The original INSTINCT detection and classification algorithm has been applied to various Alaska region cetacean species and presents a powerful CPU-based approach for those seeking to train, evaluate, and deploy a generalized detector. Furthermore, INSTINCT pipelines are in development to train and deploy GPU-based algorithms, structure annotation workflows, and pioneer complex workflows. Case studies will be presented to highlight the existing capabilities of the software and new workflows that are currently being explored. This talk will also cover design philosophy, technical specifications, and instructions for implementation. Although INSTINCT continues to progress at AFSC, it will benefit greatly from a larger network of collaborators, and ideally, will give back to the community via innovation of new AI approaches in PAM.

10:05–10:20 Break

10:20

1aAB6. Combining neural network and sequence analyses to determine animal call repertoires. Vincent M. Janik (Dept. of Biology, Univ. of St Andrews, St Andrews, Fife, United Kingdom, vj@st-andrews.ac.uk) and Julie Oswald (Dept. of Biology, Univ. of St Andrews, St Andrews, Fife, United Kingdom)

Classifying frequency modulation patterns in animal signals is a challenging task depending on the level of stereotypy of call types. Not only do graded transitions between types create problems for defining them, but animals also vary single parameters within types without changing the overall shape of a modulation. One example for the latter is the variation in overall duration within call types of delphinids. Deecke *et al*. (1999) developed an adaptive resonance theory neural network for killer whale calls to standardize classification for dialect comparisons. This was later modified to make it usable for delphinid whistles by Deecke and Janik (2006) and is available as free software under the name ARTWARP. It creates categories by comparing frequency contours while time warping them to allow for variation in overall duration. Average frequency contours of all resulting call types are provided to describe repertoires once classification is complete. ARTWARP has recently been further improved to expand its analysis outputs and speed and can be used for the classification of any animal signals with a tonal component. Tests of ARTWARP in combination with the SIGID sequence analysis for signature whistles (Janik *et al*., 2013) provides an accurate repertoire of delphinid calls.

10:40

1aAB7. Parselmouth for bioacoustics: Integrating Praat into the Python scientific ecosystem. Yannick Jadoul (Comparative Bioacoustics, Max Planck Inst. for Psycholinguistics, Wundtlaan 1, Nijmegen 6525 XD, Netherlands, Yannick.Jadoul@mpi.nl), Diandra Duengen, and Andrea Ravignani (Comparative Bioacoustics, Max Planck Inst. for Psycholinguistics, Nijmegen, Netherlands)

As collected datasets become larger and computational analyses become ever more complex, the efficient processing of bioacoustical data is a crucial problem to tackle. Often, during data exploration and analysis, different research software packages need to be flexibly combined in a script. A typical example of such a multi-faceted workflow is the extraction of acoustic parameters from a recording, which are then plotted and tested for statistical significance. Parselmouth is an open-source Python library for Praat, a widely used acoustics and phonetics software package implementing acoustic algorithms and analyses regularly adopted in bioacoustics research. Parselmouth's goal is to provide a full-fledged Python library that integrates efficiently into the larger Python ecosystem. This way, it not only simplifies the application of Praat's functionality within a typical data analysis workflow but also enables the creation of new experimental tools. Parselmouth's contribution to bioacoustics research can be highlighted through concrete examples of studies we have conducted, e.g., on vocal flexibility in seals. Moreover, the integration of Praat's functionality into a general-purpose programming language allows for novel, more complex experimental set-ups: for example, the integration of Parselmouth into a custom-created software tool permits live-monitoring and instantly evaluating the vocal development during animal training.

10:55

1aAB8. Parselmouth for bioacoustics: Analysis pipelines for seal vocalizations. Andrea Ravignani (Comparative Bioacoustics, Max Planck Inst. for Psycholinguistics, Wundtlaan 1, Nijmegen 6525 XD, Netherlands, Andrea.Ravignani@mpi.nl), Laura Torres Borda (Comparative Bioacoustics, Max Planck Inst. for Psycholinguistics, Nijmegen, Netherlands), Heikki Rasilo (Artificial Intelligence Lab, Vrije Universiteit Brussel, Elsene/Ixelles, Belgium), Anna Salazar Casals (Res. Dept., Sealcentre Pieterburen, Pieterburen, Netherlands), and Yannick Jadoul (Comparative Bioacoustics, Max Planck Inst. for Psycholinguistics, Nijmegen, Netherlands)

Every empirical research project includes bottlenecks at various levels. In bioacoustics, one of these time-consuming bottlenecks corresponds to the step of transforming a long stream of audio into acoustic properties of specific sounds. Here, we describe a data-extraction pipeline which integrates manual annotation with Parselmouth's powerful computational analyses. This semi-supervised method allows extracting a large volume of sound features with limited repetitive human "point and click." We illustrate this using recently published empirical research, where we focused on vocal production learning and plasticity in pinnipeds. Faced with a species capable of imitating sounds, fully automatic methods may misclassify individuals (because of imitation), while the large number of calls make fully manual approaches suboptimal and error-prone. Focusing on early vocal development, we tested 1–3 weeks-old harbor seal pups (*Phoca vitulina*). Noise playbacks served to induce seal pups to shift their fundamental frequency. Pups' spontaneous calls were recorded while exposed to bandpass-filtered noise, which spanned and masked the animals' fundamental frequency range. After a summary manual annotation of calls' boundaries, Parselmouth identified these boundaries in the files, and automatically extracted multiple sound parameters. Based on this, we found that pups modified their vocalizations by lowering their fundamental frequency in response to noise.

11:10

1aAB9. Automated detection of blue whale D-calls using deep learning with a double-observer performance assessment. Shyam Madhusudhana (K. Lisa Yang Ctr. for Conservation Bioacoustics, Cornell Univ., 159 Sapsucker Woods Rd., Ithaca, NY 14850, shyamm@cornell.edu), Brian S. Miller (Australian Antarctic Div., Kingston, Tasmania, Australia), Meghan G. Aulich (Ctr. for Marine Sci. & Technol., Curtin Univ., Perth, Western Australia, Australia), and Nat Kelly (Australian Antarctic Div., Kingston, Tasmania, Australia)

An automated algorithm for passive acoustic detection of blue whale D-calls is developed based on established deep learning methods for image recognition via the DenseNet architecture. Koogu—an open-source Python package—was used for developing the detector. The detector was trained on annotated acoustic recordings from the Antarctic, and the performance of the detector was assessed by calculating precision and recall using a separate independent dataset also from the Antarctic. Detections from both the human analyst and automated detector were then inspected by a more experienced analyst to identify any calls missed by either approach and to adjudicate whether the apparent false-positive detections from the automated approach were actually true-positives. Lastly, an additional performance assessment was conducted using double-platform methods (via a closed-population Huggins mark recapture model) to assess the probability of detection of both the human analyst and automated detector, based on the assumption of false-positive-free and reconciled detections. According to our double-platform analysis, the automated detector performed very well with higher recall and fewer false-positives that the original human analyst.

11:25

1aAB10. Using passive acoustic monitoring and machine learning analysis to investigate katydid ecology and behavior. Laurel Symes (K. Lisa Yang Ctr. for Conservation Bioacoustics, Cornell Univ., 159 Sapsucker Woods Rd., Ithaca, NY 14850, symes@cornell.edu), Hannah M. ter Hofstede (Dartmouth College, Hanover, NH), Sharon J. Martinson (K. Lisa Yang Ctr. for Conservation Bioacoustics, Cornell Univ., Ithaca, NY), Inga Geipel (Smithsonian Tropical Res. Inst., Gamboa, Panama), Ciara E. Kernan (Dartmouth College, Hanover, NH), and Shyam Madhusudhana (K. Lisa Yang Ctr. for Conservation Bioacoustics, Cornell Univ., Ithaca, NY)

Passive acoustic monitoring (PAM) can provide detailed information on the spatial and temporal distribution of sound producing insects. When combined with machine learning approaches for extracting data from multiple sites and multiple years, PAM can provide exceptionally detailed information about the ecology of the calling insect community. We placed recording devices in the forest canopy on Barro Colorado Island in Panamá and used a combination of manual annotation and machine learning analysis in Koogu (an open source python package) to test the following hypotheses in Neotropical forest katydids: (1) The forest canopy species assemblage will consist disproportionately of katydid species with high flight and dispersal ability (reflected by low wing-loading coefficients), (2) katydids aggregate on an individual tree during the short window when a tree flushes new leaves (resulting in short concordant peaks of signaling activity across multiple katydid species), and (3) in species with relatively short calling seasons, males will have little time to accumulate nuptial gifts for females and will instead invest in mate searching (reflected by high male:female sex ratios in insects captured at lights). In changing forests, consistent approaches for insect sampling will be key for understanding insect ecology and generating interpretable and actionable data.

11:40–12:00
Panel Discussion

Session 1aBA

Biomedical Acoustics: General Topics in Biomedical Acoustics: Bubbles and Cavitation

Kausik Sarkar, Chair
Mechanical and Aerospace Eng, George Washington University, 800 22nd NW, Washington, DC 20052

Contributed Papers

8:45

1aBA1. Spatiotemporal decomposition methods for nanobubble contrast-enhanced ultrasound. Dana Wegierak (Biomedical Eng., Case Western Reserve Univ., 2626 N Moreland Blvd, Apt. 16, Cleveland, OH 44120, dxw477@case.edu), Reshani Perera (Dept. of Radiology, Case Western Reserve Univ., Cleveland, OH), Michaela Cooley (Biomedical Eng., Case Western Reserve Univ., Cleveland, OH), Agata A. Exner (Dept. of Radiology, Case Western Reserve Univ., Cleveland, OH), and Mahdi Bayat (Case Western Reserve Univ., Cleveland, OH)

Nanobubble ultrasound contrast agents (NBs) offer enhanced access to deep tumor tissue by extrapolation from vasculature. When combined with targeting moieties, shell-stabilized nanobubbles can outperform other agents used in molecular ultrasound imaging due to their small size. The visualization and decoupling of extravasated and intravascular NBs is critical to furthering nanobubble molecular imaging with ultrasound. Spatiotemporal processing of echo data, usually via singular value decomposition (SVD), enables enhanced decoupling of signals of stationary from moving scatterers and therefore provides a novel tool for analysis of NB extravasation into the parenchyma. Instead of trivial solutions, we have developed non-negative SVD (nnSVD) methods applicable to most commercial scanners without requiring the raw data. Additionally, our formulation allows addition of proper regularizers, such as sparsity, to further expand the decomposition dimensions. We validate these methods in phantom models and present how the suggested decompositions can distinctly decouple flowing from extravasated NBs in *in vivo* tumour models with enhanced permeability. Our results show the successful tracking of nanobubble accumulation in tumour tissue and can serve as a relative metric for molecular targeting of prostate cancer. Further exploration of these methods could be used to improve long-term imaging and drug delivery capabilities of NBs.

9:00

1aBA2. Sensitivity of the subharmonic responses from contrast microbubbles to ambient pressure. Roozbeh Hassanzadeh Azami (Mech. and Aerosp. Eng., The George Washington Univ., 800 22nd St. NW, Ste. 3000, Washington, DC 20052, roozbehazami@gwu.edu), Mehmet Yapar, and Kausik Sarkar (Mech. and Aerosp. Eng., The George Washington Univ., Washington, DC)

Microbubbles are well-known ultrasound contrast agents which can produce nonlinear oscillation leading to harmonic and subharmonic responses. There has been an interest in utilizing this phenomenon for noninvasive pressure estimation and its use in diagnosis of diseases such as portal hypertension. The subharmonic response of a microbubble display three different regimes with increasing excitation: occurrence, growth, and saturation. In this study, subharmonic responses of lipid-shelled microbubbles were investigated with varying ambient pressures at different excitation parameters. We observed that subharmonic sensitivity to ambient pressure varied differently in different excitation ranges. We also noted that applying overpressure could induce subharmonic generation where there was no subharmonic without overpressure. At occurrence stage, the sensitivity of subharmonic to ambient pressure was significantly higher. This property can be utilized for improved pressure estimation with high sensitivity even at low acoustic excitation.

9:15

1aBA3. Predicting acoustic emissions of ultrasound-driven lipid-coated microbubbles. Sören Schenke, Rishav Saha (Otto-von-Guericke-Univ. Magdeburg, Magdeburg, Germany), and Fabian Denner (Otto-von-Guericke-Univ. Magdeburg, Universitätsplatz 2, Magdeburg 39106, Germany, fabian.denner@ovgu.de)

Lipid-coated microbubbles excited by ultrasound are utilized in an increasing number of diagnostic and therapeutic medical applications, in which the acoustic waves emitted by the oscillating or collapsing bubbles are the main protagonist. The emitted acoustic waves cause bioeffects, enable contrast enhancement for ultrasonic imaging, and serve as a reference to monitor cavitation activity *in situ*. However, quantifying the acoustic waves emitted by cavitation bubbles turns out to be difficult: the small and extremely transient phenomenon is challenging to measure in experiments with sufficient accuracy and state-of-the-art computational methods are limited in their ability to predict acoustic emissions reliably. In this contribution, we present our recent work on predicting acoustic emissions of microbubbles based on Rayleigh–Plesset models and, more generally, reduced-order models. Using the nonlinear Westervelt equation including the motion of the bubble wall and the background medium as a result of the bubble oscillations, we investigate the amplitude and frequency modulation (e.g., due to nonlinear Doppler effects, thermoviscous attenuation or geometric distortion) of acoustic waves emitted by lipid-coated microbubbles. Additionally, we explore the capabilities of a reduced-order modeling approach that drastically simplifies the computational complexity of simulating these acoustic emissions.

9:30

1aBA4. Boundary element simulation of a collapsing coated microbubble near a plane. Nima Mobadersany (Mech. and Aerosp. Eng, George Washington Univ., Washington, DC) and Kausik Sarkar (Mech. and Aerosp. Eng, George Washington Univ., 800 22nd NW, Washington, DC 20052, sarkar@gwu.edu)

Ultrasound contrast agents are gas core micron size bubbles coated with a layer of lipids/proteins to stabilize them against premature dissolution in the bloodstream. In addition to enhancing the contrast of the image, contrast agents have been implicated in numerous harmful and beneficial bioeffects. In the present study, ultrasound excited collapse of a coated microbubble near a plane has been studied using an axisymmetric Boundary Element method. The coating of the microbubble is modeled as a viscoelastic interface using an in-house developed strain-softening model (exponential elasticity model). The influence of the shell model on the stability of the numerical simulation during the microbubble jet formation has been investigated. The numerical approach was compared and validated with earlier studies. The effects of ultrasound excitation parameters and mechanical properties of the coating, i.e., shell viscosity and elasticity, on the bubble behaviors and the velocity and pressure in the surrounding fluid have been studied.

9:45

1aBA5. Calibration of a focused passive cavitation detector using bubble shock waves. Krit Sujarittam (Imperial College London, London, United Kingdom) and James Choi (Imperial College London, London, United Kingdom, j.choi@imperial.ac.uk)

In microbubble-mediated therapeutic ultrasound, a focused passive cavitation detector (PCD) is often used to measure the bubbles' acoustic emissions, providing useful signals for treatment monitoring. However, calibrating a spherically focused PCD is challenging, due to the difficulty of generating a spherical wave that matches the PCD's surface curvature. Here, a PCD was calibrated using broadband shock waves generated by inertial collapses of single microbubbles. Microbubbles were diluted to a very low concentration, flowed through a wall-less gel channel, and sonicated using single-cycle, 0.5-MHz-centre-frequency, 1-MPa acoustic pulses. The focused PCD to be calibrated and a reference needle hydrophone captured their emissions. The sensitivity and phase response of the PCD relative to the reference hydrophone was calculated from the single bubble signals. For comparison, the PCD was also calibrated using a focused emitter as a sound source (Rich and Mast, *JASA*, 2015). The nominal PCD sensitivities obtained using the two methods agreed within $1\% \pm 14\%$ within the PCD's bandwidth (2–10 MHz). The calibration data from the bubble method was then used to correct the PCD's signal distortions. Our method recovered the impulse waveform of the bubble-generated shock wave from the raw PCD signal, where such a waveform was not previously observed.

10:00

1aBA6. Self-organization of human stem cells into spheroids in a multinode acoustic levitation. Mauricio Hoyos (PMMH, CNRS, ESPCI, 10 rue Vauquelin, Paris 75005, France, hoyos@pmmh.espci.fr)

We present an approach based on acoustic levitation to grow the cells without wall interactions in order to promote 3D cell architecture (spheroïd, organoïd). The 3D microenvironment is indeed closer to *In Vivo* physiological behavior. The suspended cells are moved toward the acoustic pressure nodes where they are trapped and maintained in acoustic levitation in perfectly straight monolayers reaching very quickly the classical confluency of the cells in 2D culture. Interestingly, by maintaining the monolayers of MSCs in culture over a 24-h period, the MSCs spontaneously self-organized from cell sheets to cell spheroids with a characteristic time of about 10 h. This approach of 3D cell culture is based on the use of the acoustic wave coupled with microfluidics. We designed a standing wave cavity to generate a large acoustic radiation force (ARF) with an optical access, which allows the characterization of the self-organization dynamics. This 3D cell culture method has been validated on MSCs over 24h experiments. The MSCs viability has been checked. Moreover, they show a higher differentiation capacity compared to standard 2D culture conditions. These results open the path to long-time cell culture in acoustic levitation of cell sheets or spheroids for any type of cells.

10:15–10:30 Break

10:30

1aBA7. The effect of stiffness and impurities on bubble nucleation in polyacrylamide hydrogels. Ferdousi Sabera Rawnaque (Graduate Program in Acoust., Penn State Univ., 210 East Hamilton Ave., Apt. 31, State College, PA 16801, fmr5186@psu.edu) and Julianna C. Simon (Graduate Program in Acoust., Penn State Univ., University Park, PA)

Tissue mimicking hydrogels can help us understand how viscoelastic properties (stiffness, elastic modulus, etc.) of biological tissues impact bubble nucleation. Adding impurities to hydrogels introduces inhomogeneities and increases their similarity to biological tissues. In this study, we evaluated the effect of stiffness and impurities on bubble nucleation in polyacrylamide (PA) hydrogels. Bubble nuclei were evaluated in 17.5%, 20%, and 22.5% v/v PA hydrogels, after which 0.25% w/v cholesterol crystals (maximum dimension = 0.6 mm) were embedded in the gels (n = 3 each). A 1.5 MHz focused ultrasound transducer was used to induce cavitation using 10-ms pulses with pressures ranging up to $p_+ = 89$ MPa, and $p- = 26$ MPa and -6 dB focal dimensions of 9.4×1.2 mm (p−). Image analysis from high-speed photography showed bubble nucleation increases with increasing peak negative pressure and decreasing hydrogel stiffness. Adding cholesterol crystals largely decreases the acoustic cavitation threshold from $p- = 19$ MPa for 17.5% v/v hydrogels with no added impurities to $p- = 12$ MPa for the same concentration hydrogel with added cholesterol crystals. This suggests that hydrophobic cholesterol crystals weaken the gel or trap bubble nuclei, thus lowering the cavitation threshold. Future work includes investigating bubble nuclei in rat hepatocytes. [Work supported by NSF CAREER 1943937 and PSU Riess Fellowship.]

10:45

1aBA8. Histotripsy bubble dynamics in tendon and anisotropic gel phantoms. Jacob C. Elliott (Graduate Program in Acoust., The Penn State Univ., Res. West, State College, PA 16801, jce29@psu.edu), Andrea Arguelles (Eng. Sci. and Mech., The Penn State Univ., State College, PA), and Julianna Simon (Graduate Program in Acoust., The Penn State Univ., University Park, PA)

Collagenous, anisotropic tissues such as tendon have demonstrated resistance to liquefaction by histotripsy, despite the creation, oscillation, and collapse of bubbles verified using B-mode imaging. The objective of this work is to evaluate effects of anisotropy on bubble dynamics in tissue-mimicking hydrogels and compare to anisotropic tissues. Polyacrylamide, fibrin, and collagen hydrogels were fabricated; *ex vivo* bovine tendons were obtained. Sound speeds were measured in each axial direction to evaluate degree of anisotropy. Hydrogels and tendons were exposed to 1.5-MHz focused ultrasound with 10-ms pulses repeated at 1-Hz with $p+ = 89$ MPa, $p- = 26$ MPa. Cavitation activity was monitored with simultaneous high-speed photography and passive cavitation imaging using a Philips/ATL L7-4 transducer and Vantage® ultrasound system. Violent cavitation activity and fractionation was observed in polyacrylamide, collagen, and fibrin hydrogels with low degrees of anisotropy (<1.2); such behavior is unlike that of tendon. Dehydration of fibrin gels resulted in a 55% reduction in peak cavitation emission energy and a 260% increase in anisotropy compared to standard fibrin formulations. These gels demonstrated similar cavitation energy than tendon (within 4%) but 50% less anisotropy, indicating more hydrogel formulations should be explored to better mimic collagenous, anisotropic tissue. [Work supported by NIH R21EB027886.]

11:00

1aBA9. The effect of surface tension on the color Doppler ultrasound twinkling artifact. Eric Rokni (Graduate Program in Acoust., Penn State Univ., The Penn State Univ., State College, PA 16801, ezr144@psu.edu) and Julianna C. Simon (Graduate Program in Acoust., Penn State Univ., University Park, PA)

Imaging crystalline structures with Doppler ultrasound can produce a rapid color shift, termed the twinkling artifact, that can assist in diagnosing pathological mineralizations such as kidney stones, heterotopic ossification, gout, and breast microcalcifications. Twinkling is theorized to arise from scattering off surface crevice microbubbles, which are affected by the surface tension between the bubble and surrounding medium. In this study, we evaluated the effect of surface tension on twinkling in pure crystals. Cholesterol, calcium phosphate, and uric acid crystals were grown *in vitro* (n = 5 each) and imaged with a Philips/ATL L7-4 transducer and Vantage-128 research ultrasound system. Crystals were imaged in water while varying surfactant concentration (0%–4%) leading to surface tensions that ranged from 45–72 mN/m. Surface tension of the solution was determined by measuring the contact angle of 0.1-mL droplet on an acrylic sheet exposed to air. As the concentration of surfactant increased, twinkling was found to decrease by ∼20% for cholesterol and calcium phosphate and ∼10% for uric acid. These results continue to support the crevice bubble theory of twinkling and suggest the importance of surface tension when evaluating minerals with the twinkling artifact. [Work supported by NSF-CAREER-1943937 and NSF-GRFP-DGE1255832.]

11:15

1aBA10. Investigating the color Doppler twinkling artifact as an early detector of cellular mineralization. Lucas Ruge-Jones (Graduate Program in Acoust., Penn State Univ., University Park, PA), Lisa Berntsen (Penn State, University Park, PA), Fea Morgan-Curtis (Graduate Program in Acoust., Penn State Univ., 201E Appl. Sci. Bldg., University Park, PA 16802, fkm5174@psu.edu), Daniel Hayes (Penn State, University Park, PA), and Julianna C. Simon (Graduate Program in Acoust., Penn State Univ., University Park, PA)

Heterotopic ossification (HO) occurs when bone develops in areas where bone does not usually exist, often appearing after blast injuries or other musculoskeletal trauma. Current methods of diagnosis, such as three-phase bone scintigraphy, x-ray, or CT scans, take several weeks after the trauma before HO is visible. The color Doppler ultrasound twinkling artifact has been shown to detect the presence of kidney stones and other crystals. Twinkling could provide a more sensitive method of HO detection, which would allow for earlier diagnosis and treatment and limit HO severity. In our research, human bone marrow-derived stem cells (HBMSCs) were plated onto collagen scaffolds and cultured in osteogenic media to promote mineralization or growth media (controls). The cells were imaged every 3rd day from day 10 – day 25 using Phillips L7-4, L12-5, and Vermon L22-14v transducers and a Vantage-128 research ultrasound system; I/Q data were bulk saved for quantitative analysis. Samples with mineralization on 0.5% of their surface showed a 17% increase in Doppler magnitude compared to controls, suggesting that the twinkling artifact is a sensitive method for detection of mineralization *in vitro*. Further studies will investigate whether this method is effective in a murine model. [Work supported by DoD CDMRP PR201164.]

11:30

1aBA11. Passive cavitation detection with a needle hydrophone array. Zheng Jiang (Dept. of Bioengineering, Imperial College London, Royal School of Mines, South Kensington Campus London, London SW7 2AZ, zheng.jiang18@imperial.ac.uk), Krit Sujarittam, Betul Ilbilgi Yildiz, Robert J. Dickinson, and James Choi (Dept. of Bioengineering, Imperial College London, London, United Kingdom)

Therapeutic ultrasound and microbubble technologies seek to drive systemically administered microbubbles into oscillations that safely manipulate tissue or release drugs. Such procedures often listen to and then utilize microbubble emissions to control the microbubble activity. However, most sensors reported introduce large distortions to the acoustic signal. Acoustic shockwaves, a key emission from microbubbles, are largely absent in reported recordings. Here, we present a needle hydrophone array better suited for monitoring ultrasound-driven microbubble activities. The array consisted of eight polyvinylidene fluoride (PVDF, diameter: 2 mm) needle hydrophones. Each needle had a flexible coaxial wire at its end allowing flexibility to fit into a 3D-printed scaffold. Using this array, we monitored microbubbles exposed to ultrasound pulses (center frequency: 0.5 MHz, peak-rarefactional pressure: 130–-597 kPa, pulse length: 4 cycles). Our tests revealed that each needle had a broadband frequency response (1–15 MHz) and was able to capture shock waveforms generated by bubbles. The signal-to-noise ratio of the array was approximately 2 times higher than individual hydrophones. Also, the array could localize microbubble activities and determine the cavitation threshold. Thus, the array accurately monitored and localized microbubble activities, and may be an important technological step towards safer and more effective treatments.

11:45

1aBA12. The use of real-time passive cavitation imaging for monitoring boiling histotripsy treatments. Gilles P. Thomas (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, gillespierre.thomas@gmail.com), Alex T. Peek (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Tatiana D. Khokhlova (Dept. of Medicine, Univ. of Washington, Seattle, WA), and Vera A. Khokhlova (Univ. of Washington/Moscow State Univ., Moscow, Russian Federation)

Boiling histotripsy (BH) is a high intensity focused ultrasound method that induces mechanical tissue liquefaction and relies on the formation of a mm-sized vapor bubble at the focus. One challenge of transcutaneous BH is to confirm the presence of such bubble and to differentiate it from prefocal cavitation. Strong prefocal cavitation can shield the focus, prevent initiation of boiling, and lead to unsuccessful treatment, although an echogenic region would still be present on B-mode ultrasound images. Here, a methodology for real-time monitoring of different bubble activity during BH treatment is proposed. Custom BH systems of 1 and 1.5 MHz with inline ultrasound imaging probes connected to a Verasonics V1 system were used for BH treatments in pigs *in vivo* and in *ex vivo* tissues. Backscattered signals arriving to the imaging probe were recorded during the 10 ms BH pulses delivered at 1 Hz and passive cavitation imaging (PCI) reconstruction was performed and displayed on top of the B-mode images to identify and classify the regions of bubble activity. Spectral analysis of the PCI signals promoting prefocal cavitation and lack thereof was performed to identify the spectral features corresponding to prefocal cavitation and vapor bubble formation. [Work supported by NIH R01EB007643, R01GM122859, and R01EB25187.]

Session 1aNS

Noise, Architectural Acoustics, and ASA Committee on Standards: Standards, Codes, and Criteria Applications in the Real World I

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

Derrick P. Knight, Cochair
Trane Technologies, 2313 20th Street South, La Crosse, WI 54601

Chair's Introduction—9:00

Invited Papers

9:05

1aNS1. "Acoustics" as a "system"—A refreshed narrative leading change in standards, codes, and guidelines. Viken Koukounian (K.R. Moeller Assoc. Ltd., 3-1050 Pachino Court, Burlington, ON L7L 6B9, Canada, viken@logison.com)

Although "acoustics" is widely understood to be in relation to the person, it is—more correctly—a branch of physics that studies the behavior of mechanical energy in media. In contrast, "psychoacoustics" concerns itself with the perception (or psychology) of sound and the associated physiological effects. The disassociation is best exemplified by traditional architectural acoustical design schemes—i.e., "acceptable-level" and "categorization"—which are prevalent throughout existing Standards, Codes, and Guidelines. The result, as evidenced by the meta-analysis of modern and sustainable best practices by a group at Harvard, is an Indoor Environment Quality parameter that does not consistently score better, and in several studies, results in lower satisfaction scores with noise. This presentation presents posteriori—correlations between "occupant satisfaction" and "acoustical satisfaction," "acoustical satisfaction" and "acoustical privacy," and "acoustical privacy" and "health and well-being"—to justify the pitfalls of traditional strategies, and to develop a priori—a framework for "good acoustics." This refreshed narrative, to consider acoustics as a system, leads change among the most popular sustainability and well-being Standards, such as WELL, Green Globes, and LEED. This presentation endeavors to summarize notable updates and to identify new risks and challenges associated with recent versions of the aforementioned documents.

9:25

1aNS2. Challenges in the application of sound standards in the heating, ventilation, and air conditioning. Derrick P. Knight (Trane Technologies, 2313 20th St. South, La Crosse, WI 54601, Derrick.Knight@TraneTechnologies.com)

The HVAC industry in North America self-manages sound data measurement and reporting through the Air Conditioning Heating and Refrigeration Institute (AHRI). Each product is intended to fall under a specific AHRI sound standard. But in the real world, many complications exist. Some products may be configured in ways which crossover scope boundaries for multiple standards such as fan coils due to ducting or concealing. Other products are being held to inapplicable standards by specification requirements such as air-handlers using large fan arrays. Misunderstandings around application standards such as AHRI Standard 885 also generate confusion, especially when using Appendix E as a design tool. This presentation will highlight the challenges observed when applying AHRI Standards to building design.

9:45

1aNS3. Thwarted by data, saved by fundamentals. K. Anthony Hoover (McKay Conant Hoover Inc., 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362, thoover@mchinc.com), Zachery O. L'Italien, and Henry Ashburn (McKay Conant Hoover Inc., Westlake Village, CA)

The overwhelming majority of acoustic data that are available to most architectural design professionals is in abbreviated single-number ratings, without details or full test reports. Omitted key details can greatly affect the actual performance. Despite best intentions to satisfy codes and criteria, most of these designers do not adequately understand the acoustic fundamentals to recognize and then avoid significant problems and under-designed projects. Examples of problems with data regarding sound isolation, HVAC noise, and sound-absorption will be discussed, along with suggestions toward better understanding and more desirable/reliable end results.

10:05

1aNS4. Modeling a representative room to evaluate sound measurement methods with improved reproducibility for ASTM ISR testing. Sunit Girdhar (Mech. Eng., Michigan Technol. Univ., 207 Vivian St., Hancock, MI 49930, sgirdhar@mtu.edu), Andrew Barnard (Acoust., Penn State Univ., State College, PA), Jason R. Blough (Mech. Eng., Michigan Technol. Univ., Houghton, MI), John Lo Verde, and Wayland Dong (Veneklasen Assoc., Santa Monica, CA)

The low-frequency one-third octave bands for the standard ASTM ISR (Impact Sound Rating) are generally non-diffuse for small rooms we usually come across for field testing. This is because of low modal density in low frequencies in small rooms. Because of the non-diffusivity, the Sound Pressure Level (SPL) measured by microphones is controlled by the room modes. This increases the non-reproducibility of the test method. In the current standards, the modal density and the room non-diffusivity are ignored. Recent research shows that it is important to measure in lower frequencies, at least down to 50 Hz one-third octave band to characterize the impact performance of lightweight structures. With these lower frequencies, the non-diffusivity problem will be more pronounced. In our work, we are simulating a representative room and trying out different sound measurement ideas that will not depend on the non-diffusivity of the room, therefore, improving the reproducibility of the test method. This work will be followed up with some testing on real structures in real rooms. An initial study from this work was presented at the ASA 2021 Seattle meeting.

10:25–10:45 Break

Contributed Papers

10:45

1aNS5. Floor covering standards based on improvement of impact insulation ratings. Wayland Dong (Veneklasen Assoc., 1711 Sixteenth St., Santa Monica, CA 90404, wdong@veneklasen.com) and John Lo Verde (Veneklasen Assoc., Santa Monica, CA)

A common acoustical design task is predicting the impact rating of an existing floor system that is modified by adding or changing flooring elements. Most codes and regulations are based on impact testing of the resultant system. It might seem that floor coverings could instead be specified using laboratory testing of the improvement in impact sound insulation of floor coverings (ΔIIC test) per ASTM E2179. However, in practice the ΔIIC rating does not accurately predict the resultant assembly. Here, we investigate several avenues that may allow improved prediction of impact ratings. The first path is modification the ΔIIC rating by removing the rule limiting the deficiency in any third-octave band to 8 dB (“the 8 dB rule”). The 8 dB rule has been shown to result in a 4-point bias of the ΔIIC rating, which may account for a large portion of the inaccuracy. A second path is using the high-frequency impact rating, ΔHIIC instead of ΔIIC, which is promising since flooring improvements are largely limited to the high-frequency range. A third path is defining a standard lightweight (i.e., wood-framed) reference floor to complement the existing concrete reference floor. These investigation may yield impact requirements for flooring that are easier to understand, specify, and verify.

11:00

1aNS6. Application of acoustical control through building codes and regulatory requirements. Samantha Rawlings (Veneklasen Assoc., 1711 16th St., Santa Monica, CA 90404, srawlings@veneklasen.com), John Lo Verde, and Wayland Dong (Veneklasen Assoc., Santa Monica, CA)

The International Building Code (IBC) establishes minimum acoustical performance requirements for permanent and transient housing facilities. The lab parameters for acoustical performance within the IBC are useful during the design and permitting process but fail to provide code officials, builders, designers, and residents a method for evaluating the success of a completed building. The latest update to the IBC in 2021 establishes standards and metrics for use in evaluating post-construction code compliance to mutually protect all stakeholders (builders, code officials, occupants, municipalities, etc.) in housing facilities and reduces historic ambiguity in the code. Historically, the nature of the metrics and their ability to evaluate success has been limited (“Field impact insulation testing: Inadequacy of existing normalization methods and proposal for new ratings analogous to those for airborne noise reduction,” LoVerde/Dong, *JASA*, 2005). There is some historic precedent regarding the application and measurement requirement for field metrics (“Coping with uncertainties in the design and evaluation of acoustical assemblies,” LoVerde/Dong, ASA Hong Kong, 2012). This presentation expands this conversation to examine regulatory requirements and discusses application of field verification of acoustical performance metrics.

11:15

1aNS7. Sound pressure-based ratings are preferred for evaluation of *in situ* sound isolation. Wayland Dong (Veneklasen Assoc., Santa Monica, CA), John Lo Verde (Veneklasen Assoc., 1711 Sixteenth St., Santa Monica, CA 90404, jloverde@veneklasen.com), and Samantha Rawlings (Veneklasen Assoc., Santa Monica, CA)

The International Building Code has recently been revised to explicitly specify the ratings to use for field (*in situ*) measurements of airborne and impact sound isolation in residential properties. We review the differences between the various ratings, which (on the surface) are defined by their normalization procedure. Two normalization options are defined in current ASTM and ISO standards: a standard reverberation time of 0.5 s, and a standard amount of absorption of 10m2. It is not always appreciated that the choice of normalization is a choice of the type of measurement being performed. Normalization to a standard reverberation time is based on sound pressure and hence is a measurement of the acoustic isolation between spaces, while normalization to a standard amount of absorption is based on apparent sound power and hence is a measurement of the performance of the separating assembly. It is sometimes assumed the sound power-based ratings are representative of the sound isolation experienced by occupants, but they often provide an inaccurate assessment of the resultant isolation. Analysis and examples are presented, and it is demonstrated that the sound pressure-based ratings are preferred.

Session 1aPAa

Physical Acoustics, Computational Acoustics, Engineering Acoustics, Animal Bioacoustics, and Signal Processing in Acoustics: Acoustical Remote Sensing in Urban Environments

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

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

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

Invited Papers

8:00

1aPAa1. Low-frequency acoustic monitoring in urban environments. Sarah McComas (US Army Res. and Development Ctr., 3909 Halls Ferry Rd., Vicksburg, MS 39180, sarah.mccomas@usace.army.mil), Stephen Arrowsmith, Chris Hayward, Brian Stump (Southern Methodist Univ., Dallas, TX), and Mihan McKenna (US Army Res. and Development Ctr., Vicksburg, MS)

Arrays of infrasound sensors are commonly deployed in quiet rural settings to monitor high energy/low-frequency sources at distances of hundreds to thousands of kilometers. Advancements in infrasound sensor technology allow for measurement across the acoustic spectrum from infrasound (< 20 Hz) to low end audible (< 1000 Hz). This supports a growing interest in using infrasound arrays to monitor low energy/higher frequency sources at local propagation distances (<100 km). Examples of these sources include vehicles (ground and air), small explosions, and infrastructure (e.g., bridges). Many of these sources are driven by anthropogenic activity. In order to successfully monitor them, arrays of sensors will need to be installed closer to the sources of interest, thereby requiring arrays to encroach on urban spaces. The design, deployment, and utilization of these arrays will face challenges, such as limited open ground for installation and source signals that need to be separated from a complex acoustic noise field to be observed. This presentation shares techniques for instrumenting the urban environment and characterizing the ambient acoustic fields in three different urban environments. [Permission to publish was granted by the Director, Geotechnical and Structures Laboratory, U.S. Army Engineer Research and Development Center.]

8:30

1aPAa2. A survey of statistical models for urban noise and their physical interpretations. D. Keith Wilson (US Army Engineer Res. and Development Ctr., U.S Army ERDC-CRREL, 72 Lyme Rd., Hanover, NH 03755-1290, D.Keith.Wilson@usace.army.mil), Matthew J. Kamrath, Caitlin E. Haedrich, Daniel J. Breton, and Carl R. Hart (US Army Engineer Res. and Development Ctr., Hanover, NH)

Many different models have been used to describe statistical distributions of sound in urban environments. Some models may be justified empirically, whereas others are linked to physical phenomena such as random fading, multipath, multi-source mixtures, and variations in source-receiver geometries. This presentation reviews many of the available statistical distributions, what they are intended to represent physically, and their appearance on linear and logarithmic (decibel) axes. To evaluate the suitability of the various distributions, comparisons are made to an experiment in which one-third octave band sound-level data were measured at 37 locations in the North End of Boston, Massachusetts. Based on the Kullback-Leibler divergence as calculated across all the locations and frequencies, the exponentially modified Gaussian (EMG) distribution provides the most consistently good agreement with data, because it captures the positively skewed sound levels present in most of the data. The compound gamma distribution, which is applicable to situations involving varying sound levels, also fits the data well and even outperforms the EMG for the small minority of cases exhibiting negative skew. The log-normal distribution often provides a suitable fit in cases where particular non-traffic noise sources dominate.

9:00

1aPAa3. Combining remote sensing and artificial intelligence to locate sounds. Macarena Varela (Sensor and Data Fusion, Fraunhofer FKIE, Fraunhoferstr. 20, Wachtberg, NRW 53111, Germany, macarena.varela@fkie.fraunhofer.de), Ravali Nalla, and Wulf-Dieter Wirth (Sensor and Data Fusion, Fraunhofer FKIE, Wachtberg, NRW, Germany)

Smart cities use sensors to collect and analyze data to manage their resources more efficiently and thereby enhance the quality of life for residents, especially in densely populated cities. They monitor a wide range of information, including pollution, traffic, and parking. Urban environments feature large numbers of sounds, such as noise pollution and other specific sounds, which can be harmful to citizens. Therefore, this unrestrained growing issue of urban sounds should be addressed by smart cities to improve noise mitigation. To sidestep that problem, the "Listening system Using a Crow's nest arraY" (LUCY) is currently in development at Fraunhofer FKIE. The acoustic system aims are to automatically detect meaningful audio events contained in noisy data, such as impulsive sounds, and to accurately estimate their geographic locations. To accomplish these tasks in near real-time, LUCY has to combine advanced array processing techniques, including beamforming, with artificial intelligence methods, such as deep learning using spectro-temporal features. The proposed acoustic system is a low-cost, small and lightweight system. It consists of a peculiar volumetric array of tiny MEMS microphones, called the "Crow's Nest Array" (CNA), which has a crucial influence on the accuracy of the sound localization estimation, and a miniature computer to process methods including sound localization calculation. Due to its small size, LUCY can easily be deployed on numerous types of platforms, including Unmanned Aerial Vehicles (UAVs).

9:30

1aPAa4. Acoustic array signal processing with the Stockwell transform. Steven Collar (NCPA, Univ. of MS, University, MS) and Garth Frazier (NCPA, Univ. of MS, NCPA, University of MS, P.O. Box 1848, Oxford, MS 38677, frazier@olemiss.edu)

The Stockwell transform is a wavelet-like transformation based on a Fourier kernel weighted by a symmetric, frequency-scaled, time-shifted window function. Thus, it is suitable for analysis of non-stationary waveforms and transients in particular. In its discrete-time, orthogonal basis realization known as the discrete orthogonal Stockwell transform (DOST), it is possible to transform an N-point sequence in O(N log N) operations. In this presentation, we show how to perform direction-of-arrival and transient waveform estimation in the Stockwell basis in a manner that is similar to traditional multi-channel frequency-domain (discrete Fourier transform) techniques. This enhances detection of multiple transients within the same data frame sequence.

10:00

1aPAa5. Simulation of near-ground signals from a flying source on UAV over a building structure. Jiacheng Hou (Mech. and Aerosp. Eng., Utah State Univ., 4130 Old Main Hill, Logan, UT 84322-4130, jiachenghou@aggiemail.usu.edu) and Zhongquan Charlie Zheng (Mech. and Aerosp. Eng., Utah State Univ., Logan, UT)

Acoustic signals near the ground generated by a moving source on a fly-by UAV are simulated around a house. The simulation is carried out using a time-domain acoustics solver that can simulate acoustic propagations with the specified moving source, ground properties, and building geometries. The source on a UAV is approximated by a broadband source moving at a constant speed. The long-range three-dimensional computation is developed with a ground as a rigid or porous medium and a residential house with realistic geometries. Time histories and histograms of the near-ground sensors at different locations around the house are analyzed with their different behaviors due to Doppler shift, ground effect, and acoustic interference from the house structures. Comparisons will be made with literature results and available measured data.

10:30–10:45 Break

10:45

1aPAa6. Prediction of shooter localization accuracy in an urban environment. Luisa Still (Sensor Data and Information Fusion, Fraunhofer FKIE, Fraunhoferstraße 20, Wachtberg 53343, Germany, luisa.still@fkie.fraunhofer.de) and Marc Oispuu (Sensor Data and Information Fusion, Fraunhofer FKIE, Wachtberg, Nordrhein-Westf (10), Germany)

The use of acoustic sensor networks for shooter localization can provide a vital contribution to situational awareness in urban environments. During mission planning, the performance prediction for acoustic shooter localization plays a central role as it allows an optimization of the sensor positions in a sensor network. Instead of a sound propagation-based approach, in our work we focus on an information theoretical analysis using the Cramér-Rao bound to predict the achievable shooter localization accuracy. Through this approach, we have shown that accounting for incomplete and heterogeneous acoustic measurement data sets leads to maximization of the fusion gain and consequently to improved achievable localization accuracy. We validated the match between predicted and actual experimental performance in free-field measurements with supersonic gunshots including varying sensor-to-shooter geometries, weapon types, and various measurement types. By measuring signatures of impulsive gas cannon shots in urban terrain, we analyzed the effect of buildings to the sensor performance and achieved a significant improvement in the localization performance prediction by adjusting the sensor model depending on whether line-of-sight or non-line-of-sight conditions to the target exists.

11:15

1aPAa7. Improvements of the Stevens drone acoustic detection system. Daniel Kadyrov (STAR Ctr., Stevens Inst. of Technol., Hoboken, NJ, dkadyrov@research.stevens.edu), Alexander Sedunov, Nikolay Sedunov, Alexander Sutin, Hady Salloum, and Sergey Tsyuryupa (STAR Ctr., Stevens Inst. of Technol., Hoboken, NJ)

Stevens Institute of Technology (SIT) designed and built multiple acoustic sensors to detect and track drones using Steered-Response Phase Transform (SRP-PHAT) and classify them using narrow-band frequencies. SIT improved a previously built four-microphone system by increasing the number of microphones to seven, modifying the software, and improving the testing algorithm for system performance estimation. The four- and the seven-microphone systems were deployed during several tests conducted with multi-rotor UAVs of different sizes, including the DJI Inspire 2, DJI Mavic 2 Pro, Autel Robotics EVO II Pro, and the Intel Falcon 8 performing flight patterns at various distances, elevations, and speeds. The tests were conducted in an environment with low-flying aircraft and vehicular traffic noise. Acoustic signatures were collected and detection distances were compared for the tested UAVs. The improved seven-microphone system provided farther detection and classification distances than the previous four-microphone system. Typical detection distances were between 300 and 500 m with a bearing accuracy within five degrees. An analysis of the effect of external noise on performance was also performed.

11:30

1aPAa8. Role of multiple reflections in identifying street canyons in microscale urban spaces. Dorian Davis (Univ. of the District of Columbia, 4200 Connecticut Ave. NW, Washington, DC 20008, dorian.davis@udc.edu), Lirane Mandjoupa, Samba Gaye, Shehabaldin Mohamed, Etochukwu Etochukwu, Wagdy Mahmoud, Lei Wan, and Max Denis (Univ. of the District of Columbia, Washington, DC)

In this work, geometric acoustic numerical modeling of sound propagation in a micro-scale urban spaces is presented. The resultant sound propagation demonstrates an amplification of sound and increased reverberation time, due to the multiple reflections in the micro-scale urban spaces. Comparative analysis of the sound field acoustic parameters for various length and width between geometric boundaries can differentiate street canyons from other urban micro-scale spaces. Specifically, sound pressure level results are strongly influenced by the geometric configuration and material properties of the geometric boundary surfaces. The results and limitations of the geometric numerical method for predicting sound propagation in urban environments are discussed.

11:45

1aPAa9. 3D noise modeling of an urban environment: Simulated and measured noise characteristics of the University of the District of Columbia campus. Lirane Mandjoupa (Univ. of the District of Columbia, 4200 Connecticut Ave. NW, Washington, DC 20008, liranekertesse.mandj@udc.edu), Dorian Davis, Samba Gaye, Shehabaldin Mohamed, Etochukwu Etochukwu, Wagdy Mahmoud, Lei Wan, and Max Denis (Univ. of the District of Columbia, Washington, DC)

In this work, a 3-D numerical method is employed to predict the noise propagation within the University of the District of Columbia (UDC). UDC is located within an urban environment in Washington, DC. Sound pressure level measurements of various noise sources including traffic noise are obtained. Of particular interest is the method's ability to accurately simulate the urban space. Accuracy is evaluated comparatively between simulated and measured results of acoustic parameters in the mid- to high-frequency regimes. Specifically, reverberation time, energy decay time, and sound pressure level. The results and limitations of the particle-tracing method for predicting sound propagation in urban environments are discussed.

Session 1aPAb

Physical Acoustics and Biomedical Acoustics: Advances in Sonochemistry I

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

Chair's Introduction—9:55

Invited Papers

10:00

1aPAb1. Sonofragmentation and sonocrystallization: How solids break and make in cavitating liquids. Kenneth S. Suslick (Dept. of Chemistry, Univ. of Illinois at Urbana-Champaign, 600 S. Mathews Av., Urbana, IL 61822, ksuslick@uiuc.edu)

Mechanical action can produce dramatic physical and mechanochemical effects when the energy is spatially or temporally concentrated. The application of ultrasound to crystallization (i.e., sonocrystallization) can dramatically affect the properties of the crystalline products. Sonocrystallization induces rapid nucleation, generally yields smaller crystals of a more narrow size distribution compared to quiescent crystallizations, and has become increasingly important in the pharmaceutical industry for the preparation of APIs (active pharmaceutical ingredients). The control of morphology of the crystallization process is critical to reproducible dose response for APIs and is under increasing scrutiny in pharmaceutical manufacturing by the FDA. Ultrasound can induce significant improvement in the uniformity of crystallite size and rates of crystallization. We have developed a mechanistic understanding of the origin of these phenomena and begun to separate the details of the effects of ultrasound on nucleation, mass transport, shockwave fragmentation of crystallites, and inter-particulate collision. Decoupling experiments were performed to confirm that interactions between shockwaves and crystals are the main contributors to crystal breakage. We have discovered a mechanochemical extension the Bell–Evans–Polanyi principle: activation energies for solid fracture correlate with the binding energies of the solids.

10:30

1aPAb2. Sonochemical synthesis of electrocatalysts for low-temperature water electrolysers. Bruno G. Pollet (Université du Québec à Trois-Riviéres (UQTR), 3351, boulevard des Forges, Trois-Rivières, QC G8Z 4M3, Canada, bruno.pollet@uqtr.ca), Henrik E. Hansen, Svein Sunde, Odne S. Burheim, and Frode Seland (Norwegian Univ. of Sci. and Technol. (NTNU), Trondheim, Norway)

An important step in the process of producing hydrogen a viable method is to improve the efficiency and reduce the cost of low-temperature water electrolysers. One of the most crucial components is the catalyst used to drive the hydrogen evolution reaction (HER) and the oxygen evolution reaction (OER). Traditionally, the nanosized electrocatalysts are synthesized through a chemical reduction method involving a strong reducing agent like sodium borohydride, polyol, etc. Being able to control the nucleation and growth and therefore the size of the nanocatalysts, however, is not straightforward with the chemical reduction method where the use of surfactants is heavily relied upon, thus complicating the method for the industry. An alternative synthesis route involves the *in-situ* generation of radicals to serve as reducing agents through high power ultrasound (20 kHz–1 MHz) in a process where water is split into OH- and H-radicals referred to as water sonolysis. This presentation highlights the effects of ultrasonication frequencies, ultrasonication, times, pH solutions, reducing agents, and different saturation gasses on the generation of metallic nanoparticles and their subsequent electrocatalytic activities towards the HER and OER in mild acidic and alkaline environments. A series of physical characterizations on these sonochemically prepared nanoelectrocatalysts will be shown and discussed.

11:00

1aPAb3. Importance of sonoluminescence and sonochemistry for understanding and optimizing ultrasound applications. Judy Lee (Univ. of Surrey, Chemical and Process Eng., University of Surrey, Guildford GU27XH, United Kingdom, j.y.lee@surrey.ac.uk)

Ultrasound has shown to improve a spectrum of applications such as crystallization, degradation, synthesis, extraction, and cell disruption. These improvements largely stem from the cavitation induced extreme physical and/or chemical effects. Cavitation activities are often quantified using sonoluminescence intensity or sonochemical yield and are dependent on sonication conditions. For example, frequencies less than 100 kHz produces stronger physical effects and higher sonoluminescence per bubble, whereas frequencies between 200 and 800 kHz give rise to higher sonochemical yields. Similarly, solution conditions are equally important as 50% air saturation or addition of 1 mM anionic surfactant (sodium dodecyl sulfate) have shown to enhance sonoluminescence intensity. Often frequency and power are varied to assess its impact on a given application. However, this often do not reveal the underlying mechanisms to clearly assess whether the system could be further optimized or better designed to tailor for a particular application. This presentation aims to show the importance of complimentary evaluation of the sonoluminescence and sonochemical yield and also provide case studies on sonocrystallization, degradation of pollutants, and sono-grafting on membranes.

11:30

1aPAb4. Probing the mechanisms of sonodynamic therapy. Eleanor P. Stride (Univ. of Oxford, Inst. of Biomedical Eng., Oxford OX3 7DQ, United Kingdom, eleanor.stride@eng.ox.ac.uk)

Drugs that can be activated by a physical stimulus at a target site offer great potential for limiting toxic side effects, e.g., in cancer therapy. In photodynamic therapy, light is used as the stimulus, but clinical applications are limited due to the poor penetration of light in tissue. Multiple studies have demonstrated that ultrasound can also be used to activate some light-responsive drugs. This potentially extends the range of therapeutic applications considerably, but the mechanisms underpinning drug activation, dubbed sonodynamic therapy (SDT), remain unclear. It was recently demonstrated that multi-bubble sonoluminescence could be detected from a suspension of microbubble ultrasound contrast agents when exposed to ultrasound under conditions similar to those used in SDT. Moreover, the addition of an SDT drug produced a reduction in optical emissions at the wavelength corresponding to its activation. Numerous questions, however, remain and the aim of this talk is to review recent studies of SDT and new evidence for the roles of sonoluminescence, sonoporation, and other phenomena hypothesized to play a role in its mechanism of action.

MONDAY MORNING, 23 MAY 2022 GOVERNORS SQUARE 12, 8:15 A.M. TO 12:00 NOON

Session 1aSA

Structural Acoustics and Vibration and Physical Acoustics: Nonlinear Metamaterials and Phononics

Samuel P. Wallen, Cochair
Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet RD, Austin, TX 78758

Ganesh U. Patil, Cochair
*Mechanical Science and Engineering, University of Illinois Urbana-Champaign,
144 Sidney Lu Mechanical Engineering Building, MC-244, 1206 West Green Street, Urbana, IL 61801*

Chair's Introduction—8:15

Invited Papers

8:20

1aSA1. Harnessing rotational geometry to design reconfigurable dispersion and refractive index in nonlinear acoustic metamaterials. Lezheng Fang (Georgia Inst. of Technol., 771 Ferst Dr. N.W., Atlanta, GA 30332-0405, lezheng.fang@gatech.edu) and Michael Leamy (Georgia Inst. of Technol., Atlanta, GA)

We investigate 1D and 2D monatomic lattice structures composed of in-plane rotators coupled by angled elastic linkages and explore their reconfigurable dispersion, negative refraction, amplitude-dependent dynamics, and acoustoelastic effect. At small amplitude, the linear band structure can be configured to be either acoustic or optical in nature by changing the connecting locations on the rotators, which corresponds to positive and negative refractive index. An interface problem between two rotator lattices with opposite dispersion type is modeled, and illustrates negative refraction in numerical simulations—experimental measurements are in progress. Its frequency–transmission relation matches the linear theory. At higher amplitude, the hardening nonlinearity shifts the dispersion and induces amplitude-dependent transmission. A novel nonlinear phenomenon–amplitude saturation (a constant far-field transmission independent of input amplitude) is observed when the transmitted wave falls in the nonlinear stopband and simultaneously the linear passband of the receiving lattice. We analyze the nonlinear effects via perturbation methods and propose a framework for evanescent-specific nonlinear waves to explain the saturation phenomenon. Additionally, we observe a strong acoustoelastic effect in chiral-patterned rotator lattices, where a static stretch can shift the equilibrium positions of the rotators and morph the band structure from acoustic to optical.

8:40

1aSA2. Dispersion relation for harmonic generation in nonlinear elastic waves. Romik Khajehtourian (Dept. of Mech. and Process Eng., ETH Zurich, Zurich, Switzerland) and Mahmoud I. Hussein (Aerosp. Eng. Sci., Univ. of Colorado Boulder, 3775 Discovery Dr., AERO 354, 429 UCB, Boulder, CO 80303, mih@colorado.edu)

We present a theory for the dispersion of generated harmonics in a traveling nonlinear wave. The harmonics dispersion relation—derived by the theory—provides direct and exact prediction of the collective harmonics spectrum in the frequency-wavenumber domain and does so without prior knowledge of the spatial-temporal solution. The new relation is applicable to a family of initial wave functions characterized by an initial amplitude and wavenumber. We demonstrate the theory on nonlinear elastic waves in a homogeneous rod and demonstrate its extension to periodic rods. We investigate a thick elastic rod admitting longitudinal motion. In the linear limit, this rod is dispersive due to the effect of lateral inertia. The nonlinearity is introduced through either the stress–strain relation and/or the strain–displacement gradient relation. Using a theory we have developed earlier, we derive an exact general nonlinear dispersion relation for the thick rod. We then derive a special case of this relation and show that it provides an exact prediction of the generated harmonics spectrum, in frequency versus wavenumbers. Both relations are validated by direct time-domain simulations, examining both instantaneous dispersion (by direct observation) for the general nonlinear dispersion relation and short-term, pre-breaking dispersion (by Fourier transformations) for both the general and specialized relation.

Contributed Papers

9:00

1aSA3. Dispersion in lattices with patterns of hardening and softening stiffness nonlinearity. Matthew D. Fronk (Phys., U.S. Naval Acad., 572C Holloway Rd., Annapolis, MD 21402, fronk@usna.edu), Stephanie G. Konarski, Caleb F. Sieck, Alec K. Ikei, and Matthew D. Guild (Code 7160, U.S. Naval Res. Lab., Washington, DC)

Recent focus has been given to analytically predicting amplitude-dependent dispersion in periodic structures with weak stiffness nonlinearity. These dispersion relationships may inform devices such as amplitude-dependent filters, waveguides, and diodes. However, attention is generally restricted to either spatially uniform nonlinearity or mutually exclusive distributions of hardening or softening stiffness. This study investigates dispersion in lattices with spatial modulations in stiffness nonlinearity. Special attention is given to modulations consisting of both hardening and softening nonlinear terms. A multiple scales perturbation analysis reveals that patterns of hardening and softening stiffness enable both lifting and lowering of a passband's frequencies. In such cases, passbands contain discrete frequency and wavenumber pairs that prevent amplitude-dependent dispersion shifting. Numerical integration of the lattice equations of motion are carried out to confirm the analytically predicted dispersion behavior. A design of experiment is proposed in which strength and sign of nonlinearity can be tuned with the initial angle of additively manufactured grounding springs. [Work supported by the Office of Naval Research.]

9:15

1aSA4. Nonlinear elastic metamaterials as pulse shaping devices for shock test applications. 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 (Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX), and Washington DeLima (Honeywell, Federal Systems, Kansas City, MO)

Metamaterials (MM) have become a very active research topic in numerous domains of engineering because of their promise to create structures and devices that can control wave propagation in ways that exceed the capabilities of conventional homogeneous and composite materials. Most research on acoustic and elastic MM has been focused on linear behavior. However, linear MM suffer from notable drawbacks, for example, (i) the effective material properties for the application of interest are often limited to narrow frequency bands, and (ii) they have limited usefulness in applications where nonlinearity is unavoidable or essential, e.g., shock testing or high-intensity focused ultrasound. Nonlinearity has therefore been explored to expand the palette of accessible dynamic response of synthetic materials. In this work, we investigate applications for nonlinear elastic MM as pulse shaping materials for shock testing. By using high-resolution finite element methods in concert with direct numerical simulations of reduced-order dynamic models, we examine the propagation of elastic pulses under the influence of various types of nonlinear elastic response and identify means to obtain them. These simulations demonstrate the potential for nonlinear elastic MM to significantly expand the space of accessible excitations for shock testing, using a relatively small number of design parameters.

9:30

1aSA5. Inverse design of non-periodic nonlinear metamaterials for nonlinear ultrasonics. Pravinkumar R. Ghodake (Dept. of Mech. Eng., Indian Inst. of Technol., Bombay, Mumbai, Maharashtra 400076, India, mech7pkumar@gmail.com)

Previous work by the author gives an idea about the effective design of linear and phononic "periodic" metamaterial using shape optimization to suppress the system-generated 2nd harmonics of a longitudinal wave during nonlinear ultrasonic testing [Ghodake, *J. Acoust. Soc. Am.* **150**, A149 (2021)]. That work considers only the geometric nonlinearity of the layered materials and sinusoidal waves as an input short pulse. The present work implements a shape optimization technique to obtain the optimal design parameters of the "non-periodic" metamaterial by considering both material and geometric nonlinearities of the layered materials present in the phononic crystal as this assumption is close to the real experimental situations. Gaussian short input pulses are considered in this study with two different objectives such as only reducing the amplitude of the 2nd harmonics and reducing the amplitude of 2nd harmonics, maximizing the amplitude of 1st harmonics as well as maintaining the Gaussian shape of the output pulse which makes this inverse problem challenging. This study demonstrates the applicability of the non-periodic metamaterial in practical situations and the importance of the proposed inverse design approach in linear and nonlinear elastic wave propagation applications such as ultrasonic waves and seismic waves.

9:45–10:00 Break

Invited Papers

10:00

1aSA6. Amplitude-dependent edge states and discrete breathers in non-linear modulated phononic lattices. Massimo Ruzzene (Dept. of Mech. Eng.,Univ. of Colorado, P. M. Rady 1111 Eng. Dr UCB 427, Boulder, CO 80309, massimo.ruzzene@colorado.edu), Matheus Rosa (Univ. of Colorado, Boulder, CO), and Michael Leamy (Georgia Inst. of Technol., Atlanta, GA)

We explore the role of non-linearities on the spectral properties of modulated one-dimensional phononic lattices. In the linear regime, a spatial modulation of stiffness is known to produce topological gaps characterized by non-zero Chern numbers, which host topological states localized at the edges of finite domains. A continuation of the linear modes as a function of amplitude is performed, revealing a series of localization and de-localization transitions that are confirmed through direct time domain simulations. The results show that edge states whose eigenvalue branch remains within the gap remains localized, and therefore appear to be robust with respect to amplitude. In contrast, edge states whose corresponding branch approaches and remains tangential to the bulk bands experience delocalization transitions. Additionally, we observe a series of amplitude-induced phase transitions as the bulk modes become discrete breathers localized in one or more regions of the domain. Remarkably, these transitions are independent on the size of the lattice. These results bring to light the co-existence of topological edge states and discrete breathers for non-linear modulated lattices, whose interplay may exploited for amplitude-induced topological and localization transitions.

10:20

1aSA7. Collisions of nonlinear waves in flexible mechanical metamaterials. Hiromi Yasuda, Hang Shu, Weijian Jiao (Univ. of Pennsylvania, Philadelphia, PA), Vincent Tournat (CNRS, Le Mans Université, Le Mans, France), and Jordan Raney (Univ. of Pennsylvania, Towne Bldg. Rm 274, 220 S. 33rd St., Philadelphia, PA 19104-6272, raney@seas.upenn.edu)

Flexible mechanical metamaterials are compliant structures designed to achieve desired mechanical properties via large deformation or rotation of their components. While their static properties (such as Poisson's ratio) have been studied extensively, much less work has been done on their dynamic properties, especially nonlinear dynamic properties induced by large movement of internal components. Here, we examine the nonlinear dynamic response arising from impact loading of mechanical materials that consist of 1D and 2D arrangements of rotating squares, which leads to formation of solitons. Permanent magnets are added to the squares, which causes the metamaterial to become multistable. Rotations of the squares can thereby lead to sudden rearrangements of squares into new phases. We experimentally and numerically characterize the collisions of solitons in these flexible mechanical metamaterials, which, depending on their amplitude and chirality, can induce a variety of responses, including phase transitions.

Contributed Papers

10:40

1aSA8. Experimental study of nonlinear waves in phononic materials with rough contacts. Ganesh U. Patil (Mech. Sci. and Eng., Univ. of Illinois Urbana-Champaign, 144 Sidney Lu Mech. Eng. Bldg., MC-244, 1206 West Green St., Urbana, IL 61801, gupatil2@illinois.edu) and Kathryn Matlack (Univ. of Illinois at Urbana-Champaign, Urbana, IL)

Incorporating contacting interfaces in periodic media enables enriched wave dynamics from the combined effects of periodicity and nonlinearity. While Hertzian contacts have been extensively studied in the framework of engineered media, rough contacts have received relatively less attention. Recently, our numerical work showed that continuum phononic material with periodic rough contacts supports strongly nonlinear waves belonging to the family of solitary waves. In this talk, we experimentally study nonlinear waves in phononic material based on rough contacts, realized through an array of aluminum disks. The disks act as elastic layers and have roughness on either side, which is generated through surface treatment. First, we characterize a rough contact through optical imaging and ultrasonic measurements. We obtain the nonlinear contact stiffness–precompression relationship that informs the power exponent of the contact law. Then, we study nonlinear wave propagation through periodic rough contacts of the same roughness topography. We measure propagating waves through a laser vibrometer and investigate their frequency content, speed, and amplitude. This study develops a fundamental understanding of the role of rough contacts in the form of local nonlinearity on wave responses. Such understanding is useful in designing these phononic materials for wave propagation control.

10:55

1aSA9. Wave propagation in a continuum phononic material with geomaterial-inspired nonlinearity. Elizabeth Smith (Mech. Sci. and Eng., Univ. of Illinois at Urbana Champaign, Mech. Eng. Bldg., 1206 W Green St., Urbana, IL 61801, esmith19@illinois.edu) and Kathryn Matlack (Univ. of Illinois at Urbana-Champaign, Urbana, IL)

This presentation discusses the nonlinear wave propagation in a continuum phononic material with nonlinear mechanisms inspired by geomaterials. Geomaterials can exhibit nonlinear wave responses due to mechanisms such as material nonlinearity or lack of bonding between phases of material. We study the interaction of nonlinear and phononic behavior by introducing two geomaterial-inspired nonlinear mechanisms into a phononic material. The influence of material nonlinearity and nonlinearity due to delaminations between material phases are studied in a bilayer finite element model composed of alternating layers of stiff linear elastic material and soft hyperelastic material. To model delaminations, a binary contact condition is applied at the interface between layers. Measurements of the quasi-static nonlinear mechanical response of delaminations of different lengths are used as inputs into the finite element method simulations. The influence of the delamination nonlinearity plus material nonlinearity on the phononic material is studied in a full-scale time-domain finite element model. This model is used to further probe the interaction between phononic band gaps and nonlinear wave response, as well as the dependence of delamination size and location on the nonlinear wave response.

11:10

1aSA10. Breaking reciprocity to realize extreme energy isolation in coupled oscillators. Chengen Wang (Mech. and Mater. Eng., Univ. of Nebraska-Lincoln, Lincoln, NE) and Keegan J. Moore (Mech. and Mater. Eng., Univ. of Nebraska-Lincoln, W336 Nebraska Hall, Lincoln, NE 68588, kmoore@unl.edu)

This research investigates the isolation achieved by breaking the reciprocity of between two coupled oscillators. The two oscillators have equal mass and the first one is linearly grounded and called the linear oscillator (LO). The second oscillator is nonlinearly coupled to the LO and is termed the nonlinear oscillator (NO). By breaking dynamical reciprocity using asymmetry and nonlinearity, the LO–NO system is shown to exhibit regimes of extreme energy isolation in only one of the oscillators as well as regimes where energy is exchanged between them. These regimes are shown to arise under both impulsive and harmonic excitation. The resulting system is governed by two nonlinear normal modes (NNMs), which can interact with each other under internal resonance of different ratios. Under different loading scenarios, different energy isolations are illustrated. This research starts with analytical study using numerical simulations that assess how energy distributes in the structure under varying loads. The analytical predictions are validated experimentally for both impulsive and harmonic excitations. The results of this research demonstrate that there remains much to learn about energy transfer in general and the breaking of dynamic reciprocity may lead to new types of acoustic and vibrational metamaterials.

11:25

1aSA11. Experimental elastic wave control in a piezoelectric phononic crystal with spatio-temporal modulation of electrical conditions. Sarah Tessier (IEMN, Lille, FR, 41 Boulevard Vauban, Lille 59800, France, sarah.tessier@junia.com), Charles Croënne (IEMN, Lille, FR, Lille, France), Florian Allein (Junia, UMR 8520, Lille, France), Jérôme Vasseur, and Bertrand Dubus (IEMN, Lille, FR, Lille, France)

This work concerns the propagation of elastic waves in a piezoelectric phononic crystal made of several identical piezoelectric elements separated by electrodes. In such a structure, electrical conditions prescribed to the electrodes can be modulated in space and in time and therefore can strongly modify the dispersion curves of the phononic crystal. For example, a periodic grounding of the electrodes introduces a Bragg band gap which does not exist when the electrodes have a floating potential condition [Degraeve *et al.*, *J. Appl. Phys.* **115**, (2014)]. When the electrical conditions of the electrodes are modulated in space and in time, dispersion curves are tilted with respect to the configuration in the reference medium and nonreciprocal wave propagation may occur at some modulation speeds in specific frequency bands [Croënne *et al.*, *Appl. Phys. Lett.*.**110**, (2017)]. The experimental set-up is based on a stack of 113 piezoelectric rings and 114 electrodes. The spatio-temporal modulation of the electrical conditions is performed using a digital signal processor (DSP) in order to reach high modulation speeds. This modulation is performed by shifting the position of the periodic ground conditions of the electrodes versus time. The experimental dispersion curves are obtained using measurements of the normal displacement of the piezoelectric rings with a scanning laser vibrometer. The evolution of dispersion curves as a function of modulation speed is analyzed and compared to finite element simulation results.

11:40–12:00
Panel Discussion

Session 1aSC

Speech Communication: Speech Production and Acoustics I (Poster Session)

Jahnavi Narkar, Chair
Linguistics, UCLA, 3125 Campbell Hall, Los Angeles, CA 90095

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

Contributed Papers

1aSC1. Stress and vowel reduction by Korean Learners of English. Ha Lim Park (English, Colorado State Univ. Alumna, ABC House B-501, 15-7, Nampyeong-ro 26beon-gil, Yangji-myeon, Cheoin-gu, Yongin-si, Gyeonggi-do 17161, Republic of Korea, halim@colostate.edu)

Having a foreign accent is unavoidable for late second or foreign language learners. The reason is that physical changes in the brain influence learning a variety of aspects in a new language system (Flege, 1987; Patkowski, 1990). In other words, a well-established first language mediates the acquisition of a second language. The mediation can be called cross-linguistics influence or negative transfer (Sharwood Smith and Kellerma, 1986). In terms of suprasegmental aspects of pronunciation, vowel reduction occurring in stress-timed languages such as English but lacking in syllable-timed languages is noteworthy to understand the cause of nonnative accents. The present work aims to explore patterns of vowel reduction in Korean accented English, with the goal of finding pronunciation issues in the English of Korean learners that might be due to transfer effects from an acoustic-phonetic standpoint. English speech samples from native English and Korean speakers of the Wildcat Corpus were used for data analysis. The vowel qualities, the duration ratios, and formant values of unstressed vowels produced by Korean and English speakers were analyzed. Results show that the Korean speakers tend not to reduce English unstressed vowels, assimilate them into similar vowels in their vowel inventory, and produce them as full vowels.

1aSC2. Speaker sincerity influences F0-coupled head motion in spontaneous monologues. Mark Tiede (Haskins Labs., 300 George St., Ste. 900, New Haven, CT 06511, tiede@haskins.yale.edu), Wei-Rong Chen (Haskins Labs., New Haven, CT), and D. H. Whalen (Haskins Labs., New York, NY)

Movements of the head during speech serve multiple communicative purposes, including perceptual enhancement of prosodic F0 contours. However, it remains uncertain how much of any correlation between head movement and F0 may be due to physiological coupling mechanisms exerting effects on glottal tension. In this work, six native speakers of American English (3F) were videorecorded during production of spontaneous monologues lasting 90 seconds. Each speaker extemporized twice on two topics chosen from a list (e.g., video games waste time). One of these they agreed with (sincere), and the other they pretended to agree with (insincere). Speaker head movements were quantified as displacement/rotation of a six DOF rigid body from the videos using OpenFace (Baltrušaitis, 2018). F0 for each speaker was converted to semitones relative to the median value for that speaker. Results show larger semitone excursion range but smaller head movement range for the sincere condition, with no significant differences in syllable rate across conditions. LME models predicting absolute semitone changes from head displacement show significant positive slopes for main effects of movement and sincerity, and an additive interaction for sincere productions. These differences suggest that, while head movement correlates with F0 overall, speaker sincerity can interact with the relation.

1aSC3. Acoustic exponents of emphasis in Jordanian Arabic. Aziz Jaber (English Lang., Yarmouk Univ., Dept. of English Lang., Yarmouk University, Irbid 21163, Jordan, aziz@yu.edu.jo)

This paper investigates the domain of emphasis spreading in Urban Jordanian Arabic. Emphasis spreading is examined in two types of words: polysyllabic monomorphemic and polymorphemic. F1 raising, F2 lowering, and F3 raising in the vowels preceding and following the emphatic sound are used as the acoustic correlates of emphasis spreading in the participants' speech who are 10 native speakers of Jordanian Arabic. The findings show that emphasis is a morphophonemic process in that the domain of emphasis spread in Urban Jordanian Arabic is the morpheme rather than the syllable or the word. Second, the study shows that emphasis spreading applies leftward and rightward in all environments but with some significant differences. Morpheme boundaries are found opaque to emphasis spreading.

1aSC4. An acoustic study on the voice of emotion in Beijing Chinese adults. Hanbo Yan (School of Chinese Studies and Exchange, Shanghai Int. Studies Univ., 550 West Dalian Rd., Bldg. 2, Rm. 418, Shanghai, Shanghai 200083, China, yanhanbo@shisu.edu.cn) and Yu-Fu Chien (Chinese Lang. and Lit., Fudan Univ., Shanghai, Shanghai, China)

Although Mandarin and Beijing Chinese are both tone language, Beijing Chinese, a dialect spoken in Beijing, is considered more colloquial. This study examines the acoustic properties of the vocal expression of emotions in Beijing Chinese. Using an emotional-stories elicitation method (Anolli *et al*., 2008), 30 native speakers of Beijing Chinese produced one target sentence in eight emotions, including anger, contempt, fear, guilt, joy, pride, sadness, and shame. The same sentence produced in a neutral condition was used as the baseline. The results showed that mean f0 values were high for shame and fear, while low for pride and anger. Pitch variation was high for shame and guilt, but low in sad and pride. Shame and sad also yielded high intensity variation. For speech rate, guilt was expressed with the fastest speech rate, while sad with the slowest. Longer pauses were found for sad, and shorter pauses for anger and guilt. Unlike Anolli *et al*. (2008), in which Mandarin speakers showed low f0 variation and slow speech rate for guilt, the current results showed that pitch variation was high for guilt, and it was characterized by the fastest speech rate in Beijing. It suggests that for tone language, regardless of the similar phonological systems, emotions were expressed differently in a more colloquial situation.

1aSC5. The four-way voicing distinction in Bengali infant directed speech. Jahnavi Narkar (Linguist, UCLA, 5057 Woodward, Detroit, MI 48202, jnarkar@ucla.edu) and Megha Sundara (Linguist, UCLA, Los Angeles, CA)

Although vowels in infant-directed Speech (IDS) are thought to be hyper-articulated, findings from consonants are more mixed. Stops in languages with three- and four-way voicing distinctions, in particular,

have been found to be hypo-articulated in IDS (Narayan and Yoon, 2011—Korean; Benders *et al*., 2019—Nepali) compared to adult-directed speech (ADS). We investigated the production of stops and affricates in Bengali which has a four-way voicing and aspiration distinction. IDS and ADS samples of connected speech produced by 10 native speakers (5 male and 5 female) of Bangladeshi Bengali (Yu *et al*., 2014) were compared. Stops and affricates were analyzed for voice onset time, onset f0 and lenition of the closure. Analyses controlled for prosodic position, lexical stress, and speaker sex—variables that are known to influence hypo-articulation. To compare the overlap between categories, acoustic measures were combined to obtain Pillai scores (Hay *et al*., 2006). Categories with low Pillai scores have more overlap and are thus, less distinct. The implications of the results for acquisition of phonetic categories will be discussed.

1aSC6. Evaluating the learnability of vowel categories from infant-directed speech. Jahnavi Narkar (Linguist, UCLA, 5057 Woodward, Detroit, MI 48202, jnarkar@ucla.edu), Ekaterina A. Khlystova (Linguist, UCLA, Los Angeles, CA), Ann M. Aly (TechFlow, Cape Coral, FL), Ji Young Kim (Spanish and Portuguese, UCLA, Los Angeles, CA), and Megha Sundara (Linguist, UCLA, Los Angeles, CA)

Hyper-articulated vowels are considered a hallmark of infant-directed Speech (IDS) and are thought to facilitate acquisition. Consistent with this idea, vowels in IDS are reported to be more peripheral than in adult-directed speech (ADS) (e.g., Trainor and Desjardins, 2002; Liu *et al*., 2005). However, there are also reports that vowels in IDS are more variable (Cristia and Seidl, 2014; Martin *et al*., 2015). We evaluated the learnability of vowel categories from IDS in two languages—American English, which has a crowded vowel space, and Mexican Spanish, which has fewer vowels. First, we used k-means clustering, an unsupervised learning algorithm, to partition vowels based on spectral and duration measures from conversational IDS. The algorithm was trained on IDS from 5 English speakers (Providence Corpus, Demuth *et al*., 2006) and 12 Spanish speakers (Aly *et al*., 2016). The vowel categories extracted from IDS were used to classify ADS vowels from 5 English speakers from the Buckeye corpus and 12 Spanish speakers (Kim and Repiso Puigderlliura, 2021). Next, we switched the training and testing set, by training on ADS, and testing on IDS. If IDS is facilitative, we expect a higher accuracy of vowel classification with training on IDS rather than ADS.

1aSC7. Acoustic characterization of the Costa Rican Non-Standard Trills. Mariana Cortes Kandler (Linguist, Univ. of Ottawa, 60 University Private, Office 333E, Ottawa, ON K1N 8Z4, Canada, mcort012@uottawa.ca)

This study characterizes the Costa Rican Spanish trill realizations acoustically, identifies the number of variants, and determines if they are acoustically distinguishable. Costa Rican trills typically lack vibration of the tongue tip and are thus categorized as 'non-canonical' compared to the normative rolled trill. Only impressionistic descriptions have been provided, posing challenges to comparisons across studies and dialects. Using Audacity, 18 speakers (9 female, $M = 37.4$) recorded their productions of 72 tokens containing the trill in word-initial/medial position and in stressed/unstressed syllables. The study was conducted remotely through Gorilla Experiment Builder. Duration, percentage of voicing and center of gravity were measured in PRAAT. Classification was based on visual inspection of the spectrograms and waveform, with the audio as a guide. ANOVAs were used to examine the distribution of the variants and their acoustic differences. Ten categories are characterized acoustically, the fricative being the most common variant. Stress made realizations longer but did not reach significance. Word-initially, fricatives and trills had a significantly higher percentage of devoicing, and while approximants and trills tended to be shorter in this position, fricatives had the opposite direction of effect. Fricatives also showed considerable variation in center of gravity ($M = 2159.2$ Hz, Range = 5622.3 Hz). This acoustic description of the non-standard variants serves as a parameter for future comparisons with other dialects.

1aSC8. Quantifying multiplex mergers in progress between moving targets. Valerie Freeman (Dept. of Comm Sci & Disord., Oklahoma State Univ., 042 Murray Hall, Stillwater, OK 74078, valerie.freeman@okstate.edu)

Pillai scores are becoming popular for quantifying degree of overlap, separation, or shift between two vowel distributions in F1 x F2 space (Nycz & Hall-Lew, *POMA*, 2013). However, some changes in progress involve more than two shifting vowels, multiple target locations, or variable directions of shift, making them difficult to quantify with single measures. An example are back/central pre-lateral mergers in Oklahoma, where /ʌl/ shifts toward /ol/ *dull-dole* and /ʊl/ shifts toward /ol/ or /ul/ *pull-pole* or *pull-pool*. This presentation compares several methods of numerically describing the location of /ʊl/ *pull* in relation to /ol, ul/, which tend to be separate but do not necessarily back to the same degree. They can serve as targets or "anchors" for the more variable /ʊl/, but without stable locations themselves, some sort of rotation, geometric transformation, or dimensional reduction is necessary. New methods under consideration include proportional locations between anchors, ratios of Pillai scores with each anchor, and areas of triangles between vowels. All will be evaluated for their specific usefulness to the example merger and for their applicability to less complicated shifts, like *dull-dole* in the same speakers, and differently complex shifts like three-way prevelar merger (*bag-beg-bagel*) in Pacific Northwest English.

1aSC9. Syllable-wide distribution of acoustic cues to coda voicing. Lian J. Arzbecker (Speech and Hearing Sci., The Ohio State Univ., 1070 Carmack Rd., 37 Pressey Hall, Columbus, OH 43210, arzbecker.1@osu.edu), Ewa Jacewicz, Riley Goebel, and Robert A. Fox (Speech and Hearing Sci., The Ohio State Univ., Columbus, OH)

In English, the strongest acoustic cues to preserving the voicing contrast in coda stops are in the preceding vowel: a voiced coda is associated with a longer vowel and a voiceless coda with a shorter vowel. Our recent work (Jacewicz *et al*., 2021) examining structured variability in stop voicing implementation in female productions showed that the cues to stop coda voicing extended to the syllable initial stop. In running speech, the /b/-closure in "bad" was shorter when coda was voiced, and it was longer when coda was voiceless ("bat"). Here, we test the hypothesis that voicing contrast cueing the "bad-bat" distinction is reinforced syllable-wide and involves specific long-distance timing relationships between closures of both stops, the extent of their closure voicing, vowel duration, and positive VOT. The current dataset consists of 2610 productions by 45 adult males, who are also diversified by dialect. Preliminary analyses confirmed that segmental voicing information is distributed over long domains (here, a monosyllabic word) and that cues to coda voicing are available in the syllable onset. These findings imply that temporal relationships among acoustic phonetic detail cueing lexical distinctions can potentially enhance the perceptual dimensions of perceived syllable- or word-wide voicelessness and voicing.

1aSC10. Acoustic analysis of vowel production in children with childhood apraxia of speech. Michelle T. Swartz (Speech Lang. Pathol., Thomas Jefferson Univ., 901 Walnut St., 23rd Fl., Philadelphia, PA 19107, michelle.swartz@jefferson.edu), Laura Koenig (Haskins Labs, New Haven, NY), and Elaine R. Hitchcock (Commun. Sci. & Disord., Montclair State Univ., Bloomfield, NJ)

Childhood apraxia of speech (CAS) is a complex neurological subtype of speech sound disorder (SSD), involving impaired speech motor planning and programming. Speech characteristics of CAS include difficulty in sequencing speech movements in the absence of muscle weakness resulting in segmental (e.g., vowel/consonant distortions) and suprasegmental (e.g., inappropriate lexical stress) speech deficits. Acoustic analysis offers a robust objective diagnostic measurement of CAS for lexical stress and consonant accuracy/consistency. However, other reported CAS features, such as vowel errors and distortions, have yet to be extensively validated using acoustic

analyses. This study will acoustically analyze vowel errors and inconsistencies for corner vowels (/i, u, æ, ɑ/) in children with CAS (aged 8;0 to 15;11) compared to peers with SSD and typical development (TD). Vowel space measures (e.g., vowel space area and formant centralization ratio) and consistency (e.g., vowel cluster distribution) will be assessed in 24 children (CAS = 4, SSD = 10, and TD = 10) across various phonetic contexts (e.g., syllable sequencing between anterior to posterior voiced/voiceless stops of increasing syllable length). It is predicted that children with CAS will demonstrate (1) greater vowel inconsistencies as context complexity increases and (2) a neutralized vowel space based on the corner vowels relative to children with SSD and TD.

1aSC11. Effect of nasendoscopy procedure on acoustic features of voice and speech. Liran Oren (Otolaryngol., Univ. of Cincinnati, University of Cincinnati, PO Box 670528, Cincinnati, OH 45267, orenl@ucmail.uc.edu)

Nasendoscopy procedure enables direct visualization of voice and speech anatomy. The procedure passes a flexible scope transnasally to observe velopharyngeal mechanism or the larynx during speech and phonation. This imaging technique is commonly used simultaneously with audio recordings. Studies have identified important statistically-significant relationships between imaging-based and acoustic measures related to voice and speech quality. However, the effect of the nasendoscopy procedure itself on specific acoustic features is rarely being considered. Nasendoscopy is likely to cause some speakers to deviate from their typical vocal function by inducing stress and tension during recordings. Furthermore, the flexible nasendoscope can also change the sound and airflow behavior inside the vocal tract. The current study attempts to quantify how nasendoscopy influences acoustic features using a physical model based on a child's vocal tract during a sibilant sound. An electrolarynx was used to provide a constant sound source with the input airflow to the model. Audio recordings were captured while retracting the nasendoscope from the level of the hypopharynx. Results show that compared to the same data taken without nasendoscopy, the scope can change measures of signal-to-noise and cepstral-peak-prominence by 15% and 7%, respectively. The nasendoscope had a minimal effect (<3%) on the SPL measure.

1aSC12. The time course of first formant movement in normal and loud speech. Laura Koenig (Commun. Sci. and Disord., Adelphi Univ., 300 George St., New Haven, NY 06511, koenig@haskins.yale.edu) and Susanne Fuchs (Leibniz-Zentrum für Allgemeine Sprachwissenschaft, Berlin, Germany)

Louder speech tends to be slower than typical speech. For some vowels (viz., low and lax ones), louder speech may also be associated with higher first formant frequencies (F1s) at vowel midpoint. Possibly, increased duration allows speakers more time to reach a lower jaw position, thus a higher value of F1, in loud speech. In earlier work, we found that some loudness differences appeared at 25% and 75% of vowel durations, i.e., they were seen for much of the vowel. The current study presents data from 11 German-speaking women producing naturally elicited typical and loud speech. Target words included high, low, tense, and lax vowels with surrounding consonantal place controlled. We will present full vowel trajectories, i.e., from preceding to following consonants, to assess the time course of F1 differences across vowel types and loudness conditions. F1 will also be considered within normalized time intervals to control for durational differences between conditions. We expect that F1 formant trajectories will show minimal loudness differences throughout the vowel for high tense vowels, but time-varying differences will be more apparent for low and lax vowels, and appear rather early in the vowel.

1aSC13. An acoustic analysis of French vowel phoneme substitutions in native English speakers. Madeline G. Strah-Farber (None, 1201 Davidson Dr. N15, Fort Collins, CO 80525, mgfarber1@gmail.com)

In the last 60 years of second language acquisition research, much has changed from the Critical Period Hypothesis (CPH) and the belief from Eric Lenneberg that producing second language vowels and consonants was too tricky without a first language accent (Harley, 1997). These casual remarks and observations would set in motion further extensions of CPH to help focus equal attention on second language learning. It would not be until 1987 and research conducted by Jim Flege that the world would begin to see how American college students can produce and differentiate between the close fronted /y/ and its back closed counterpart /u/ (Flege, 2005). Flege would conclude through his experiment that second language learners (L2) of French had an easier time producing a "new" vowel or vowel that is absent from their first language (L1) inventory because the habit of substitution would not be as frequent (Flege, 2005). The research that will be attempted will try to show how with more experience as a French student, students will be able to form a separate category for the high fronted close /y/, regardless of how close it is to its American English counterpart /u/, or the predictions set forth by the Learning Speech Model.

1aSC14. Acoustic effects of syllable position of /s/ in American English: Is there a "dark" [s]? Danilo A. Lombardo (Speech-Language-Hearing Sci., The Graduate Ctr., City Univ. of New York, 365 Fifth Ave. New York, NY 10016-4309, dlombardo1@gradcenter.cuny.edu), D. H. Whalen (Haskins Labs., New York, NY), Wei-Rong Chen, and Christine H. Shadle (Haskins Labs., New Haven, CT)

The effect of syllable position on the acoustic structure of speech sounds has been studied for different consonants. A well-known example is the lateral approximant /l/ in American English (AE), whose spectral properties vary between initial and final position, as characterized by the light–dark [l] distinction, as in [l]*eaf* versus *fee*[ĕ̈]. Spectral differences between initial and final /l/ have been widely reported, with final [ĕ̈] displaying a higher concentration of acoustic energy in the low-frequency range. /l/ shares articulatory properties with the alveolar fricative /s/, and both sounds display similar trajectories of diachronic change at initial versus final position. However, it is unknown whether syllable position modulates the acoustic properties of /s/ in a way similar to that observed for /l/. We explored the AE /s/ in initial and final position using sVC and CVs words with symmetrical contexts. Acoustic parameters were computed from multitaper spectra. Preliminary results show that the amplitude at higher frequencies is less intense in final than initial [s]. Moreover, the frequency of the main spectral peak was lower in final [s]. It appears that similarities between /s/ and /l/ extend to the effect of syllable position on spectral properties.

1aSC15. Information redundancy across spectral regions for sentence recognition in noise. Yi Shen (Speech and Hearing Sci., Univ. of Washington, 1417 NE 42nd St., Seattle, WA 98105-6246, shenyi@uw.edu)

Two approaches are examined to quantify the synergetic and redundant interactions across spectral regions for sentence recognition in noise. In one approach, the Speech Intelligibility Index is extended such that speech intelligibility is not only influenced by the speech audibility metrics in individual frequency bands but also their pairwise products to capture the synergetic and redundant interactions between pairs of bands. In a second approach, it is assumed that successful keyword recognition is determined by how many independent channels of information is received by the listener. Each channel of information may be uniquely carried by a frequency band, representing the independent contribution of that band, or shared among multiple bands, representing information redundancy. A previously collected dataset was re-analyzed using the two quantification approaches. For each of the 30 listeners in the dataset, correctness in keyword recognition in the IEEE sentences was available for 600 trials. From trial to trial, the sentence-in-noise stimuli varied in terms of signal-to-noise ratio and which frequency bands were removed from the spectrum via filtering. Both approaches provided consistent depictions of how information is shared across spectral regions for the IEEE sentences. Moreover, the information redundancies across frequency regions did not equally impact all listeners.

1aSC16. Perceived naturalness of typical disfluent speech and speech of adults who stutter. Ksenia Kampf (Linguist, Univ. of Oregon, 3655 W 13th Ave. Apt 354, Eugene, OR 97402, kseniag@uoregon.edu)

Disfluencies in spontaneous speech have been shown to be sensitive to the syntactic constituency (Levelt, 1989): production is restarted at the boundary nearest to the point of interruption. While there is a production-internal account of constituency structure in the speech of both adults who stutter (AWS) and typical speakers (AWNS), there is no evidence of constituency realization in the perceptional account of disfluent speech. The current study explores whether the constituency structure contributes to the perception of disfluent speech of AWS and AWNS. More specifically, the study investigates whether syntactic constituents are a cohesive unit of perception and whether altering the constituency structure impacts the perceived naturalness of disfluent speech. The participants rated the samples of disfluencies produced by AWS and AWNS for naturalness. The stimuli were presented in their original form as well as in a form with an artificially altered disfluency position. The results indicate that the location of the repetition disfluency is one of the correlates of speech naturalness, both in stuttered and typical speech. At the same time, speech naturalness ratings are minimally influenced by the type of speaker, suggesting that stuttering and typical disfluencies follow similar perceptual patterns.

1aSC17. Phonation in verbal tics and true words is distinguishable on the basis of voice quality. Mairym Llorens (Linguist, Univ. of Southern California, 3601 Watt Way, Grace Ford Salvatori 301, Los Angeles, CA 90089-1693, llorensm@usc.edu) and Louis Goldstein (Linguist, Univ. of Southern California, Los Angeles, CA)

Adults with Tourette syndrome produce vocal tics that mimic true words and phrases. These verbal tics often occur while the ticcer is actively engaged in speaking. By virtue of their isomorphism with true words, co-speech verbal tics can introduce unintended meanings into a talker's utterances. How does the ticcer-talker compensate for the presence of intrusive verbal tics that threaten to confuse their interlocutor? In the present study, the acoustics of phonation in naturalistic co-speech ticcing by four adult speakers of English was examined. Six acoustic parameters previously identified as indices of the distinction between falsetto and modal voice were extracted from ticced and spoken stressed vowel and whole-word intervals. For all participants, at least three out of the six markers of falsetto were observed in verbal tics and were absent in true words. Furthermore, transitions into and out of falsetto were concomitant with transitions across speech-tic and tic-speech junctures, respectively. These results suggest that phonation tasks for verbal tics are aimed at achieving falsetto mode of laryngeal vibration, in contrast to phonation in spoken English, which is largely modal. Findings are interpreted in terms of a discourse-level task that organizes ticced and spoken vocalizations in the service of clear communication.

1aSC18. Word and phrase duration in Mandarin-speaking individuals with Alzheimer's disease. Kaiwen Yu (Linguist, UBC, 5960 Student Union Boulvard, 05-308D, Vancouver, BC V6T 1Z1, Canada, yu_kaiwen2018@outlook.com), Yadong Liu, Arian Shamei, and Bryan Gick (Linguist, UBC, Vancouver, BC, Canada)

Alzheimer's disease (AD) is a progressive neurodegenerative disorder characterized by irreversible cognitive deterioration, often manifesting in pathological speech patterns [Baker *et al.*, *Clin. Linguist. Phon.* **21**(11–12), 859–867 (2007)]. Previous research has examined temporal and acoustic features of AD patients' speech, including pitch, volume, and voice quality, showing that these measures can be used to discriminate between people with AD and healthy older adults [Meilan *et al.*, *Dement. Geriatr. Cogn. Disord.* **37**(5–6), 327–334 (2014)]. However, whether word and phrase duration are affected by AD remains unclear. The present study measured word and phrase duration from nine Mandarin-speaking AD patients and nine gender-matched neurotypical controls undertaking picture description and naming tasks. Preliminary results show AD patients exhibited a significant difference ($t(1688.1) = -5.88$, $p < .001$) in word duration compared to controls, with shorter word duration in the naming task ($t(1088.8) = -8.66$, $p < .001$) and longer in the picture description task ($t(484.13) = 2.40$, $p = 0.017$); AD patients also uttered significantly shorter phrases than controls in the picture description task ($t(86.01) = -2.17$, $p = 0.033$). The preliminary result suggests that AD may affect word and phrase duration. [Work supported by NIH and NSERC].

Session 1aUW

Underwater Acoustics and Acoustical Oceanography: Understanding and Representing Uncertainty in Underwater Acoustic Models I

Sean Pecknold, Cochair
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Martin Siderius, Cochair
Portland State Univ., 1600 SW 4th Avenue, Suite 260, Portland, OR 97201

Chair's Introduction—8:35

Invited Paper

8:40

1aUW1. Towards a process description of acoustic uncertainty in the upper ocean. John A. Colosi (Dept. of Oceanogr., Naval Postgrad. School, Monterey, CA 93943, jacolosi@nps.edu) and Jennifer MacKinnon (Scripps Inst. of Oceanogr., UCSD, San Diego, CA)

The upper ocean is a region that hosts the worlds largest biomass, is the boundary layer mediating the exchange of momentum, energy, heat, gasses, etc. between the air and sea, and it is a region of critical Naval tactical importance. Acoustic modalities are important tools for sensing this region as evidenced by the ground-breaking results of bio-acousticans monitoring whales, fish, zooplankton, and other marine organisms. But while the significance of upper ocean acoustics has been long appreciated, the understanding of key ocean acoustic processes is lacking. Here, we define upper ocean as the region of the water column where the impacts of winter mixing are evident, usually two to three times the depth of the maximum mixed layer. The classical view of the thermohaline structure posits a combination of mostly one-dimensional mixing processes, slow mesoscale stirring and advection, and subsequent subduction towards the main thermocline. However, a number of recent studies points to an important role played by three-dimensional, small-scale, rapidly evolving, submesoscale fronts and their potential interaction with internal waves. A few observational highlights will be reviewed with an emphasis on sound-speed and spice distributions, and implications for deterministic and stochastic sound propagation effects will be discussed.

Contributed Paper

9:00

1aUW2. Influence of environmental perturbations on acoustic signals in the 2021 New England Shelf Bread Acoustics experiment. Brendan J. DeCourcy (Appl. Ocean Phys. & Eng., Woods Hole Oceanographic Inst., 86 Water St., Falmouth, MA 02543, bdecourcy@whoi.edu), Ying-Tsong Lin (Appl. Ocean Phys. & Eng., Woods Hole Oceanographic Inst., Woods Hole, MA), Glen Gawarkiewicz (Physical Oceanogr., Woods Hole Oceanographic Inst., Woods Hole, MA), Jennifer Johnson, and Weifeng G. Zhang (Appl. Ocean Phys. & Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

During the 2021 New England Shelf Break Acoustics (NESBA) experiment, real-time hydrodynamic and acoustic models deployed on shipboard high-performance computers (HPC) predicted acoustic signals sent and received within the NESBA mooring network. The hydrodynamic model incorporated field measurements of the physical environment in real time. While the resulting comparisons between models and received data produced similar patterns for strong arrivals, the cause for variation in weaker arrivals by detailed physical oceanographic variability is not trivial to deduce. In this presentation, the influence of perturbations to sub-bottom properties and small-scale water column variability on signals in the NESBA acoustic model is explored using sensitivity kernels. The predicted acoustic signal uncertainty will be further compared with measured variability to evaluate the model skill to capture acoustic fluctuation in the varying shelf break environment. [Work supported by the Office of Naval Research.]

Invited Papers

9:15

1aUW3. Coupled modeling of the seasonal transmission loss in the Beaufort Sea. David R. Barclay (Dept of Oceanogr., Dalhousie Univ., 1355 Oxford St., LSC Bldg., Halifax, NS B3H 4R2, Canada, dbarcl@gmail.com), S. B. Martin (Halifax, JASCO Appl. Sci., Dartmouth, NS, Canada), Paul C. Hines (Hines S&T, Halifax, NS, Canada), Pablo Borys, Calder L. Robinson (Halifax, JASCO Appl. Sci., Dartmouth, NS, Canada), James Hamilton (None, Halifax, NS, Canada), and Terry Deveau (Halifax, JASCO Appl. Sci., Halifax, NS, Canada)

An eight-element drifting vertical line array was deployed overwinter 2019–20 in the Beaufort Sea to record transmissions from two moored 35 Hz sources deployed as part of the Coordinated Arctic Acoustic Thermometry Experiment (CAATEX), and seven 925 Hz sources deployed by the Arctic Mobile Observing Systems (AMOS) experiment. Transmissions were received every day on the array at ranges of 10 to 850 km. A probabilistic rough-ice canopy was integrated to a Parabolic Equation code to predict the propagation loss. Sound speed profiles and percent ice cover along the propagation paths were obtained at daily resolution from data-assimilative Global Ice Ocean Prediction System (GIOPS, 20 km spatial resolution). Sea ice roughness and keel properties are empirically determined from the GIOPS ice thickness. The ice roughness is generated using a random pulse-train model with a characteristic wavenumber spectrum slope determined by ice age. Empirical probability distributions of ice keel depth, slope, and spatial densities are used to randomly add keels. Spatial averaging is used to account for signal bandwidth and uncertainty in receiver positions. The environment files and modeling are performed in piece-wise steps to manage memory footprints over long ranges. Modeled propagation losses compare well with the data and reproduce the seasonal variability observed in the measured results.

9:35

1aUW4. Spatial, temporal, and spectral variability of ambient noise in the Arctic. Calder L. Robinson (Halifax, JASCO Appl. Sci., 202-32 Troop Ave., Dartmouth, NS B3B 1Z1, Canada, calder.robinson@jasco.com), Paul C. Hines (ECE, Dalhousie, Halifax, NS, Canada), S. B. Martin (Halifax, JASCO Appl. Sci., Dartmouth, NS, Canada), and David R. Barclay (Dept. of Oceanogr., Dalhousie Univ., Halifax, NS, Canada)

Ambient noise in the Arctic exhibits significant spatial, temporal and spectral variability. This poses significant challenges for passive sonar performance estimation since the probability of detection is measured against this noise background. This is exacerbated by the relative paucity of measurements made in this remote region with which to estimate the statistics that describe the noise variability. To help address this deficiency, ambient noise measurements were collected during a year-long deployment for the CAATEX trial conducted in 2020–2021. In this paper, the measurements are summarized and the spatial, temporal and spectral variability of the ambient noise is presented.

9:55

1aUW5. Modeling the effects of internal waves on synthetic aperture sonar resolution. Nicholas A. La Manna (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, 50 Pointe PL, apt 413, Dover, NH 03820, nlamanna@ccom.unh.edu), Anthony P. Lyons (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, Durham, NH), Daniel C. Brown (Penn State Univ., State College, PA), and Roy E. Hansen (Dept. of Informatics, Univ. of Oslo, Kjeller, Norway)

Synthetic aperture sonar (SAS) beamforming is performed under the assumption of a constant sound speed within the water column or by utilizing a measured sound speed profile. Water column features, such as internal waves or internal wave boluses that cause a sound speed structure different than that assumed, can break the constant sound speed assumption resulting in image defocusing and loss of resolution. The resultant loss of resolution can be quantified using the point spread function (PSF). An ellipse fitting technique was applied to SAS images collected by the Norwegian Defence Research Establishment (FFI) using the HISAS system. In conjunction with the analysis conducted on real data, a ray tracing method was applied to model the broadening effects on the PSF for sound speed differences caused by internal wave boluses. A wavenumber beamforming algorithm was also applied to artificial echo data, in which sound speed errors were introduced. It is hoped that the two models can be used in conjunction with SAS data to determine the sound speed within an internal wave feature from the observed loss in resolution.

10:15–10:30 Break

Contributed Papers

10:30

1aUW6. The Beaufort Sea acoustic duct's variability and its impact on acoustic propagation using the mode interaction parameter. Murat Kucukosmanoglu (Ocean Sci., Univ. of California, Santa Cruz, 604 Koshland Way, Santa Cruz, CA 95064, mkucukos@ucsc.edu), John A. Colosi, Christopher W. Miller (Oceanogr., Naval Postgrad. School, Monterey, CA), Peter F. Worcester, and Matthew A. Dzieciuch (Scripps Inst. of Oceanogr., Univ. of California, San Diego, San Diego, CA)

The Beaufort duct is a subsurface sound channel formed by cold Pacific Winter Water sandwiched between warmer Pacific Summer Water and Atlantic Water in the Western Arctic Ocean. This duct traps sound waves and allows them to travel long distances without losing energy to lossy interactions with sea ice and surface waves. This study quantifies Beaufort duct variability based on Canada Basin Acoustic Propagation Experiment (CANAPE) and Coordinated Arctic Acoustic Thermometry Experiment (CAATEX) oceanographic observations. Deterministic ocean features induce coupling between acoustic modes confined to the Beaufort duct and non-ducted modes by weakening the duct or causing it to take on an asymmetric form. A non-dimensional mode interaction parameter (MIP) can be defined using the acoustic frequency and the vertical and horizontal scales of sound speed perturbation to characterize coupling strength. It identifies

three important wave propagation regimes: sudden approximation, adiabatic approximation, and maximum interaction regime (Colosi and Zinicola-Lapin, 2021). When the MIP between ducted and lossy modes is more and less than 1, strong and weak acoustic variability is predicted, respectively. Variability is high when the MIP between two ducted modes surpasses 1, but modei–mode interference patterns grow more complicated. Acoustic numerical simulations are used to demonstrate various effects.

10:45

1aUW7. Time frequency investigation and representation of acoustic backscatter signatures. Wendy N. Newcomb (GTRI, Georgia Tech, 7220 Richardson Rd., CCRF 5-145, Smyrna, GA 30080, wendy.newcomb@gtri.gatech.edu)

Observation of an object's structural acoustic response, considered in the aspect-dependent frequency response domain, is a data representation used for characterizing the target strength of an object. Geometric and elastic wave features in the angle-frequency domain are difficult to distinguish. Time varying characteristics such as attenuation are lost with a single frequency transform across the entire time collection of the acoustic return. Research in the aspect angle-frequency domain primarily applies the fast Fourier transform to an entire time series collection for a single ping cycle. This work aims to extend the dimensional space of analysis for the acoustic backscatter representation by investigating time-frequency resolution across aspect angle for acoustic elastic responses of an object. Empirical analysis of the data is presented with application of various time frequency transforms. These methods are assessed using time frequency localization metrics to quantify their performance with the AirSAS dataset. This exploratory work will develop the intuition regarding how the time frequency distribution can be localized, measured, and vary among objects of differing compositions. The ways in which time frequency localization can support distinguishing features of the acoustic return beyond the traditional aspect angle-frequency domain will be discussed as well.

11:00

1aUW8. Viability of the mixed layer duct with observed dynamics. Edward Richards (Ocean Sci., UC Santa Cruz, Ocean Sci., 1156 High St., Santa Cruz, CA 95064, edwardlrichards@gmail.com) and John A. Colosi (Oceanogr., Naval Postgrad. School, Monterey, CA)

While the upper ocean of the north Pacific is largely mixed due to air-sea interaction, a 1000 km sea-soar transect shows the existence of stratification in the mixed layer that persists over many tens of kilometers. The effects of this stratification on mixed layer acoustic propagation depends on the sound speed differences across the different density layers, and observations show the sound speed varies rapidly both between and along the isopycnals. Acoustic models are used to separately quantify the importance of the mixed layer stratification and the density compensated sound speed variability on propagation in the mixed layer duct. The observed sound speed field is modeled as the superposition of three fields: a smooth background, the stratification field with slow isopycnal sound speed variability, and the remnant unstratified sound speed variation. The sound speed variation in the stratified mixed layer is attributed to internal waves and eddy filaments and in the unstratified field to ocean spice. Both types of observed sound speed variation significantly change, or completely block, mixed layer propagation. Examples are presented of blocking features in the tilt or spice fields separately, and where they exist only in the superposition of the two fields.

11:15

1aUW9. Environmental characterization from passive acoustic data collected on a partially-spanning array. Franklin H. Akins (Scripps, UCSD, 562 Arenas St., La Jolla, CA 92037, fakins@ucsd.edu) and William Kuperman (Scripps, UCSD, La Jolla, CA)

We show that with knowledge of the sound speed along a receiving, partially water-column-spanning vertical line array, mode wavenumber and geoacoustic estimates can be obtained without requiring knowledge of source location or range-rate. This estimation is accomplished simply by applying spatial smoothing and MUSIC to the sample covariance matrix. The method is demonstrated with simulation and subsequently on SWellEx-96 experimental data.

11:30

1aUW10. The holographic signal processing for the estimation of sound mode parameters in shallow water. Sergey A. Pereselkov (Mathematical Phys. and Information Technol., Voronezh State Univ., Russia, Voronezh, Universitetskay pl, 1, Voronezh 394018, Russian Federation, pereselkov@yandex.ru), Venedikt Kuz'kin (Hydrophysics Lab., General Phys. Inst. RAS, Moscow, Russian Federation, Moscow, Russian Federation), Elena Kaznacheeva, and Ilya Kaznacheev (Mathematical Phys. and Information Technol., Voronezh State Univ., Voronezh / Russia, Russian Federation)

The holographic method for estimation of the broadband sound field modes parameters by using a single receiver in shallow water is presented in the paper. The holographic method of signal processing is based on the two-dimensional Fourier transform (2D-FT) of the source moving in the shallow water waveguide. The sound field creates in waveguide a stable interference pattern (interferogram) in the frequency-time domain. The result of the 2D-FT of the interferogram is called the Fourier-hologram (hologram). It is shown in the paper that different sound field modes create different focal spots in the hologram domain. The filtering focal spots and applying the inverse 2D-FT to them allows restoring the field of the selected mode. In relation to the time-warping operator, the proposed method of signal processing provides a smaller error in the mode group velocity estimation. The results of the numerical simulation as well as the sound mode parameters estimation are presented and discussed in the paper. [Work supported by the RFBR (19-29-06075) and MK-6144.2021.4.]

11:45

1aUW11. Model validation for simulated synthetic aperture sonar time series data. Jason Philtron (Penn State Univ., 201 Old Main, University Park, PA 16802, jhb186@psu.edu) and Brian Reinhardt (Penn State Univ., University Park, PA)

Synthetic aperture sonar (SAS) systems have grown in popularity due to their ability to create high-resolution sonar imagery with a high area coverage rate. Researchers continue to find new uses for SAS data products. To support the development of new uses and algorithms, it is important to be able to accurately simulate SAS systems with synthetic time series data. This simulated data can be used for many purposes, including estimating system signal-to-noise ratio and performance in a variety of environments, testing signal processing algorithms, and training algorithms to recognize objects in SAS imagery. For these processing tasks to be successful, the simulator must accurately represent physical phenomena in the oceanic underwater environment, which can be extremely complex. This presentation will outline a process for validation and verification of a high-frequency SAS time series simulator, and discuss several useful tests for validating model fidelity. Two fundamental concepts will be explored in depth: the sonar equation and spatial coherence theory. The sonar equation is important because it accounts for correct implementation of multiple lower level models such as geometry, beam patterns, propagation, and seafloor scattering strength. Spatial coherence is important because it is relied on to form coherent imagery.

Session 1pAA

Architectural Acoustics, Noise, Speech Communication, and Psychological and Physiological Acoustics: Balancing Speech Intelligibility with Privacy for Indoor Spaces

Joonhee Lee, Cochair
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Kenneth W. Good, Cochair
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Roderick Mackenzie, Cochair
Soft Db, Montreal, Canada

Chair's Introduction—1:30

Invited Papers

1:35

1pAA1. Optimization of a partition's ASTC to reach speech privacy goals, including innovative construction administration methods. Bonnie Schnitta (SoundSense, LLC, 39 Industrial Rd., Unit 6, PO Box1360, Wainscott, NY 11975, bonnie@soundsense.com) and Sean Harkin (SoundSense, LLC, Wainscott, NY)

In order to achieve speech privacy in a room, it must be accurately engineered from the partitions, such as the walls, to the door. The speech privacy specification results from calculations based on the partitions' STC. It is referenced by most building codes that these partitions, when built outside of the controlled lab, have an ASTC that is 5 points lower than the STC. When the ASTC is more than 5 points lower, strict speech privacy requirements are not met. This presentation discusses flanking and acoustic leakage points which degrade the wall by more than 5 points. Methodologies for resolving these issues of flanking or acoustic leakage will be presented that help bring the ASTC close to the STC and thereby meet the goals for speech privacy. A further discussion of innovative methods of construction administration to achieve these goals will also be discussed

1:55

1pAA2. A systematic investigation to assess the merits of measurement preconditions critical to speech privacy standards. Viken Koukounian (K.R. Moeller Assoc. Ltd., 3-1050 Pachino Court, Burlington, ON L7L 6B9, Canada, viken@logison.com)

As with any field of science, the limits of theory are regularly tested to make way for more reliable testing standards. However, without a defined understanding of scope and purpose, the line between theory and practice—that which is already obscure—becomes increasingly difficult to identify. In the following investigation, the author reviews the scope and purpose of certain acoustical standards (ASTM E90, E336, E1130, E1573, E2638, and ANSI/ASA S12.2, S12.72) to clarify the context for which meaningful guidance is needed. More specifically, topics of interest include (1) effects of measurement locations and positions, (2) effects of measurement procedure, (3) effects of measurement instrumentation, (4) differences between operators, and (5) effects of architectural features on sound (i.e., spatial, spectral and temporal properties). To offer resolute understanding of the impact of each recommendation, the measurement methods within this investigation exceed those provided by the Standards. This provision is intended to afford meaningful insight as to the theoretical and practical limitations of the measurement of sound in the built environment—in contrast to the more limited data that would be available if testing is in accordance with the relevant Standards. The variability due to each guideline is quantified and reported.

Contributed Papers

2:15

1pAA3. Comparing testing methodologies of speech privacy class in closed offices. Rewan Toubar (Dept. of Bldg., Civil and Environ. Eng., Concordia Univ., 1515 Saint-Catherine St. W, EV-0S3.412, Montreal, QC H3G 1S6, Canada, rewantoubar@yahoo.com), Roderick Mackenzie (Soft dB, Montreal, QC, Canada), and Joonhee Lee (Dept. of Bldg., Civil and Environ. Eng., Concordia Univ., Montreal, QC, Canada)

Speech privacy is a principal research area in speech communication. The term speech privacy is generally interpreted a condition where speech cannot be readily understood by, but may be audible to, an unintended listener. The need to prevent sound from intruding into adjacent spaces in both closed and open-plan settings is a concern in various office buildings. Speech privacy is essentially a function of the signal-to-noise ratio, comprising the noise reduction between source and receiver positions, and the masking effective of background noise. Speech Privacy Class (SPC) is one of the commonly used metrics for speech privacy assessment in closed plan offices in North America. However, there is a reluctance to use SPC between closed rooms due to laborious testing requirements as per ASTM E2638; there is a supposed preference to use either Speech Privacy Potential (SPP) based on simpler NIC testing, or to use Articulation Index (AI) despite the AI being created solely for testing open-plan settings. The ASTM E2638-10 standard SPC testing procedure does not assume a diffuse field in the receiving space but rather evaluates the performance at each potential eavesdropping location on the basis that there may be an intentional listener. Thus, to better apply the SPC to unintended listening and speech privacy in typical commercial spaces, the goal is to compare the current methodology as per ASTM E2638-10 with alternative sampling methods considering the talker location and unintended listening positions outside the source room.

2:30

1pAA4. Categorizing auditory objects of perceived sounds in the built environment. Semiha Yilmazer (Ray W. Herrick Labs., Purdue Univ., Ray W Herrick Lab 177, S Russell St., West Lafayette, IN 47907, syilmaze@purdue.edu), Volkan Acun, Donya Dalirnaghadeh, Ela Fasllija, Zekiye Şahin, and Elif Mercan (Interior Architecture and Environ. Design, Bilkent Univ., Ankara, Cankaya, Turkey)

This exploratory study focuses on a cognitive approach to categorize complex auditory scenes in the built environment, while prior studies have concentrated on outdoor acoustic environments. Six experts and 30 non-experts performed a free chip sorting task to assess 70 binaural recordings taken from indoor spaces. Healthcare, working, cultural, educational, leisure, worship, and transportation spaces (e.g., bus, train, metro stations, and airports) were chosen as public spaces. The participants were asked to classify the auditory objects into sound categories based on the descriptive labels provided by the authors to identify the sound sources. The findings, obtained through hierarchical agglomerative clustering and non-metric multidimensional scaling (MDS), show that there are three prominent category labels regarding perceived sound in the indoor acoustic environment: (1) intelligible and unintelligible speech; (2) periodic and transient sounds; and (3) stationary and non-stationary sounds. Human-generated sounds such as conversation, laughter, footsteps, and coughing vary over time according to the context of the built environment. Moreover, technology-related sounds, such as mechanical and electronic ones, have a deterministic and random nature that differ according to the function of the spaces.

Session 1pBA

Biomedical Acoustics, Physical Acoustics, and Signal Processing in Acoustics: Emerging Techniques Involving Super-Resolution Imaging Including Image Processing Methods, Applications, and Instrumentation

Jeffrey Ketterling, Cochair
Riverside Research, 156 William St, 9th FL, New York, NY 10038

Michael L. Oelze, Cochair
ECE, University of Illinois at Urbana-Champaign, 405 N Mathews, Urbana, IL 61801

Chair's Introduction—1:00

Invited Papers

1:05

1pBA1. Principles and techniques of microbubble localization and tracking in super-resolution ultrasound microvascular imaging. Pengfei Song (Elec. and Comput. Eng., Univ. of Illinois Urbana-Champaign, 405 N. Mathews Ave., Beckman Inst. 4041, Urbana, IL 61801, songp@illinois.edu)

Super-resolution ultrasound microvascular imaging (SR-UMI) is an emerging technology that uses microbubbles as point targets to break the diffraction limit of ultrasound and track blood flow. Robust microbubble localization and tracking are essential for successful SR-UMI. However, in practice, microbubble localization and tracking are often hampered by the complex flow dynamics of microbubbles (e.g., microbubble signal overlap, merging and splitting) and suboptimal ultrasound imaging quality (e.g., low SNR, tissue motion). More importantly, the tradeoff between microbubble localization and tracking accuracy and SR-UMI imaging speed remains a key barrier for practical implementation of SR-UMI. In this presentation, I will first review the principles and state-of-the-art techniques of microbubble localization and tracking. I will then introduce several new methods recently developed by our group that demonstrate improved microbubble localization and tracking performance under high microbubble concentrations (which leads to accelerated SR-UMI). Finally, I will introduce a new deep learning-based SR-UMI method that is both microbubble localization- and tracking-free. Because this new method is no longer constrained by challenges associated with microbubble localization and tracking, it achieves very fast SR-UMI imaging speed (tens of milliseconds to several seconds on a moderate GPU) and has the potential to be implemented in real-time.

1:25

1pBA2. Superresolution imaging of microvasculature at the capillary level using nanobubble contrast agents. Agata A. Exner (Dept. of Radiology, Case Western Reserve Univ., Case Western reserve University, Cleveland, OH 44106-7078, agata.exner@case.edu), Dana Wegierak (Dept. of Biomedical Eng., Case Western Reserve Univ., Cleveland, OH), and Mahdi Bayat (Dept. of Elec. Eng. and Comput. Sci., Case Western Reserve Univ., Cleveland, OH)

In this work, we describe applications of nanobubbles in superresolution imaging. Conventional ultrasound localization microscopy (ULM) techniques require sparse scatterers in the form of echogenic microbubbles (MB, $\sim$1–10 μm diameter). ULM enables imaging of capillaries larger than $\sim$50 μm in diameter but becomes too time consuming in smaller capillaries due to insufficient MB density. A viable alternative for microvascular imaging may be the use of submicron bubbles, or nanobubbles (NBs). NBs are $\sim$100i–500 nm in diameter and have a particle density 5 orders of magnitude higher than MBs without associated attenuation artifacts. In addition to strong nonlinear response, under some conditions NBs can produce SNR linear signals stronger than erythrocytes. This allows for formulating a superresolution imaging problem using computational approaches suitable for dense sources instead of relying on physical sparsity. With much smaller sizes compared to MBs, NBs can access the smallest capillaries at much higher concentrations ($\sim$$10^{11}$), offering enhanced detection and faster superresolution acquisition. In addition, NB nonlinear signals can be decomposed to elucidate intravascular and extravasation into tissues with vascular hyperpermeability, such as tumors. Thus, when combined with appropriate processing, NBs can provide information about vascular architecture and extravascular transport which is not achievable with MBs.

1:45

1pBA3. Visualization of tumor angiogenesis using super-resolution ultrasound imaging. Katherine Brown (Bioengineering, Univ. of Texas at Dallas, Richardson, TX) and Kenneth Hoyt (Bioengineering, Univ. of Texas at Dallas, 800 W Campbell Rd. BSB 13.929, Richardson, TX 75080, kenneth.hoyt@utdallas.edu)

Super-resolution ultrasound imaging (SR-US) is a new technique that breaks the diffraction limit and allows visualization of microvascular structures down to tens of microns. The objective of this presentation is to briefly introduce SR-US imaging and use in various small animal models. Engineering challenges and ongoing opportunities will also be introduced. SR-US is predominantly a software advance that can be integrated with both clinical and preclinical US imaging platforms. After injection of an intravascular microbubble (MB) contrast agent, a time-series of contrast-enhanced ultrasound (CEUS) images are acquired. Spatiotemporal filtering of the CEUS images helps remove the tissue clutter signal and segmentation of the MB signal. Precise localization of the center of each agent is then determined. This process is repeated for each CEUS image in the time series. The resultant SR-US images depict fine microvascular detail typically unresolved using traditional CEUS imaging methods. During this presentation, we will explore use of methods to help improve the MB detection and localization process in space and time like with the use of motion correction and machine learning algorithms. We will also present a series of preclinical examples highlighting the potential of this new technology in cancer research. Results demonstrate that SR-US imaging depicts microvascular structures not visible using traditional CEUS methods. The utility of this new diagnostic tool will play more of a role in theranostics over the next decade.

2:05–2:20 Break

2:20

1pBA4. Super-resolution imaging using conventional and non-conventional beamforming. Francisco Santibanez, Thomas Kierski, Ryan Deruiter, Rebecca Jones, Danai Soulioti, Jake McCall, Hatim Belgharbi, Davis Crews (Biomedical Eng., Univ. of North Carolina at Chapel Hill and North Carolina State Univ., Chapel Hill, NC), Paul A. Dayton (Biomedical Eng., UNC Chapel Hill, Chapel Hill, NC), and Gianmarco Pinton (Biomedical Eng., Univ. of North Carolina at Chapel Hill and North Carolina State Univ., 116 Manning Dr., Mary Ellen Jones Rm. 9212A, Chapel Hill, NC 27599, gia@email.unc.edu)

Super-resolution ultrasound surpasses diffraction limits by localizing spatio-temporally separable contrast agents and generates images that significantly exceed the resolution of conventional B-mode imaging methods. However, the ability to detect contrast agents and to separate their signal from the underlying tissue and from sources of image degradation such as phase aberration or reverberation clutter remains a process that is governed by fundamental wave propagation and beamforming. Here, super-resolution imaging and improvements in contrast detection, imaging depth, resolution, and registration accuracy are demonstrated using conventional and non-conventional beamforming methods in 2D and 3D. Three different imaging schemes: (a) single plane-wave, (b) three steered plane-wave compounding, and (c) 256 focused transmits are compared *in vivo* to quantify the improvements in contrast detection. Wide-beam 3D transcranial super-resolution and power Doppler images through a human and macaque skull are demonstrated using a 1.5 MHz sparse matrix array. These partially and fully focused methods are also demonstrated transcranially in rodents using 2D imaging at 15 MHz and volumetric imaging at 8 MHz. Finally super-harmonic super-resolution imaging approaches are demonstrated for stationary and moving bubbles in murine tumors. These imaging methods extend the capabilities of super-resolution imaging and may improve the clinical translatability of the technique.

2:40

1pBA5. Super-resolution Imaging with a multifrequency large-aperture hemispherical phased array. Lulu Deng (Physical Sci. Platform, Sunnybrook Res. Inst., 2075 Bayview Ave., Toronto, ON M4N 3M5, Canada, lldeng@sri.utoronto.ca), Harriet Lea-Banks, Ryan Jones, Meaghan O'Reilly, and Kullervo Hynynen (Physical Sci. Platform, Sunnybrook Res. Inst., Toronto, ON, Canada)

High-resolution imaging of microvasculature is important for diagnostic and therapeutic applications. Here, the utility of a multi-frequency large-aperture hemispherical phased array in the context of vasculature imaging via passive beamforming and super-resolution techniques was explored at various receive frequencies, to investigate the capability of mapping the emissions of lipid coated phase-change nanodroplets flowing through tube phantoms as well as rabbit renal and cerebral vasculatures *in-vivo*. With super-resolution techniques, the mean lateral (axial) full-width-at-half-maximum image intensity was estimated to be 16 ± 3 (32 ± 6) μm and 7 ± 1 (15 ± 2) μm, corresponding to approximately 1/85 of normal resolution when received with the subarrays at 612 and 1224 kHz, respectively. The mean positional uncertainties were approximately 1/350 (lateral) and 1/180 (axial) of the receive wavelength in water. The renal and cerebral vasculatures of rabbits were partially visualized. A correlation between transmit pulse and nanodroplet vaporization is observed, demonstrating potentials for nanodroplet vaporization modulation and signal-to-noise ratio enhancement. This study demonstrates the feasibility of mapping vaporized nanodroplets with passive beamforming and super-resolution imaging techniques. It may provide means to map complete vasculature pre- or post-treatment with therapeutic focused ultrasound in organs that can be surrounded by the hemispherical array.

3:00

1pBA6. Preclinical and clinical evaluation of super-resolution ultrasound imaging. Kang Kim (Bioengineering, Univ. of Pittsburgh, 623A Scaife, 3550 Terrace St., Pittsburgh, PA 15213, kangkim@upmc.edu), Qiyang Chen, and Roderick Tan (Univ. of Pittsburgh, Pittsburgh, PA)

Super-resolution ultrasound (SRU) imaging is an emerging technology that can visualize microvessels with unprecedented high spatial resolution compared to conventional contrast enhanced ultrasound imaging methods. Microvascular rarefaction occurs in both acute kidney injury (AKI) and chronic kidney disease (CKD) and leads to progressive injury, but no technology currently exists for safe, noninvasive assessment of kidney microvessels. In a preclinical mouse SRU study, we utilized a high-frequency small animal transducer to demonstrate reduction in microvascular density and increase in vessel tortuosity in a model of AKI due to ischemia-reperfusion injury. To translate SRU into clinical applications on human subjects, a fast SRU algorithm was implemented on a clinical curved array transducer for a deeper imaging depth and a larger imaging field of view and applied on healthy and CKD human subjects under an approved IRB. SRU did demonstrate reduced vascular density and blood flow in CKD patients compared to controls. SRU is thereby a novel and promising diagnostic and prognostic tool for kidney disease. Some encouraging pilot data and technical limitations will be discussed.

Session 1pCAa

Computational Acoustics, Biomedical Acoustics, Physical Acoustics, Underwater Acoustics, Structural Acoustics and Vibration, Noise, Musical Acoustics, and Architectural Acoustics: Finite and Boundary Element Methods Across Acoustics

Jennifer Cooper, Cochair
Johns Hopkins University Applied Physics Laboratory, 11100 Johns Hopkins Rd., Mailstop 8-220, Laurel, MD 20723

Michelle E. Swearingen, Cochair
Construction Engineering Research Laboratory, US Army ERDC, PO Box 9005, Champaign, IL 61826

Subha Maruvada, Cochair
U.S. Food and Drug Administration, 10903 New Hampshire Ave., Bldg. WO 62-2222, Silver Spring, MD 20993

Invited Papers

1:00

1pCAa1. Influence of the model parameters for the finite element simulation of bone conduction in the human head. Simon Kersten (Inst. for Hearing Technol. and Acoust., RWTH Aachen Univ., Kopernikusstraße 5, Aachen 52074, Germany, simon.kersten@akustik.rwth-aachen.de) and Michael Vorlaender (Inst. for Hearing Technol. and Acoust., RWTH Aachen Univ., Aachen, Germany)

Bone conduction (BC) describes the transmission of vibrations via osseous, cartilage, and soft tissue pathways that contribute to the hearing sensation. It is important, e.g., in the context of BC hearing aids as well as for the occlusion effect and one's own voice perception when using hearing protection or hearing devices. Finite element (FE) simulations of BC allow to overcome practical and ethical limitations of measurements with real heads and help to gain insights into underlying mechanisms. However, FE models inevitably require substantial simplifications due to the assignment and definition of volumetric domains and corresponding materials. Additionally, the material parameters are often only known within certain ranges. In this presentation, an overview on FE models of human heads used for BC research is given, and similarities and differences are discussed. The influence of the FE-inherent simplifications and the challenges in determining the model parameters are illustrated by varying the geometry and the material properties of a simple ellipsoidal model. The variability of the model predictions is compared to simulation results obtained with more realistic geometries and to the inter-subject variability observed in measurements with real human heads reported in the literature.

1:20

1pCAa2. 3D finite element modeling techniques and application to underwater target scattering. Aaron Gunderson (Appl. Res. Labs., The Univ. of Texas at Austin, 10,000 Burnet Rd., Austin, TX 78758, aarong@arlut.utexas.edu)

Underwater acoustic target scattering measurements rely on high-fidelity modeling for experimental comparison and understanding. Three-dimensional (3D) finite element models are well suited for this purpose, as they can account for arbitrary or unknown target properties and configurations/orientations within complex and asymmetrical seafloor environments. High acoustic frequencies and large physical distances associated with *in situ* scattering measurements pose challenges to 3D modeling efforts in terms of model sizes and runtimes. Certain model considerations must be made to keep the 3D model computationally efficient, yet accurate in predictive capability. Numerically determined Green's functions are demonstrated to permit 3D model reduction, while still preserving far-field scattering prediction capability through the Helmholtz–Kirchhoff integral. By determining Green's functions within the model, they need not be known or estimated for complex ocean environments a priori. Nontraditional scattering formulations and a survey of boundary truncation methods also are explored and implemented for maximal accuracy within small 3D computational domains. Model results for canonical elastic targets within varying seafloor environments are shown and compared to theory and experimentation. [Work supported by the Strategic Environmental Research and Development Program and by the Office of Naval Research, Ocean Acoustics.]

Contributed Papers

1:40

1pCAa3. Time-domain simulation of acoustic scattering and internal propagation from multiple gas bubbles. Jiacheng Hou (Mech. and Aerosp. Eng., Utah State Univ., 4130 Old Main Hill, Logan, UT 84322-4130, jiachenghou@aggiemail.usu.edu), Zhongquan Charlie Zheng (Mech. and Aerosp. Eng., Utah State Univ., Logan, UT), and John S. Allen (Dept. of Mech. Eng., Univ. of Hawai'i at Mānoa, Honolulu, HI)

Acoustic scattering and resonance responses from multiple gas bubbles are computed using a time-domain simulation based on the numerical solutions of the conservation laws. The time histories of scattered pressure and velocity, both outside and inside the bubbles, are obtained simultaneously from an immersed-boundary method facilitating the investigation of both exterior and interior acoustic fields for non-spherical bubbles. Agreement is found with both analytical scattering solutions and those from empirical shape factors in limiting cases. In addition, the time-domain method allows for the study of the transient acoustic scattering from pulse forcing (Gaussian). In this case, the interior gas oscillates in an off-center, non-uniform manner compared to the steady state forcing cases. The time history and the interior gas behavior are presented in detail for combinations of two bubbles of different sizes and shapes (spherical, prolate, oblate). A comparison and analysis of this immersed-boundary method to the finite element method and boundary element methods is provided. Also, the increasingly important role of computational acoustics in the study scattering of multiple targets of diverse sizes and shapes is highlighted.

1:55

1pCAa4. Simulation of acoustic liner performance in a grazing flow duct. Zhongquan Charlie Zheng (Mech. and Aerosp. Eng., Utah State Univ., 4130 Old Main Hill, Mech. and Aerosp. Eng., Logan, UT 84322, zzheng@usu.edu), Tayanne Alves (Aerosp. Eng., Univ. of Kansas, Lawrence, KS), Jiacheng Hou (Mech. and Aerosp. Eng., Utah State Univ., Logan, UT), and Huixuan Wu (Aerosp. Eng., Univ. of Kansas, Lawrence, KS)

Acoustic liners are commonly used to reduce aircraft engine noise. The performance of acoustic liners can be tested in a grazing flow duct. In order to simulate the new extended-reaction liners which allow lateral acoustic propagation inside the liner, finite element methods have been used in the frequency domain computation. In this study, different liner core materials, including honeycomb and foam-metal, are simulated with uniform or varied material properties and geometries. Frequency responses of liners are studied with different materials under several flow conditions. An immersed-boundary, time-domain simulation is also implemented to simulate the broad-band behavior of the liners and to compare with the frequency-domain computation. Optimized material properties and liner arrangement for noise reduction will be investigated.

MONDAY AFTERNOON, 23 MAY 2022 GOVERNORS SQUARE 12, 2:25 P.M. TO 3:25 P.M.

Session 1pCAb

Computational Acoustics, Signal Processing in Acoustics, Biomedical Acoustics, Physical Acoustics, and Structural Acoustics and Vibration: Contributions in Emerging Methods for Design and Optimization in Computational Acoustics

Amanda Hanford, Chair
Penn State University, PO Box 30, State College, PA 16802

Contributed Papers

2:25

1pCAb1. Transmission loss and end-correction modeling of extended inlet-outlet expansion chamber muffler and the concentric tube resonator. Archana S. Kale (Dept. of Phys., Nowrosjee Wadia College, V. K. Joag Path, Pune, Maharashtra 411009, India, sarchkale@gmail.com) and Farhat Surve (Dept. of Phys., Nowrosjee Wadia College, Pune, Maharashtra, India)

Measuring transmission loss of a muffler in a lab-setting may not always be feasible, as it ideally requires an anechoic termination at one end. Instead, a simulation algorithm employed as an effective tool that can handle large data and produce results in a short time is offered. Lab-grade expansion chamber mufflers with cross–sectional ratios 4.78, 7.27, and 12.14 are converted to extended inlet–outlet expansion chamber mufflers and then to concentric tube resonators employing simulation that concurrently optimizes their geometry. An algorithm to obtain the exact geometry using end-correction and extending the same to concentric tube resonators by varying the porosity handle is discussed.

2:40

1pCAb2. Enhancing acoustical design using parametric optimization of surface geometries. Benjamin C. Koziczinski (KOST Design, Münzstraße 10, Berlin, Berlin 10178, Germany, koziczinski@kost.design), Ivan C. Nieto (BCK Architektur, Berlin, Berlin, Germany), and Karlheinz Stegmaier (KOST Design, Berlin, Berlin, Germany)

Raytracing is commonly used to design surface geometries. To optimize this process, we propose combining a parametric design approach with an evolutionary solver. These optimization principles are widely implemented across engineering fields and could be exploited for acoustical design.

Evolutionary optimization is an iterative tool to increase the "fitness"—or optimality—of a predefined target value, in this case reflecting and non-reflecting geometries given any acoustical design brief. The iterative nature of an evolutionary solver—which evaluates each generation in terms of its proximity to the target value—enables acousticians/designers to continually assess results at different stages of the optimization process. The proposed algorithms optimize parameters such as size, shape, number or orientation of surfaces by defining raytracing target values (e.g., number of rays reaching a receiver after n-reflections, density of rays in a diffuse field). We implement this approach using standard software: the Galapagos solver within the Grasshopper framework, a parametric design tool in the Rhino 3D environment. The limitless design options offered by the parametric approach are thus made available for acoustical optimization. This method can be used on all scales of geometrical acoustics, from alignment of absorber panels to geometries of enclosed spaces, such as concert halls.

2:55

1pCAb3. Design optimization of metamaterials to control waves in cylindrical rods. Pravinkumar R. Ghodake (Dept. of Mech. Eng., Indian Inst. of Technol., Bombay, Mumbai, Maharashtra 400076, India, mech7pkumar@gmail.com)

Metamaterials can control different modes of waves by tuning their bandgap structures. Various design strategies are implemented to obtain desired responses of metamaterials by solving multiple forward problems using parametric sweeps, topology, and shape optimization problems. This study focuses on the design of cylindrical metamaterials to control wave propagation in a cylindrical rod using a shape optimization approach. Topology optimization requires relatively more computational resources and time as well as it also gives relatively complex structures in comparison with shape optimization. Optimal widths of layered elastic materials arranged periodically along an axis, on subsurface, and an outer surface of cylindrical surface are obtained by solving multiple time-dependent shape optimization problems. Optimization problems are solved using the finite element method and non-gradient optimization algorithm. Single and multi-objective functions are defined to reduce only axial and both axial as well as radial displacements integrated over the end surface, and monochromatic Gaussian input pulse is considered in this study. Every design solution obtained during each design iteration can be easily manufactured as a weak constraint is applied to the total length of the metamaterial. Prior knowledge of possible bandgaps helps to set an effective optimization problem, but it is not a necessary condition.

3:10

1pCAb4. Building information modeling (BIM) automation for architectural acoustics, mechanical noise calculation. Josh Thede (Henderson Engineers, Inc., 8345 Lenexa Dr. Ste. 300, Lenexa, KS 66214, josh.thede@hendersonengineers.com), Nick Boyts, Jeff Teel, Kevin Butler, Elaina Bargas, and Adam Roth (Henderson Engineers, Inc., Lenexa, KS)

There are several established procedures and design tools for estimating the background sound levels from heating, ventilation and air-conditioning (HVAC) building systems. Building Information Modeling (BIM) is a process commonly used in architecture, engineering, and construction that uses data to generate digital representations of a building's physical and functional characteristics. We will demonstrate a BIM-automated design tool which uses a Dynamo script and a Revit model duct layout to estimate HVAC noise levels in buildings. With just a few inputs in the model, the script calculates the sound attenuation of each duct element and receiver room sound correction to estimate the HVAC background sound level in any space or building type. Leveraging existing data through BIM automation can help create efficient, optimized designs and improve coordination between mechanical engineers, architects, and acoustical consultants to meet acoustical design criteria.

Session 1pEA

Engineering Acoustics: General Topics in Engineering Acoustics

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

Chair's Introduction—1:00

Contributed Papers

1:05

1pEA1. Development of an in-ear microphone for individualized measurement of hearing protection device output. David A. Anderson (Rocky Mountain Div., Appl. Res. Assoc., Inc., 1250 S Monaco PkwyUnit 42, Denver, CO 80224, danderson@ara.com), Andrew D. Brown (Speech and Hearing Sci., Univ. of Washington, Seattle, WA), Nathaniel Greene (Otolaryngol., Univ. of Colorado School of Medicine, Aurora, CO), Theodore F. Argo, and Bruno Mary (Rocky Mountain Div., Appl. Res. Assoc., Inc., Littleton, CO)

Hearing protection devices (HPDs) are designed to reduce the amplitude of sound reaching the ear, but can also distort the acoustical cues necessary for sound localization. In order to quantify the acoustic impact of an HPD on the signal reaching the tympanic membrane, the signals deep in the ear canal with and without the HPD must be compared. While some commercial microphone solutions for measuring in-ear pressures with HPDs exist, they are device-specific and/or may not provide reliable isolation of the microphone transducer from the (potentially high-intensity) ambient sound field. Here, we describe a new assembly using a small MEMS microphone attached to the end of a thin flexible circuit, which can be inserted either alongside or through virtually any HPD. Following a description of the electromechanical design of the assembly, we present measurements of insertion loss for HPDs with and without the microphone using an acoustic test fixture. These measurements demonstrate that the presence of the microphone does not significantly disrupt the sound attenuation characteristics of a range of HPDs. Preliminary measurements of insertion loss and head-related transfer functions in human subjects are also presented.

1:20

1pEA2. A passive radio frequency excited acoustic transducer. Charles Thompson (Elec. and Comput. Eng, UMASS Lowell, 1 Univ. Ave. Lowell, MA 01854, charles_thompson@uml.edu), Lejun Hu, Grace Remillard, Kavitha Chandra, Vacharaporn Paradorn, and Azizat Lawal (Elec. and Comput. Eng, UMASS Lowell, Lowell, MA)

The physical processes that govern the operation of a passive acoustic transducer driven into operation by a broadcast radio-frequency electromagnetic wave are presented. The transducer is comprised of a re-entrant cavity resonator operating in its resonant TEM mode and a sensing membrane. By displacing the membrane using the acoustic pressure, one can generate an amplitude modulated radio frequency signal proportional to the product of the displacement and the excitation signal. One may use the backscattered electromagnetic signal to recover the audio signal. The results for power generation, sound transduction, and radio-frequency backscatter transmission of the audio signal are presented.

1:35

1pEA3. Projector design for imaging of buried unexploded ordnance. Daniel C. Brown (Appl. Res. Lab., Penn State Univ., PO Box 30, State College, PA 16804, dcb19@psu.edu), Scott Brumbaugh, Mark Fanton, Richard Meyer, Shawn Johnson, and Cale Brownstead (Appl. Res. Lab., Penn State Univ., State College, PA)

Recent modeling and field experimentation has shown very-shallow-water acoustic imaging of buried unexploded ordnance (UXO) can place stringent requirements on the sonar transmit system. Achieving high-resolution imagery requires large transmit bandwidth. Imaging deeply buried UXO requires low operating frequencies and high source levels to overcome sediment attenuation. Imaging UXO near the sediment–water interface requires short transmit waveforms that require projectors with a low mechanical quality factor for accurate reproduction. This work presents the results of a recent design study, device fabrication, and acoustic testing of a new projector intended for operation as part of a buried UXO imaging system. The analysis will focus on the design and material selection of the head mass and the ring stack used in the tonpilz device. The head mass design trade considered construction from aluminum or an aluminum beryllium metal matrix composite (AlBeMet). The ring stack design trade considered traditional polycrystalline lead zirconate titanate (PZT), ternary single crystal lead indium niobium–lead indium manganese niobium titanate (PIN-PMN-PT), and textured manganese doped lead magnesium niobate–lead zirconate titanate (Mn-PMN-PZT). Finite element simulations and measured results will be presented.

1:50

1pEA4. One-dimensional modeling of the air motion transformer. Peter J. Riccardi (The Penn State Univ., State College, PA) and Thomas E. Blanford (The Penn State Univ., The Penn State Univ., State College, PA 16804, teb217@psu.edu)

The Air Motion Transformer (AMT), since its invention in the 1970s, has found application as a high-frequency tweeter for loudspeakers. It is perceived as reproducing audio with superior clarity and fidelity of transients compared to moving coil tweeter designs. It is well known that many factors, such as frequency response and properties of nonlinear distortion products, influence human perfect of audio quality. Despite the perceived superiority of AMT performance, few studies have investigated this transducer to accurately model their operation and understand the physical mechanisms that influence their sound quality. This presentation will describe a one-dimensional model for an AMT loudspeaker that can be used to predict both linear and nonlinear behavior of the device. The model will be compared to measured data from commercially available AMT tweeters.

Sources of nonlinear distortion in the AMT device that may influence perceived sound quality will be discussed. The similarities and differences between the electroacoustic properties of the AMT and comparable moving coil tweeters, and how these relate to sound quality, will also be discussed.

2:05

1pEA5. Monitoring industrial acoustics with distributed acoustic sensing. Derrick Chambers (Geophys., Colorado School of Mines, 1905 Tangent Way, Unit 77, Golden, CO 80401, derrickchambers@mines.edu), Peiyao Li, Harpreet Sethi, and Jeffery Shragge (Geophys., Colorado School of Mines, Golden, CO)

True-phase distributed acoustic sensing (DAS), a technique which uses low-power laser pulses to monitor along-fiber strain in optical cable, has proven useful in many geophysical research areas, including down-hole monitoring in oil/gas extraction, near-surface characterization, detecting and locating regional and global earthquakes, urban monitoring. Most of the geophysical applications to date, however, have focused on recording elastic waves propagating through solid media. In this work, we explore the response of DAS for recording acoustic propagation in air, as a function of fiber type and configuration, over frequency bands useful for monitoring industrial environments. We also present methods of creating simple fiber-composite sensing units for improving sensitivity, and strategies for combining solid-earth and acoustic monitoring to create an effective seismoacoustic array with a single DAS interrogator.

2:20

1pEA6. An acoustic near surface soil profiler using surface wave method. Zhiqu Lu (National Ctr. for Physical Acoust., 145 Hill Rd., University, MS 38677, zhiqulu@olemiss.edu)

An acoustic soil profiler, using a so-called the high-frequency multi-channel analysis of surface waves (HF-MASW) method, has been developed, which uses surface (Rayleigh) waves to measure soil profile in terms of the shear (S) wave velocity as a function of depth, up to a 2.5 m deep below the surface. Several practical techniques have been developed to enhance the HF-MASW method, including (1) a variable sensor spacing configuration, (2) the self-adaptive method, and (3) the phase-only signal processing. Fundamentally, the S-wave velocity is related to soil mechanical and hydrological properties through the principle of effective stress. Therefore, the measured two-dimensional S-wave velocity images reflect the temporal and spatial variations of soils due to weather effects, geological anomalies, and anthropologic activities. Several HF-MASW applications will be reported, including (1) near surface soil profiling, (2) a long-term-survey for studying weather and seasonal effects, (3) short-term monitoring rain fall events, (4) detecting fraigpan layers, and (5) a farmland compaction study. This acoustic soil profiler can be used for agricultural, environmental, civil engineering, and military applications.

2:35

1pEA7. Acoustic oscillations of drag-reducing air pocket under fast boat with stepped bottom. Konstantin I. Matveev (Washington State Univ., Sloan Hall, Pullman, WA 99164, matveev@wsu.edu)

One promising method for reducing water resistance of marine vessels is to create air pockets on their wetted hull surface. However, such pockets can experience acoustic oscillations when excited by periodic sea waves. These oscillations may reach high amplitudes at resonance conditions, when the air-pocket natural acoustic frequency approaches to the excitation frequency. This can lead to excessive air leakage from the pockets and eventual disintegration of air cavities, resulting in a loss of their drag-reduction effectiveness. A reduced-order acoustic model has been developed to analyze this process. Using realistic hull and air-pocket parameters, the model was applied to simulate acoustic oscillations of an air pocket under a fast boat. It is shown that increased damping inside air cavities can limit oscillation magnitudes to acceptable levels. Another strategy involves quick acceleration of the hull, so that time spent near the resonance condition is insufficient for substantial growth of high-amplitude acoustic oscillations. Active acoustic control can also be used to suppress the oscillations. [Work supported by the U.S. National Science Foundation under Grant No. 1800135.]

Session 1pMU

Musical Acoustics: General Topics in Musical Acoustics I

Kurt R. Hoffman, Cochair
Physics, Whitman College, 345 Boyer Ave., Hall of Science, Walla Walla, WA 99362

Taffeta Elliott, Cochair
CLASS, New Mexico Inst. of Mining and Technology, 801 Leroy Pl, Class Dept, Socorro, NM 87801

Chair's Introduction—1:00

Contributed Papers

1:05

1pMU1. Simulation of acoustic effects of postural laryngeal oscillatory movement during vocalization. Ingo R. Titze (Utah Ctr. for Vocology, Univ. of Utah, 240 S 1500 E, Rm. 206, Salt Lake City, UT 84112, ingo.titze@utah.edu), Anuja Sharma (Dept. of Radiology and Imaging Sci. and VAST Res. Lab, Univ. of Utah, Salt Lake City, UT), Ganesh Adluru (Dept. of Radiology and Imaging Sci. and; Dept. of Biomedical Eng., Univ. of Utah, Salt Lake City, UT), and Julie Barkmeier-Kraemer (Div. of Otolaryngol. and VAST Res. Lab, Univ. of Utah, Salt Lake City, UT)

This paper addresses the contributions of low-frequency postural laryngeal kinematics to frequency and amplitude modulations of the vocal output signal. A one-dimensional Navier–Stokes solution for fluid transport and wave propagation is used along with a laryngeal source model to simulate voice production. Vocal fold adduction and vertical laryngeal posture are modulated sinusoidally at 2–10 Hz with various amplitudes to approximate kinematic phenomena observed with MRI during production of vocal vibrato and voice tremor. The acoustic modulations are quantified in terms of formant (resonance) and fundamental frequency shifts in the radiated output signal from the mouth. Postural oscillation of the larynx is a common kinematic component of vocal tremor, although less documented during production of vibrato.

1:20

1pMU2. The effect of acoustic environments on singers' vocal behaviors and mental effort. Keiko Ishikawa (Speech and Hearing Sci., Univ. of Illinois, Urbana-Champaign, 901 S. 6th St., Champaign, IL 61820, ishikak@illinois.edu), Elisabeth Coster, Silvia Murgia, and PASQUALE BOTTALICO (Dept. of Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, Champaign, IL)

Singers perform in various acoustic environments; however, how the environments affect their mental effort is poorly understood. This study virtually simulated acoustic environments to examine the effect of reverberation time on singers' vocal behaviors and mental effort. Participants were five vocally healthy singers who completed a minimum of undergraduate-level training. For the virtual environments, the voice signal was digitally processed by adding reverberation using a real-time effect processor of the digital mixer. The processed sound was played back to the participant using open headphones. The average T30 was 1.13 s in Low T30 condition, 1.39 s in Medium T30 condition, and 1.90 s in High T30 condition. Participants sang the American national anthem and rated their perceived level of overall effort, mental effort, physical effort, frustration, and performance using a modified NASA-TLX scale for each acoustic condition. Pupillometry was also used to measure the mental effort during each task. Their voice recordings were acoustically analyzed for intensity, singing-power ratio, alpha ratio, pitch accuracy, and vibrato rate and extent. The presentation will discuss: (1) individual differences in the participants' response to the reverberation time and (2) the degree of agreement between the subjective rating and physical measurement of mental effort.

1:35

1pMU3. Cue selection in the perception of pitch in music and speech. May Pik Yu Chan (Linguist, Univ. of Pennsylvania, Dept. of Linguist 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)

Spectral shape affects pitch perception; sounds with more high energy harmonics sound higher than sounds with low energy in higher harmonics. Flatter spectral slope corresponds to tenser voices while steeper spectral slope correlates to breathier voices in speech. In string instruments, the spectral slope differentiates *sul ponticello* and *sul tasto*. Listeners were found to integrate spectral slope cues in pitch perception in speech; however, work on music focused on cross-instrument differences, glossing over cue integration within instruments with fixed-formant frequencies. Furthermore, spectral cues and F0 co-vary in human pitch production, but are largely independent in instrumental music. It remains unclear whether music processing is as integrative as speech processing. In this study, listeners were given either speech or violin stimuli with identical pitch contour pairs, and were asked to decide whether the second contour was higher or lower in pitch compared to the first. The spectral slope of each sound was manipulated to include all combinations of "breathier"/"*sul tasto*" and "tenser"/"*sul ponticello*" sounding pairs. Results show that listeners integrate spectral slope cues in pitch perception in speech and violin stimuli similarly, with similar categoricity and shift. Overall, listeners with higher musicality have more categorical responses but no differences in shift.

1:50

1pMU4. Flow visualization and aerosols in performance. Tehya Stockman (Civil, Environ., and Architectural Eng., Univ. of Colorado Boulder, Boulder, CO), Jean Hertzberg (Mech. Eng., Univ. of Colorado Boulder, Boulder, CO), and Abhishek Kumar (Mech. Eng., Univ. of Colorado Boulder, 11972 W Dakota Dr., Lakewood, CO 80228, abku6744@colorado.edu)

The COVID-19 pandemic has created a great deal of fear and uncertainty around whether aerosols from singers and musical instruments can transmit the virus. The World Health Organization has recognized that SARS-COV-2 is transmissible through aerosols. Outbreaks from choir performances, such as the Skagit Valley Choir, showed that singing brings potential risk of COVID-19 infection. Additionally, outbreaks from singing have resulted in superspreader events leading to many infections and deaths. There is less known about the risks of airborne infection from other musical

performance, such as playing wind instruments or performing theater. Hence, aerosol generation in performance (playing brass and woodwind instruments, singing, and theater speech delivery) should be quantified, monitored, and mitigated. To tackle this issue our research questions were as follows: (i) What is the aerosol generation rate? (ii) How does air flow from the performer's mouth/instrument and how does it disseminate into the environment? and (iii) What control methods can be employed to mitigate risk? In this study, we used a variety of methods, including flow visualization, aerosol and CO_2 measurements, and computational fluid dynamics (CFD) modeling to understand the different components that can lead to transmission risk from musical performance and risk mitigation.

2:05

1pMU5. Does musical training affect neuro-cognition of emotions? An EEG study with instrumental Indian classical music. Medha Basu (Jadavpur Univ., Kolkata, West Bengal 700032, India, medhabasu1996@gmail.com), SHANKHA SANYAL (Lang. and Linguist, Jadavpur Univ., Kolkata, West Bengal, India), Archi Banerjee (IIT Kharagpur, Kharagpur, India), Kumardeb Banerjee, and Dipak Ghosh (Jadavpur Univ., Kolkata, West Bengal, India)

Music across all genres evokes a variety of emotions, irrespective of its timbre and tempo. Indian classical music (ICM) is no exception. Although being biased towards vocal musical styles, instrumental music forms one broad section of ICM. In this study, we have tried to compare the neural responses of music practitioners and non-musicians towards different emotions using audio clips from two popular plucked string instruments used in ICM, *Sitar* and *Sarod*. From pre-recorded performances of two eminent maestros, 20 clips of approximately 30 s duration were selected from the *Alaap* sections (initial introductory section without any rhythmic accompaniment) of different *Raagas* played in the two instruments. From an audience response assessment of 100 participants, a total of eight clips having maximum arousal for happy and sad emotions were identified from the 20 clips, using which EEG (Electroencephalography) recordings were collected from five musicians and five non-musicians. Robust nonlinear Multifractal Detrended Fluctuation Analysis technique (MFDFA) was applied to quantitatively measure the brain-state changes in different lobes for both categories of participants. In essence, this study attempts to encapsulate if and how prior musical training influences the brain responses towards two basic musical emotions in ICM using two instruments of same family.

MONDAY AFTERNOON, 23 MAY 2022 — GOVERNORS SQUARE 16, 1:00 P.M. TO 3:00 P.M.

Session 1pPA

Physical Acoustics and Biomedical Acoustics: Advances in Sonochemistry II

James Kwan, Chair

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

Invited Paper

1:00

1pPA1. Ammonia chemistry: Sounds better with ultrasound. Prince N. Amaniampong (CNRS-Univ. of Poitiers, France, Faculté des Sci. Fondamentales et Appliquées (UFR SFA) Institut de Chimie des Milieux et Matériaux de Poitiers (IC2MP, CNRS) Catalysis and Unconventional Media Team(Equip E4) Bâtiment B1, Rue Marcel Doré, TSA41105 86073 - Poitiers Cedex 9 (France), Poitiers 86000, France, prince.nana.amaniampong@univ-poitiers.fr), Anaelle Humblot, Karine De Oliveira Vigier, and Francois Jerome (CNRS-Univ. of Poitiers, France, Poitiers, France)

Hydrazine is a chemical of utmost importance in our society, either for organic synthesis or energy use. The direct conversion of NH_3 to hydrazine is highly appealing, but it remains a very difficult task because the degradation of hydrazine is thermodynamically more feasible than the cleavage of the N–H bond of NH_3. As a result, any catalyst capable of activating NH_3 will thus unavoidably decompose N_2H_4. Here, we show that cavitation bubbles, created by ultrasonic irradiation of aqueous NH_3 at a high frequency, act as microreactors to activate and convert NH_3 to NH species, without assistance of any catalyst, yielding hydrazine at the bubble–liquid interface. The compartmentation of *in-situ*-produced hydrazine in the bulk solution, which is maintained close to 30 °C, advantageously prevents its thermal degradation, a recurrent problem faced by previous technologies. This work also points towards a path to scavenge .OH radicals by adjusting the NH_3 concentration.

Contributed Paper

1:30

1pPA2. A multi-frequency sonochemical reactor utilizing a cylindrically focused acoustic wavefield for improved sonochemical efficiency. Jason L. Raymond, Ronald A. Roy (Dept. of Eng. Sci., Univ. of Oxford, Oxford, United Kingdom), and James Kwan (Dept. of Eng. Sci., University of Oxford, Parks Rd., Oxford OX1 3PJ, United Kingdom, james.kwan@eng.ox.ac.uk)

Sonochemistry involves several steps. First, an acoustic wave must be generated with sufficient energy to nucleate cavitation. This cavitation event must then be energetic enough to produce radicals through the sonolysis of the gas or vapor molecules in the bubble. Finally, these radical species must survive long enough to react with other chemical species to yield the desired product. The sonochemical efficiency (SE) is thus defined as the ratio of the product yield to the input acoustic power. Here, we put forward a novel approach to enhancing the SE by designing a multi-frequency (0.25 to 1 MHz) sonochemical reactor that reflects a cylindrically spreading acoustic wave 180°. The reflected wave converges to cylindrically shaped high-intensity "hot zone" positioned along the axis of the reaction vessel. At modest power input, we detected noise with a passive cavitation detector in air-saturated water, suggesting the direct nucleation of cavitation. Using the Weissler reaction to determine the production of hydroxyl radicals, we determined the SE of our sonochemical reactor at various operation conditions. We also compared these values to conventional bath, horn, and plate sonochemical reactors. The comparison suggested that our reactor design had a better sonochemical efficiency for the Weissler reaction.

Invited Papers

1:45

1pPA3. Nanostructured TiO_2 as a sonophotocatalytic cavitation agent for spatially controlled sonochemistry. Umesh S. Jonnalagadda (Nanyang Technol. Univ., Chemical and Biomedical Eng. (NTU), 62 Nanyang Dr., Singapore 637459, Singapore, umeshsj@ntu.edu.sg), Fan Qianwenhao, Xiaoqian Su, Wen Liu (Nanyang Technol. Univ., Singapore, Singapore), and James Kwan (Eng. Sci., Univ. of Oxford, Oxford, United Kingdom)

Sonochemistry has garnered interest in its ability to facilitate greener chemistry by minimizing the use of hazardous reagents and harsh reaction conditions commonly used in chemical synthesis. The operating principle for sonochemistry is inertial cavitation, whereby bubble collapse will induce water pyrolysis to generate radicals for use in chemical reactions. Unfortunately, this process is limited by the low efficiencies in converting electrical energy to cavitation energy for free radical generation. To remedy this, researchers have exploited phenomena, e.g., sonoluminescence, using heterogenous catalysts to increase the radical generation rate; however, the stochastic nature of cavitation requires prolonged periods of high intensity, continuous wave irradiation to generate sufficient reaction rates. To address these limitations, we nanostructured TiO_2 to function as sonophotocatalytic cavitation agents (SCAs). Our SCAs demonstrate that sonoluminescence can more efficiently occur at the catalyst surface and at lower energies using pulsed ultrasound. We evaluated the functionality of our SCAs by benzyl alcohol oxidation to benzaldehyde, where we further modified the catalysts using AuPd nanoparticles to function as a co-catalyst for enhanced sonoreactivity. Our findings present the key design elements for developing an effective SCA, allowing future work to explore the versatility of SCAs for a wide range of chemical applications.

2:15

1pPA4. Synergistic effect between nanostructured catalysts and high-frequency ultrasound: Application in biomass conversion to specialty chemicals. Sabine Valange (CNRS-Univ. of Poitiers, France, ENSI Poitiers, B1, 1 rue Marcel Doré, TSA 41105, Poitiers 86073, France, sabine.valange@univ-poitiers.fr), Prince N. Amaniampong, Karine De Oliveira Vigier, and Francois Jerome (CNRS-Univ. of Poitiers, France, Poitiers, France)

Within the context of sustainable chemistry, sonochemistry is now emerging as an alternative unconventional technology in catalysis. While ultrasound-generated radicals can participate in chemical reactions, mastering the reaction selectivity of polyfunctional substrates remains an elusive task. To address this challenge, we designed nanostructured metal oxides with leaf-like morphologies (e.g., CuO) by sonochemical synthesis (20 kHz) and investigated their activity for controlling the selectivity of oxidation reactions under ultrasonic irradiation in aqueous solution. We demonstrated that colocalization of the cavitation event onto the CuO surface active sites enables the direct utilization of radicals generated by cavitation for chemo-selective chemical reactions. In particular, we provided evidence for an alternative reaction pathway in glucose selective oxidation through synergistic cavitation–catalyst interactions at 550 kHz. We showed that the unwanted H• radicals stemming from water sonolysis are trapped by the surface lattice oxygen of CuO, thereby increasing the coverage of •OH radicals on the catalyst surface, and steering the selective oxidation of glucose to glucuronic acid, a valuable chemical whose synthesis remains a formidable challenge in the field of catalysis. This work also highlights that the particle size of the sonocatalyst is a key parameter governing an optimal transfer of radicals from the cavitation bubbles to the catalyst surface.

Contributed Paper

2:45

1pPA5. Acoustics of Drop Impact on Hydrophobic Surfaces and Liquid Pools. John S. Allen (Mech. Eng., Univ. of Hawaii Manoa, Holmes 302, 2540 Dole St., Honolulu, HI 96822, alleniii@hawaii.edu), Rafsan Rabbi, Akihito Kiyama (Dept. of Mech. Eng., Utah State Univ., Logan, UT), and Tadd Truscott (Dept. of Mech. Eng., King Abdullah Univ. of Sci. and Technol., Thuwal, Saudi Arabia)

Drop impact on surfaces has been studied comprehensively as it has a wide range of fundamental and practical implications. The splash, spreading, and rebound have been investigated with respect to substrate interfaces with hydrophobic surfaces being of particular interest. However, impact on wetted substrates is less understood for the hydrophobic surfaces. Also the associated acoustics of the drop impact has not be explored in terms of air entrainment and substrate vibrations. We investigate the impact of water drops (~2–10 mm diameter) from heights 100–400 cm upon hydrophilic– and hydrophobic–coated solid surfaces as well as free floating liquid pools (0.15–0.45 ml). A contact microphone coated with the Glaxco water repellant provides a hydrophobic surface upon which impact vibrations can be quantified. An air microphone is synchronized to a high-speed Phantom camera for sound recording and optical visualization of the impact and rebound phases. Cross-correlation of the two microphones reveals distinct differences between the two solid substrates. The initial rebound and subsequent jet break-up are found in an analysis of the signal's amplitude and phase. These results are compared to those from gel spheres and discussed for applications of non-contact coating monitoring.

MONDAY AFTERNOON, 23 MAY 2022 — PLAZA EXHIBIT HALL, 1:00 P.M. TO 4:00 P.M.

Session 1pSC

Speech Communication: Speech Production and Acoustics II (Poster Session)

Yoonjeong Lee, Chair

Linguistics, University of California, Los Angeles, 31-19 Rehabilitation Center, 1000 Veteran Ave., Los Angeles, CA 90095

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

Contributed Papers

1pSC1. Phonetic alignment to regional variants in interaction. Lotte Eijk (Ctr. for Lang. Studies, Radboud Univ., Erasmusplein 1, Nijmegen 6525HT, Netherlands, lotte.eijk@ru.nl), Herbert Schriefers (Donders Inst. for Brain, Cognition and Behaviour, Radboud Univ., Nijmegen, Netherlands), and Mirjam Ernestus (Ctr. for Lang. Studies, Radboud Univ., Nijmegen, Netherlands)

Alignment is the process of adapting speech to another interlocutor's speech. We set out to explore whether speakers align phonetically to regional variants since no clear evidence has been reported (Gessinger *et al.*, 2019; Earnshaw, 2021) and if so, what the driving mechanism is (local versus global context). We investigated phonetic alignment to two regional variants of a Dutch phoneme, known as the "hard g" or "soft g." Participants interacted with two confederates differing in their regional variants in a sentence completion task (total of 268 sentences). In a pre-test, participants completed sentences by themselves. They then interacted with Confederate 1 (Round 1), with Confederate 2 (Round 2), and again with Confederate 1 in Round 3, and lastly by themselves again in the post-test. We investigated the duration and Centre of Gravity of the 15 085 fricatives of 36 participants. We examined three different predictors testing for differences between short-term and long-term alignment effects. None of these predictors showed significant effects of alignment. Descriptive analyses showed tremendous variation among speakers. We conclude that phonetic alignment of regional variants is not as clear as phonetic alignment previously demonstrated in less ecologically valid studies.

1pSC2. Linguistic and personal influences on speaker variability. Yoonjeong Lee (Linguist, Univ. of California, Los Angeles, 31-19 Rehabilitation Ctr., 1000 Veteran Ave., Los Angeles, CA 90095, yoonjeonglee@ucla.edu) and Jody Kreiman (Linguist, Univ. of California, Los Angeles, Los Angeles, CA)

Our previous studies examined the manner in which within- and between-speaker acoustic variability in voice follow patterns determined by biological factors, the language spoken, and individual idiosyncrasies. To date, we have analyzed data from speakers of English, Seoul Korean, and Hmong, which differ in whether they contrast phonation type and/or tone. We found several factors that consistently account for acoustic variability across languages, but also factors that vary with phonology. The present study adds Gujarati (which contrasts breathy with modal phonation) and Thai (a tone language without contrastive phonation) to this work. We hypothesize that F0 will emerge from analyses of Thai, as it did for Hmong and Korean (but not for English), and that differences in the amplitudes of lower harmonics will emerge for Gujarati, as occurred for Hmong, but not for English or Korean. We further hypothesize that two factors—the balance

of high-frequency harmonic and inharmonic energy and formant dispersion—will emerge as the most important factors explaining acoustic variance for these new sets of speakers and languages, as they have in our previous studies. Such a result would be consistent with the view that speaker variability is governed by both biological and linguistic factors.

1pSC3. Sociophonetic variation among Asian Americans: The role of ethnicity and style. Danielle Dionne (Dept. of Linguist, Boston Univ., 621 Commonwealth Ave., Boston University, Boston, MA 02215, ddionne@bu.edu) and Charles B. Chang (Dept. of Linguist, Boston Univ., Boston, MA)

The present study examined sociophonetic variation in a small sample of Asian Americans in Boston, Massachusetts ($N = 8$; 4f, 4m; $M_{age} = 23$) representing four ethnic groups—Chinese, Filipino, Korean, and Vietnamese. Analyzing these speakers' English production in tasks eliciting both casual and careful speech, we focused on four linguistic features comprising features observed in New England and in certain Asian American groups. Three features (L/R-CONFLATION, L-VOCALIZATION, and R-DELETION) were coded auditorily and one (LOW BACK RAISING of /ɑ/ to /ɔ/) acoustically; the dataset included approximately 1500 tokens of each feature, for a total of around 6000 tokens. Mixed-effects modeling results on the casual speech data indicated that Ethnicity was a significant predictor (p's $< .05$) of the occurrence of L-VOCALIZATION ($M = 86.6\%$ overall) and R-DELETION ($M = 8.5\%$), but not of L/R-CONFLATION (which did not occur at all) or LOW BACK RAISING (which did not clearly occur in F_1 or F_2). Preliminary analysis of the careful speech data showed lower rates of L-VOCALIZATION ($M = 70.0\%$) and R-DELETION ($M = 1.5\%$) than in casual speech and, again, no occurrence of L/R-CONFLATION or LOW BACK RAISING. These findings reveal similarities and differences in speech production among ethnically diverse Asian Americans, point to a possible role of style, and highlight the need for further investigation of phonetic variation within this community.

1pSC4. Speech-based biomarkers for automatic detection of GERD: A pilot study. Mary Pietrowicz (Appl. Res. Inst., Univ. of Illinois at Urbana-Champaign, 2100 S Oak St., Champaign, IL 61820, Champaign, IL 61820, marybp@illinois.edu), Amrit Kamboj, Manoj Yarlagadda, Kevin Buller (Div. of Gastroenterology and Hepatology, Mayo Clinic, Rochester, MN), Keiko Ishikawa (Speech and Hearing Sci., Univ. of Illinois, Urbana-Champaign, Champaign, IL), Diana Orbelo (Dept. of Otolaryngol., Mayo Clinic, Rochester, MN), and Cadman Leggett (Div. of Gastroenterology and Hepatology, Mayo Clinic, Rochester, MN)

This study explores the feasibility of detecting gastroesophageal reflux disease (GERD) and Barrett's esophagus (BE) using acoustic features extracted from speech recordings. GERD is a common condition affecting up to 27.8% of adults in the US. Chronic GERD is associated with BE, the precursor to esophageal adenocarcinoma. An automatic screening tool for GERD would improve clinical outcomes by improving early detection. Biomarkers based on acoustic features of speech may hold promise as hoarseness often co-occurs with GERD. The study cohort consisted of 49 adults with and 63 without GERD based on esophagogastroduodenoscopy or ambulatory pH studies. Speech, including sustained vowels, sentences, and reading, was systematically recorded. Voice quality was validated through auditory-perceptual ratings with three speech-language pathologists. Automated, cross-validated, search across multiple machine and deep learning model types was performed using the recorded speech. Detection of condition presence and severity are explored, and model performance is reported, along with an assessment of the ability of acoustic features to discern across conditions.

1pSC5. Triggering and spreading of lateral tongue posture. Yadong Liu (Linguist, Univ. of BC, 202-2720 Acadia Rd., Vancouver, BC V6T 1R9, Canada, yadong@connect.hku.hk) and Bryan Gick (Linguist, Univ. of BC, Vancouver, BC, Canada)

Lateral bracing is a tongue posture that is pervasively maintained during speech across languages (Liu *et al*., in press), released only for a few English sounds ([l] and occasional low vowels; Gick *et al*., 2017). While bracing is required for many (e.g., coronal) sounds, it is unclear why bracing is maintained during non-lingual sounds (e.g., labial/glottal consonants). Specifically, it is unknown whether bracing is maintained during non-lingual sounds because it is a pervasive default posture for speech (i.e., turned "on" unless actively suppressed, e.g., for /l/) or whether an unbraced posture can be maintained during extended periods. We collected coronal cross-section ultrasound imaging of the dorsal tongue while participants said "hubba-bubba" (a four-syllable sequence of non-lingual consonants and low vowels) in two contexts: surrounded by mandatorily braced sounds (e.g., /s/) versus unbraced sounds (e.g., /l/). Preliminary results show that the lateral tongue stayed in a lowered position throughout "hubba-bubba" when surrounded by unbraced sounds compared to braced sounds. Results suggest the lateral tongue may be held in one of at least two distinct postures whose activation is controlled by "trigger" events. [Work supported by NIH and NSERC.]

1pSC6. An ultrasound study of high vowel devoicing in Tokyo Japanese: Evidence for the vowel gesture retention. Rion Iwasaki (Speech-Language-Hearing Sci., City Univ. of New York, 365 Fifth Ave., Rm. 7304, New York, NY 10016, riwasaki@gradcenter.cuny.edu), Kevin D. Roon (Speech-Language-Hearing Sci., City Univ. of New York, New York, NY), Jason A. Shaw (Dept. of Linguist, Yale Univ., New Haven, CT), Mark Tiede (Haskins Labs., New Haven, CT), and D. H. Whalen (Speech-Language-Hearing Sci., City Univ. of New York, New York, NY)

High vowels in Tokyo Japanese are typically devoiced between voiceless obstruents, but controversy remains over whether vowel gestures persist when devoiced or are instead deleted. A previous ultrasound study (Iwasaki *et al*., 2020) showed that the lingual articulation of the release burst of /kV/ differs by vowel context even when the vowels are devoiced. This study uses tongue surface contours derived from midsagittal ultrasound images to investigate the effects of vowel devoicing on changes in quantified tongue shape over time. Native speakers of Tokyo Japanese produced word pairs (/C_1VC_2e/) that contrasted in the voicing of V, which was either /i/ or /u/. Tongue shape was characterized by Fourier transforming tangent angles along each contour (Dawson *et al*., 2016). Time-normalized trajectories over the /C_1VC_2e/ sequence were compared by vowel context (/i/ *versus* /u/) and voicing environment (devoiced *versus* voiced). Preliminary results show that the real component of the first Fourier coefficient is sensitive to detecting evolving shape differences between the two vowel contexts over the sequence, not just when the vowels are voiced, but also when they are devoiced. Based on these results, the high vowel contrast persists even in the devoiced environment, suggesting that devoiced vowels are not deleted.

1pSC7. Effect of speech-related smile suppression on emotional valence. Nicole Ebbutt (Linguist, Univ. of BC, 6368 Stores Rd., Vancouver, BC V6T 2B4, Canada, nicoleebbutt@gmail.com), Kyra Hung (Linguist, Univ. of BC, Vancouver, BC, Canada), Magdalena Ivok (Linguist, Simon Fraser Univ., Vancouver, BC, Canada), Charissa Purnomo, Farhan Samir, Gillian de Boer, and Bryan Gick (Linguist, Univ. of BC, Vancouver, BC, Canada)

The physical act of smiling has direct positive effects on mood [Kleinke *et al*., *Pers. Soc. Psychol.* **74**, 272–279 (1998)]. Relatedly, Rummer *et al*. [*Emotion*, **14**(2), 246–250 (2014)] observed that participants rated comics as funnier if they had just produced /i/ (which requires adopting a smile-like position) than if they produced /o/. The present study tests whether suppression of smile posture by speech movements can cause individuals to view a subject less positively. To do so, we ask participants to maintain a smile while we present a series of visual stimuli labeled with target and control sounds. Bilabial sounds are targeted as Liu *et al*. found that bilabial stops (/p/ and /b/) suppress smile posture [ISSP12, 130–133 (2021)]. After articulating a sound that either suppresses or does not suppress their smile posture, participants rate each image set on a measure of emotion. Results will be presented and discussed bearing on the prediction that in the smile condition, participants will rate the image as less positive if they have just produced a sentence which includes a bilabial stop. [Work supported by NIH and NSERC.]

1p MON. PM

1pSC8. Cortical dynamics underlying speech planning in rapid conversational turn-taking. Gregg A. Castellucci (Neurosci. Inst., NYU School of Medicine, 435 E30th St., NYULMC Sci. Bldg., New York, NY 10016, gregg.castellucci@nyulangone.org), Christopher Kovach, Jeremy Greenlee (Neurosurgery, Univ. of Iowa, Iowa City, IA), and Michael Long (Neurosci. Inst., NYU School of Medicine, New York, NY)

During conversation, silent gaps between speakers are typically 200ms or less—suggesting that speech planning often occurs while speakers listen to their partners' turns. Though the psycholinguistic mechanisms of planning during turn-taking are well-studied, its neural underpinnings are largely unknown. Using intracranial electrocorticography in neurosurgical patient-volunteers, we delineated planning-related activity from dynamics underlying sensorimotor processes using an interactive question–answer paradigm (adapted from Bögels *et al*., 2015) and found that planning activity is largely restricted to a left hemisphere network centered on caudal inferior and middle frontal gyri. In a separate task, participants performed both speech and non-speech actions, and we found that this planning network is most active while planning spoken output. We then examined neural activity during unconstrained conversation and found that the identified planning circuit is active prior to participant turn onset and while participants listen to the turns of conversational partners. Finally, in preliminary direct cortical stimulation experiments, we found that perturbation of this planning network results in significantly longer reaction times and lexical errors but not gross articulatory disruptions. In conclusion, using controlled interactive language tasks, we uncovered a speech-selective cortical planning circuit that is active during natural conversation and required to execute rapid turn-taking.

1pSC9. A sensorimotor approach to disfluency adaptation in typically fluent adults. Torrey Loucks (Commun. Sci. and Disord., Univ. of AB, 8205 114 St NW, Edmonton, AB t6g 2g4, Canada, loucks@ualberta.ca) and Daniel Aalto (Commun. Sci. and Disord., Univ. of AB, Edmonton, AB, Canada)

Delaying auditory feedback (DAF) during speech is a potent auditory perturbation that can interrupt and prolong syllables and words, even though these disfluencies rarely affect typical speakers under non-altered feedback (NAF). Adaptation or compensation to auditory perturbations has been shown in typical speakers through a gradual reduction in the amplitude of formant shift responses. However, despite considerable research on DAF, it is still not known whether typical speakers adapt to DAF. In this study, we tested whether a comparable form of adaptation occurs with DAF in typical speakers as shown by a reduction in altered feedback disfluencies (AFD) during repeated consecutive readings, after a pause between readings and to a novel reading. We then tested for carryover effects after a single DAF exposure. A significant decrease in AFD rate was observed in 38 speakers that was sustained after a pause and for a novel reading. The adaptation effect extended to articulation rate (syllables/sec) in that rate increased for all speakers across readings. Evidence for carryover effects was inconclusive. By showing that typical speakers can adapt to DAF, the findings support a sensorimotor approach to auditory perturbation adaptation achieved through motor practice.

1pSC10. Acoustic Voice Analysis during Phonation Onset, Measured from High-Speed Videoendoscopy in Connected Speech. Trent M. Henry (Communicative Sci. and Disord., Michigan State Univ., 1026 Red Cedar Rd., Rm. 207, East Lansing, MI 48824, henrytr2@msu.edu), Dimitar Deliyski (Communicative Sci. and Disord., Michigan State Univ., East Lansing, MI), Stephanie R. Zacharias (Head and Neck Regenerative Medicine Program, Mayo Clinic, Scottsdale, AZ), and Maryam Naghibolhosseini (Communicative Sci. and Disord., Michigan State Univ., East Lansing, MI)

Adductor spasmodic dysphonia (AdSD) is a neurological voice disorder, which affects the control of intrinsic muscles of the larynx during speech. Our previous research has shown that high-speed videoendoscopy (HSV) can be used to measure the glottal attack time for patients with AdSD during connected speech. This study builds upon that to analyze the acoustic signal during the phonation onset and prephonatory adjustments. The phonation onset events are quantified using the HSV data that were obtained simultaneously with the acoustic data. We analyze the power and energy waveform of the acoustic signal during vocal folds' first contact, first oscillation, and transitory behaviors, extracted from high-temporal resolution HSV data. The data were obtained from five patients with AdSD and five vocally normal participants. The data were recorded while the participant produced: 2 productions of vowel /i/ at habitual pitch and loudness, 2 with a soft glottal attack and 2 with a hard glottal attack, six CAPE-V sentences, and the "Rainbow Passage." The HSV system included a monochrome high-speed camera coupled with a flexible nasolaryngoscope. The results of this study could assist for more accurate clinical management of adductor spasmodic dysphonia.

1pSC11. Speed of velum movement during nasal segments and rest intervals: A cineradiographic study of French and English speech. Md Jahurul Islam (Linguist, The Univ. of BC, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, jahurul.islam741@gmail.com), Gillian de Boer, and Bryan Gick (Linguist, The Univ. of BC, Vancouver, BC, Canada)

While opening/closing of the velopharyngeal port (VPP) in speech has been much studied, the speed of these movements has been largely overlooked. The present study compares opening/closing velocities of the VPP in French and English, testing relation to distance traveled and speech-related versus physiological movements. Running speech samples from nine Quebecois French speakers and four Canadian English speakers were obtained from the Université Laval X-ray videofluorography database [Munhall *et al*., *J. Acoust. Soc. Am*., **98**(2), 1222–1224 (1995)]. Using ImageJ software, we tracked VPP opening/closing movements during two types of events: phonologically nasal segments and rest intervals between chunks of speech. We calculated velocity of VPP opening/closing during these events and analyzed the data using linear mixed-effects models to identify differences between the nasals and rest intervals as well as for any cross-linguistic differences. Results indicated that: (1) VPP closure was faster following English nasals than French nasals; (2) VPP opening was faster than closure for rest intervals in French; and (3) VPP opening/closing speeds were faster coming into/out of rest position than into/out of nasals in French. Preliminary cross-language observations support a correspondence between velocity and distance. [Work supported by NIH and NSERC.]

1pSC12. Realizations of Malagasy vowel devoicing. Jake Aziz (Linguist, Univ. of California, Los Angeles, 3125 Campbell Hall, Los Angeles, CA 90095, jakeaziz@ucla.edu)

This paper investigates the acoustic realizations of so-called "devoiced" vowels in Merina Malagasy. Malagasy has five monophthongs (/aeiou/; Howe, 2019) of which /a/, /i/, and /u/ have been said to be devoiced. This paper represents the first thorough description of the acoustics of these vowels. In a production experiment, speakers pronounced 115 tokens involving /a/, /i/, and /u/ in prosodic environments described as causing devoicing. Preliminary results indicate that so-called devoiced vowels may be realized as one of at least three variants: devoiced, co-articulated, or deleted. When devoiced, which typically occurs following a voiceless obstruent, the vowel is realized as a lengthening of the aperiodic noise associated with the preceding obstruent. When co-articulated, which is common following some sonorants, the vowel is realized as a gesture on the preceding sonorant without taking up its own time slot; for example, /u/ is realized as a lowered F2 on the preceding sonorant, indicating rounding. Finally, a vowel may be fully deleted, in which case the vowel has no acoustic representation, neither as a gestural overlap with, nor lengthening of the preceding consonant.

1pSC13. Vocalic contrasts in Hnaring Lutuv. Amanda Bohnert (Linguist, Indiana Univ., Bloomington, Bloomington, IN) and Kelly H. Berkson (Linguist, Indiana Univ., Bloomington, 1020 E. Kirkwood Ave., Ballantine Hall 852, Bloomington, IN 47405-2201, kberkson@indiana.edu)

Lutuv, sometimes called Lautu, is an under-documented Chin language from the Tibeto–Burman language family spoken in Chin State in Western Burma by approximately 15 000 people (Eberhard *et al*., 2021, citing 2005 data that do not account for current military violence and displacement). Lutuv is also spoken in diaspora communities worldwide, including by about 1000 people in the Chin refugee community of Indianapolis (community estimate). Ongoing fieldwork with the Hnaring variety has revealed

that, compared to related languages, Lutuv has undergone radical syllable structure simplification and attendant vowel inventory expansion. In addition to the low vowel /a/ and mid vowels /e ə ɔ/, Lutuv contains six to ten high vowels. These tentatively include four diphthongal vowels (/ie y$^{\text{ə}}$ uɔ ɯ$^{\text{ə}}$) as well as six monophthongs /i y ɨ ʉ ɯ u/. We explore the distribution of the vowels via a combination of acoustic and comparative data in an effort to better understand the articulatory and featural specifications thereof, with special attention paid to the high vowels. If all of these monophthongs truly are high, it would represent a startlingly large and typologically uncommon high vowel inventory.

1pSC14. Oral vibratory sensation at different laryngeal and semi-occluded vocal tract configurations during voice production. Zhaoyan Zhang (Dept. of Head and Neck Surgery, Univ. of California, Los Angeles, 1000 Veteran Ave. 31-24 Rehab Ctr., Los Angeles, CA 90095, zyzhang@ucla.edu)

Voice therapy aimed at improving vocal efficiency, such as resonant voice therapy or semi-occluded vocal tract exercises, often emphasizes vibratory sensations in the front part of the vocal tract during phonation. It remains unclear what laryngeal and vocal tract adjustments are elicited in patients by this emphasis on oral vibratory sensations and how these adjustments improve voice production. This study aims to identify laryngeal and vocal tract adjustments that maximize oral vibratory sensations during phonation, as quantified by the oral sound pressure level (SPL), at different laryngeal and semi-occluded vocal tract conditions in a three-dimensional phonation model. Results show that maximum oral SPL is reached at intermediate vocal fold adduction conditions. Epilaryngeal tube narrowing further increases the oral SPL in an open vocal tract, but this effect is much reduced and even reversed in a semi-occluded vocal tract, due to the reduced sensitivity of the first formant to epilaryngeal manipulation in a constricted vocal tract. These findings suggest that emphasis on oral vibratory sensations generally elicits a laryngeal configuration that is neither too tight nor too open. In a semi-occluded vocal tract, this emphasis may also reduce the degree of epilaryngeal narrowing as often observed in recent imaging studies.

1pSC15. Dynamic targets in second language vowel articulations. Madeleine Oakley (North Carolina State Univ., 2211 Hillsborough St., Raleigh, NC 27607, mo643@georgetown.edu)

This study examines L1 English-L2 French vowel productions, with the goal of exploring whether representing vowel targets as dynamic rather than "steady-state" better predict L2 production patterns. It is hypothesized that learners will transfer L1 vowel dynamic information to produce L2 vowels if the two are perceived as "similar," but it is unclear whether learners will transfer dynamic vowel information to "new" L2. As such, six L1 English-L2 French learners completed production tasks in English and French with the target vowels /i, u, e, o/ and /y, ø/ in real words. Participants were recorded using a stabilization headset attached to an ultrasound probe and microphone. F1 and F2 measurements and tongue contours were extracted at three timepoints throughout the vowel duration. Results shows that while learners produce L2 French vowels /i, e, u, o/ with a similar amount of formant and gestural movement to L1 vowels, they do not transfer midpoint F1-F2 values. Learners *do* produce new L2 vowels /y, ø/ with similar formant and gestural movements to English vowels. The results of this study indicate that learners transfer dynamic acoustic information and articulatory gestures from their L1 to produce L2 vowels, which has implications for the representation of vowels.

1pSC16. The degree and time course of nasal coarticulation across communicative contexts: A study of the LUCID corpus. Zhe-chen Guo (Linguist, The Univ. of Texas at Austin, 307 E 31st ST APT 105, Austin, TX 78705, y9024131@gmail.com) and Rajka Smiljanic (Univ. of Texas at Austin, Austin, TX)

We analyzed the degree and time course of coarticulatory vowel nasalization in hyperarticulated clear speech produced in real and imagined communicatively challenging conditions from the LUCID corpus (Baker & Hazan, 2011). Southern British English speakers completed an interactive spot-the-difference task in pairs when there was no overt communication barrier (NB), when the partner was non-native (L2), and when one speaker's speech was vocoded (VOC) or mixed with talker babble (BABBLE). They also read sentences conversationally (READ-CO) and clearly (READ-CL). The results showed significantly greater overall keyword vowel nasalization in the BABBLE and L2 conditions than in NB, READ-CO, and READ-CL, as well as in VOC than in NB and READ-CO. READ-CL and READ-CO did not differ significantly. Examining vowel nasalization over time revealed that the significant differences emerged, on average, at 7.4% into the vowel. Speakers increased coarticulatory nasalization early in the vowel when producing hyperarticulated speech in response to a real communicative barrier, consistent with the idea that nasal coarticulation facilitates speech processing by cueing upcoming segments. It remains to be determined why nasal coarticulation is increased while coarticulation for other consonant-vowel sequences is decreased during the production of hyperarticulated listener-oriented speaking styles (Guo and Smiljanic, 2021).

1pSC17. Enhancing secondary voicing cues in words with minimal pairs. Justin Bai (Linguist, Univ. of Colorado Boulder, Boulder, CO) and Rebecca Scarborough (Linguist, Univ. of Colorado Boulder, 295 UCB, Dept. of Linguist, Boulder, CO 80309, rebecca.scarborough@colorado.edu)

Lexical competition conditions phonetic variation. For example, words with more phonological neighbors are produced with more hyperarticulated vowels than words with fewer neighbors, and /p,t,k/-initial words with voiced stop minimal pairs have longer VOT. Such effects enhance contrastive features and may be aimed at improving perceptibility of more potentially confusable words. Secondary features may also help distinguish words, as with lengthened vowels that cue coda consonant voicing in English. Our study investigates whether secondary features are enhanced in the production of lexically confusable words, comparing vowel duration in words with final voiced consonants between words with and without minimal pair competitors (e.g., *bag* with minimal pair *back* vs. *gag* with no pair *gack*). Fifteen American English speakers produced 30 monosyllabic English words with voiced final consonants, half with minimal pair competitors and half without (matched for frequency and neighborhood density). Prevoiced vowels were found to be longer in minimal pair words than in non-minimal pair words; interactions indicate that the effect is stronger for some vowels. These findings show that it is not just contrastive features, but secondary features too that are subject to enhancement in lexical competition contexts, since they too can contribute to successful lexical perception.

1pSC18. Production accuracy of word-initial consonants in Mandarin-speaking children with cochlear implants. Jing Yang (Commun. Sci. and Disord., Univ. of Wisconsin-Milwaukee, 2400 E Hartford Ave. Enderis 873, Dept. Commun. Sci. and Disord., Milwaukee, WI 53201, jyang888@gmail.com), Xianhui Wang (CSD, Ohio Univ., Athens, OH), Jue Yu (Tongji Univ., Shanghai, China), and Li Xu (CSD, Ohio Univ., Athens, OH)

The purpose of this study was to assess the accuracy of consonant production of children with cochlear implants (CIs) judged by naïve adult listeners. A total of 57 Mandarin-speaking children (22 with normal hearing and 35 with CIs) were recruited to produce a list of Mandarin words composed of 17 word-initial obstruent consonants in three different vowel contexts. A total number of 2628 tokens were generated and were divided into 10 subsets. One hundred Mandarin-speaking naïve adult listeners were recruited to identify the consonant productions through Gorilla, the online research platform. Each listener was randomly assigned to one subset. For each child speaker, the consonant productions were judged by 7–12 adult listeners and an average accuracy rate was calculated across all listeners for each consonant. The results revealed that the children with CIs showed lower accuracies and different confusion patterns on their consonant productions than the normal hearing controls. In particular, they demonstrated higher accuracy for stops but had major problems with the fricatives and affricates involved in the alveolar—alveolopalatal—retroflex postalveolar three-way sibilant contrast. Of the three places of the sibilant contrast, they showed the greatest difficulties for the alveolar sounds.

Session 1pUW

Underwater Acoustics and Acoustical Oceanography: Understanding and Representing Uncertainty in Underwater Acoustic Models II

Sean Pecknold, Cochair
DRDC Atlantic Research Centre, 9 Grove St, Halifax, B2Y3Z7, Canada

Martin Siderius, Cochair
Portland State Univ., 1600 SW 4th Avenue, Suite 260, Portland, OR 97201

Invited Papers

1:00

1pUW1. Quantifying uncertainty of shipping source levels for marine soundscape modeling. Alexander O. MacGillivray (JASCO Appl. Sci., 2305–4464 Markham St., Victoria, BC V8Z7X8, Canada, alex@jasco.com) and Laurie Ainsworth (ERM Consultants Canada Ltd., Vancouver, BC, Canada)

Soundscape modeling is increasingly being used in assessments of marine habitat quality and to inform marine spatial planning. In most locations of interest, noise from shipping traffic and other marine vessels is the dominant anthropogenic contributor to the underwater soundscape. The noise contributions of marine vessels are typically calculated using tracking data from sources such as the automated identification system (AIS) and the vessel monitoring system (VMS). However, source levels of individual vessels in these datasets are often highly uncertain and must be estimated from incomplete information using data-driven models. Such models may be used to reliably predict mean source levels for large aggregations of vessels, but comparisons with real datasets show substantial statistical variability about the mean. Recent work, carried out using a large database of vessel noise measurements from the Enhancing Cetacean Habitat and Observation (ECHO) Program, has been used to develop statistical models that predict source levels based on ship type, design characteristics, and operating conditions. These models include a predictive component, based on observed trends, as well as a random component based on the residual variability of the data. This random component can be used to quantify the uncertainty of vessel noise emissions for soundscape modeling.

1:20

1pUW2. Uncertainty reduction in matched field inversion using Gaussian Processes. 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)

Gaussian Processes have been shown to be an effective tool for predicting an acoustic field in the ocean at a densely populated virtual array. This property is employed in this work for pre-processing acoustic data before these are used for source localization and geoacoustic parameter estimation using matched-field inversion (MFI). The process increases the accuracy of MFI as uncertainty in the estimation process is reduced. Via the application of Gaussian Processes, the data are denoised and interpolated and the new, enhanced fields are used as input to MFI along with replicas calculated at the virtual receivers at which field predictions are made. Kernel functions, implicit in the implementation of Gaussian Processes, quantify the correlation among field values at neighboring spatial points. Employing different kernels, the approach is tested on synthetic and real data with both an exhaustive search and genetic algorithms and is found to be superior to conventional Bartlett MFI in source localization and geoacoustic inversion. [Work supported by ONR.]

1:40

1pUW3. Quantifying and transferring environmental uncertainties in underwater acoustic modeling. Stan Dosso (School of Earth and Ocean Sci., Univ. of Victoria, SVictoria, BC V8W 2Y2, Canada, sdosso@uvic.ca) and Dag Tollefsen (Norwegian Defence Res. Establishment, Horten, Norway)

Uncertainties in environmental inputs represent a major source of uncertainty in underwater acoustic model outputs and applications thereof (e.g., transmission-loss estimation, source localization). Seabed geoacoustic parameters are often estimated by the inversion of ocean acoustic data. Hence, rigorous quantification of geoacoustic inversion uncertainties and the transfer of these uncertainties to modeling applications are of key importance. Uncertainty estimation in geoacoustic inversion is naturally accommodated in a Bayesian formulation, which combines data and prior information to form the posterior probability density (PPD) of seabed parameters. Important components of this approach include quantitative model selection for seabed parameterizations consistent with the information content of the data; an appropriate model for residual data errors that specifies the likelihood function; and nonlinear estimation of the PPD, which is normally carried out using Markov-chain Monte Carlo (MCMC) methods. MCMC characterizes the PPD using a large ensemble of dependent random samples of the geoacoustic parameters, which can be computationally demanding. However, these uncertainties can be transferred efficiently to subsequent propagation-modeling applications using a much-smaller, randomly chosen

(independent) subset from the ensemble. The approach is illustrated using simulations and inversion of ship noise recorded on a horizontal array of hydrophones at the New England Mud Patch.

2:00

1pUW4. The sonar equation versus reality: Inherent non-environmental variability in detection range. Ronald Kessel (Seamount Analytics, 34 Muriel St., Ottawa, ON K1S4E1, Canada, rkessel@seamountanalytics.ca)

Acoustic Range Prediction uses the sonar equation to forecast the "*detection range*," which is naturally understood by many to be the *expected* detection range in a closing-target encounter. Its *variability* is then the difference between actual and forecast detection ranges. The difference is often attributed to imperfect representation of the ocean environment (propagation and background levels). But the sonar equation's original design-orientation and its terminology have at times caused misunderstanding. It is shown that the sonar equation's detection range marks the sweet-spot of coverage, in which target detection is expected to be easy (fast and reliable), but which generally cannot be interpreted as the expected range or outer limits of detection in closing-target encounters. The classical and expected detection ranges concepts are disentangled here, giving each its appropriate roles in design, planning, and operations. It is also shown that, quite apart from environmental variability, there remains an inherent, significant variability of detection range simply because (1) the probability of detection increases slowly for a distant closing target, and (2) the target depth is unknown (unalerted detection). A significant degree of coverage variability therefore generally remains even if the environmental knowledge is perfect.

2:20–2:35 Break

Contributed Papers

2:35

1pUW5. Understanding uncertain acoustic propagation in deep ocean environments for feature engineering in corresponding machine learning tasks. Brandon M. Lee (Mech. Eng., Univ. of Michigan, 1231 Beal Ave. Ann Arbor, MI 48109, leebm@umich.edu), Jay R. Johnson, and David R. Dowling (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

Transmission loss (TL) predictions obtained using models of deep ocean environments are often uncertain due to imperfect knowledge of environmental properties such as sound speed, bathymetry, and seabed properties. These environmental uncertainties can be transferred to TL-prediction uncertainty by Monte Carlo (MC) sampling over environmental parameters and performing TL-field calculations to obtain an MC probability density function (PDF) of TL. Unfortunately, thousands of TL-field calculations are often required to quantify the TL uncertainty making this approach ill-suited to real-time applications. In an alternative, supervised learning approach, neural networks can be trained to quickly estimate the MC PDF of TL at a point of interest by analyzing the variability in the values of a baseline TL-field prediction within a region surrounding that point. The size, shape, and number of local TL region(s) used as inputs to the neural network can be reengineered by better understanding the means by which the uncertainties in environmental properties affect the TL uncertainty. This process and the resulting improvements in predictive performance are demonstrated for acoustic frequencies of 50 to 600 Hz, and source-receiver ranges up to 100 km. [Work sponsored by an NDSEG Fellowship.]

2:50

1pUW6. New England Shelf Break Acoustic (NESBA) experiment: Stochastic acoustic prediction. Bill Stevens (Portland State Univ., San Diego, CA), Martin Siderius (Portland State Univ., 1600 SW 4th Ave., Ste. 260, Portland, OR 97201, siderius@pdx.edu), and Matthew Carrier (Code 7321, Naval Res. Lab., Stennis Space Ctr., MS)

Sonar performance prediction has been a topic of interest for decades. Predictions are based on acoustic propagation modeling that is typically deterministic, i.e., lacking quantified confidence or uncertainty bounds. While this may be adequate in some locations, in oceanographically complex regions it can result in misleading predictions and conclusions that sonar performance predictions are unreliable. A better approach includes explicit calculations of the underlying source, receiver, and environmental uncertainties to provide a prediction confidence level. In oceanographically complex areas, the resulting uncertainties are likely to be high; however, knowledge of this allows uncertainties to be managed. The NESBA Signals and Noise experiment was conducted in April–May 2021. The goal of the experiment was to assess the potential for sonar prediction effectiveness gains given improved environmental awareness. This talk addresses the NESBA sub-goal of demonstrating that high-resolution ocean modeling with ensemble forecasts (in this case, Regional NCOM) could be leveraged for stochastic acoustic predictions, resulting in increased accuracy in performance predictions with quantified uncertainty bounds. NESBA results will be presented leveraging the high-resolution ocean model forecasts with ensembles run concurrently with NESBA and the numerous temperature/salinity profiles taken during the experiment. [Work supported by the Office of Naval Research.]

3:05

1pUW7. Layered and gradient model parameterizations in geoacoustic inversion. Stan Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC V8W 2Y2, Canada, sdosso@uvic.ca) and Julien Bonnel (Woods Hole Oceanographic Inst., Woods Hole, MA)

This paper considers the importance of model parameterization in geoacoustic inversion and uncertainty estimation, including quantitative approaches to model selection as well as potential limitations of the information content of acoustic data to determine the form of geoacoustic profiles, e.g., differentiating between layered and gradient structures. In particular, general parameterizations are considered based on trans-dimensional (trans-D) inversion, which represents profiles as an unknown number of uniform layers, and Bernstein polynomial (BP) inversion, which represents smooth gradients as polynomials of unknown order. These approaches are illustrated and compared for the inversion of high-order modal-dispersion data collected at the New England Mud Patch. It is shown that while the data constrain the sound-speed profile in the mud layer to reasonably high precision, the data cannot differentiate between trans-D layered or BP gradient representations. However, simpler (fixed) parameterizations, such as a homogeneous layer or linear gradient, can be ruled out based on the Bayesian information criterion. Furthermore, the prior choice of parameterization (layers or gradient) has implications on whether the sound-speed ratio at the water–sediment interface is estimated to be less or greater than one with high probability (an issue other acoustic datasets may share). [Work supported by the Office of Naval Research]

Session 1eID

Interdisciplinary: Keynote Lecture

Maureen Stone, Chair
University of Maryland School of Dentistry, Baltimore, MD 21201

Chair's Introduction—4:00

Introduced by: Timothy K. Stanton, Woods Hole Oceanographic Institution, Woods Hole, MA

Keynote Lecture

4:05

1pID. Understanding echoes. Wu-Jung Lee (Applied Physics Laboratory, University of Washington, Seattle, WA 98105)

By sending out sounds and analyzing the returning echoes, both humans and animals use active acoustic sensing systems to probe and understand the environment. High-frequency sonar systems, or echosounders, are the workhorse for observing fish and zooplankton in the ocean. Toothed whales and bats navigate and forage via echolocation in the air and under water. In this talk, I will discuss our work with both engineered and biological sonar systems to enable effective extraction and interpretation of information embedded in the echoes. We are developing data-driven methodologies and open-source software tools to tackle challenges imposed by large volumes of echosounder data rapidly accumulating across the global ocean. Using echolocating toothed whales as a model, we are combining experimental and computational approaches to understand biological processing of echo information. Throughout the talk, I will highlight the pivotal role of collaboration in my professional and personal development, and discuss efforts by colleagues and myself to cultivate a sense of community in our field.

Meeting of Accredited Standards Committee (ASC) S2 Mechanical Vibration and Shock

J. T. Nelson, Chair ASC S2
Wilson Ihrig & Associates, Inc., 6001 Shellmound St., Suite 400, Emeryville, CA 94608

A. P. Nash, Vice Chair ASC S2
Charles M. Salter Associates, 130 Sutter St., Suite 300, San Fransisco, CA 94101

Accredited Standards Committee S2 on Mechanical Vibration and Shock. Working group chairs will report on the status of various shock and vibration standards currently under development. Consideration will be given to new standards that might be needed over the next few years. Open discussion of committee reports is encouraged.

People interested in attending the meeting of the TAG for ISO/TC 108, Mechanical vibration, shock and condition monitoring, and four of its subcommittees, take note that meeting will be held in conjunction with the Standards Plenary meeting at 9:15 a.m. on Tuesday, 24 May 2022.

Scope of S2: Standards, specification, methods of measurement and test, and terminology in the field of mechanical vibration and shock, and condition monitoring and diagnostics of machines, including the effects of exposure to mechanical vibration and shock on humans, including those aspects which pertain to biological safety, tolerance and comfort.

Session 2aAA

Architectural Acoustics: Acoustic Comfort in Healthcare Facilities I

Lucky S. Tsaih, Cochair
Dept. of Architecture, National Taiwan Univ. of Sci. and Tech., 43 Keelung Rd., Sec. 4, Taipei 10607, Taiwan

Joanne Solet, Cochair
Harvard Medical School, Cambridge, MA 02138

Chair's Introduction—8:25

Invited Papers

8:30

2aAA1. Applying unsupervised machine learning clustering techniques to hospital soundscapes. Kenton Hummel (Architectural Eng., Univ. of Nebraska - Lincoln, 1110 S. 67th St., Omaha, NE 68182, khummel@huskers.unl.edu), Erica E. Ryherd (Architectural Eng., Univ. of Nebraska - Lincoln, Omaha, NE), and Bethany Lowndes (Neurological Sci., Univ. of Nebraska Medical Ctr., Omaha, NE)

Noise in hospitals can be problematic for both patients and staff and is consistently rated poorly on national patient satisfaction surveys. Previous studies have linked negative outcomes of hospital noise to numerous patient and staff challenges, such as reduced sleep and disrupted communication. Existing articles and guidelines commonly use equivalent sound pressure level as a primary noise metric. Additional insights into typical sound levels experienced by occupants can be found through more detailed statistical analyses of sound, such as by applying unsupervised machine learning clustering techniques. In this talk, clustering techniques will be explored in an effort to provide a more detailed analysis of the soundscape and various patterns of room activity. Noise data collected in three adult, inpatient hospital units will be analyzed using clustering techniques and compared against patient satisfaction scores. This more thorough, statistical characterization of the hospital soundscape can lead to better understanding of patterns of noise conditions and resultant occupant perceptions.

8:50

2aAA2. Sound quality metrics for improved prediction of hospital sleep disruption. S. Hales Swift (Photonic and Phononic Microsystems, Sandia National Labs., P.O. Box 5800, MS 1082, Albuquerque, NM 87123-1082, shswift@sandia.gov) and Joanne Solet (Div. of Sleep Medicine, Harvard Med. School, Boston, MA)

Most policy, including for the healthcare environment, uses A-weighted decibels as a standard metric. A-weighting is, thus, treated as a catch-all intended to represent all or most relevant human factors. However, for protecting sleep in the healthcare environment and its attendant recuperative benefits, A-weighted metrics neglect a great deal of essential information. To illustrate this, a dataset from a prior study of sleep disruption resulting from hospital noises measured using polysomnography [Sleep disruption due to hospital noise, Buxton *et al*. (2012)] is reanalyzed using measures of loudness, sharpness, and tonality. Linear combinations of these variables result in improved prediction of sleep state transitions relative to the A-weighted measure initially employed. Sound quality metrics can provide additional insight into how best to ensure a sonic environment conducive to a good night's sleep and can, thus, help decision makers build an environment that delivers restorative sleep in the service of restoration of health. [Work supported by SAND2022-0675 A.]

9:10

2aAA3. Acoustic comfort in health care facilities: Perspective of FGI-Facility Guidelines Institute. Kurt Rockstroh (Facility Guidelines Inst., 4141 N Scottsdale Rd., Ste. 150, Scottsdale, AZ 85251, kurtrockstroh@gmail.com), David Sykes (Facility Guidelines Inst., Boston, MA), and Mandy Kachur (Facility Guidelines Inst., Ann Arbor, MI)

Acoustic comfort is a welcome concept for the design and management of healthcare facilities, potentially providing methods to address acoustic environment deficiencies revealed by experience and by the Affordable Care Act's CAHPS® Hospital Survey, on which the noise question historically performs worst yielding public disclosure and financial penalties to hospitals. One possible vehicle for introducing quantified acoustic comfort into buildings is the healthcare design guidelines published by the Facility Guidelines Institute (FGI), which are adopted as building code in a majority of the United States, provide reference standards for the Leadership in Energy and Environmental Design (LEED) Rating System and the International Green Construction Code, and are cited in 87 countries. For acceptance into the FGI guidelines, "acoustic comfort" requires a formal definition accepted by standards organizations and clinical research, which is free from conflicts of interest based on research conducted by recognized third-party organizations on the physiological and psychological effects of noise on humans, particularly those with compromised health, like patients in hospitals and skilled

nursing facilities. FGI encourages proposals from the public, particularly valuing advice from members of professional societies, and relies on the research community to provide evidence-based support for all acoustics topics.

9:30

2aAA4. Integrated healthcare acoustics for patients, patient advocates, and staff in environments of care. Glenn E. Sweitzer (None, 4504 N. Hereford Dr., Muncie, IN 47304-1235, glenn.sweitzer@gmail.com)

COVID surges have demonstrated worldwide that overloaded critical care facilities cannot transform environments of care rapidly enough to satisfy patient, patient advocate, or staff expectations. Healthcare acoustics is identified as a vital yet understudied factor, linking the front lines of healthcare to patient well-being. Acoustic concerns are categorized, and existing architectural hardware and software means and proposed strategies are here discussed to be applied in both new and existing critical care facilities. Priority concerns focus on staff verbal and machine communications between single patient rooms with integral toilet/shower, adjacent corridors, nurse stations, and remote staff locations. The intent is to maintain uniformly high patient/staff communications and relevant documentation (e.g., for continuity of care, and liability) across all shifts. Integrated visual and acoustic communications between staff and equipment are explored, including projected machine display images minus noisy signals. Staff speech, background masking, and messaging to patients are considered via an "acoustic pillow," and via a wireless headset for staff, permitting hands-free focus on empathetic patient care. HVAC can be supplied via low-velocity displacement air, served via perimeter floor ducting, and exhausted via toilet/showers. Integrated healthcare acoustics, arguably, can reduce patient and staff stresses critical to recovery and retention, respectively.

9:50–10:10 Break

10:10

2aAA5. Acoustic comfort in pediatric noninvasive exam: A study of bedside renal sonography for children. Yi-Hsuan Tang (Dept. of Pediatrics, Div. of Pediatric Immunology and Nephrology, Inst. of Emergency and Critical Care Medicine, College of Medicine, Taipei Veterans General Hospital, National Yang Ming Chiao Tung Univ., No.201, Sec. 2, Shipai Rd., Beitou District, Taipei 11217, Taiwan, yishuanthebest@gmail.com), Julie Chia-Ping Chen, Lucky S. Tsaih (Dept. of Architecture, National Taiwan Univ. of Sci. and Tech., Taipei, Taiwan), Hsin-Hui Wang (Dept. of Pediatrics, Div. of Pediatric Immunology and Nephrologŷ, Dept. of Pediatrics, School of Medicine, and Inst. of Emergency and Critical Care Medicine, College of Medicine, Taipei Veterans General Hospital, National Yang Ming Chiao Tung Univ., Taipei, Taiwan), Chien-Hung Lin (Dept. of Pediatrics, Div. of Pediatric Immunology and Nephrology, Dept. of Pediatrics, School of Medicine, College of Medicine, Taipei Veterans General Hospital, National Yang Ming Chiao Tung Univ., Taipei, Taiwan), and Hui-Lan Chen (Dept. of Pediatrics, Div. of Pediatric Immunology and Nephrology, Inst. of Emergency and Critical Care Medicine, College of Medicine, Taipei Veterans General Hospital, National Yang Ming Chiao Tung Univ., Taipei, Taiwan)

Pediatric renal sonography is a non-radiation exposure and non-invasive procedure to evaluate kidneys and urinary tract structures. It is frequently performed in clinics and wards. Although there is no physical discomfort during ultrasounds, children would still experience agitation or crying due to unfamiliarity with medical environment. To provide comfortable medical experience and improve the quality of healthcare, we design a renal sonography process integrated with acoustic components specific for children. Patients under 7 years old are divided into three groups, under the age of 1, 1–3, and above 3. Pre-recorded music with nursery rhymes and fairy tales is played. Two time points are designed to evaluate acoustic integrated effect, along with preparing phase or real time of sonography starting. Data are collected from three timeframes: before, during, and after the exam. Physiological data will include heart rate, respiratory rate, and O2 saturation. Emotional behavior, such as agitation, crying, and smiling, will be recorded. This study will establish specific acoustic components and suitable intervention timeframe for children of different age groups. By having acoustic comfort, we hope children will have good experiences during sonography exam. As a result, the quality of medical examination and children wellbeing can be achieved.

10:30

2aAA6. Impact of sound intervention on patient wellbeing during pediatric peritoneal dialysis. Hsin-Hui Wang (Dept. of Pediatrics, Div. of Pediatric Immunology and Nephrology, No.201, Sec. 2, Shipai Rd., Beitou District, Taipei 11217, Taiwan, hhwang@vghtpe.gov.tw), Yi-Hsuan Tang (Dept. of Pediatrics, Div. of Pediatric Immunology and Nephrology, Taipei, Taiwan), Lucky S. Tsaih (Dept. of Architecture, National Taiwan Univ. of Sci. and Tech., Taipei, Taiwan), Julie Chia-Ping Chen (Dept. of Architecture, National Taiwan Univ. of Sci. and Tech., Taipei, Taiwan), Chien-Hung Lin (Dept. of Pediatrics, Div. of Pediatric Immunology and Nephrology, Taipei, Taiwan), and Hui-Lan Chen (Dept. of Pediatrics, Div. of Pediatric Immunology and Nephrology, Taipei, Taiwan)

Peritoneal dialysis (PD) is a treatment suitable for pediatric patients for less food restrictions, better school attendance and requires less hospital visits than hemodialysis. However, it may still cause stress and affect children's wellbeing as children with end-stage renal disease have to undergo treatment four times a day, 30 min each time. Thus, sound interventions used to improve children's comfort during PD treatments are worthy of investigating and are the aim of this study. Respondents will include PD patients under 18 years of age. Three types of sound will be played during PD treatment via headphone, including natural sound, classical music, and songs chosen by individual patient. Vital signs will be measured with oximeter to include heart rate, respiratory rate, O2 saturation, and blood pressure. Neurological activities will be recorded by electroencephalography (EEG). PD treatment efficiency will be measured by calculating fluid removal volume. Questionnaires will be given to measure the subjective wellbeing of respondents. Correlations among chosen music, vital signs, EEG readings, treatment efficiency, and patient's subjective wellbeing will be investigated. The findings of this research will add knowledge to improve subjective patient wellbeing and dialysis efficiency by acoustic interventions.

10:50

2aAA7. Relaxation response with gentle stream sound: A study using brainwave. Jennifer Tantra (Dept. of Architecture, National Taiwan Univ. of Sci. and Tech., Taipei, Taiwan), Shiang-I Juan (Taiwan Bldg. Technol. Ctr., National Taiwan Univ. of Sci. and Technol., 43 Keelung Rd. Sec. 4, Taipei 106, Taiwan, ukla2005@hotmail.com), Lucky S. Tsaih, and Wei-Hwa Chiang (Dept. of Architecture, National Taiwan Univ. of Sci. and Tech., Taipei, Taiwan)

Respite place in healthcare facility has been introduced and promoted in USGBC LEED to reduce stress and fatigue for caregivers, patients, and their families. Studies showed that people stay in natural environment can experience a sense of wellbeing and also suggested gentle stream water sound is highly associated with "relax" listening impression. Thus, as gentle stream is one of the familiar nature elements and is often used as a landscape architecture element for relaxation along with greens in respite place, the objective of this study was to learn the effect of the relaxation state with gentle stream water sound by studying the alpha brainwave response and physiological responses such as heart rate and blood pressure of 20 participants. During the evaluation, stress inducing period and relaxation period were conducted. Two-sample t-test was used to find out if there were any significant changes in participants' brainwave and physiological responses. The results showed that based on alpha brainwave, more female participants experienced relaxation state than males. After listening to the water sound, the systolic blood pressure showed the most obvious changes, and heart rate is slower.

11:10

2aAA8. Acoustical measurements and user evaluations of three new and existing ICUs. Gary W. Siebein (Siebein Assoc.,Inc., 625 NW 60TH St., Ste. C, Gainesville, FL 32607, gsiebein@siebeinacoustic.com), Lesa Lorusso (Healthcare, Gresham Smith, Nashville, TN), Jennifer Miller, Matthew Vetterick, and Ian Jones (Siebein Assoc.,Inc., Gainesville, FL)

Acoustical measurements were taken in three existing ICUs in a hospital. New ICUs were designed to address acoustical and other issues identified in the three existing ICUs. The results of acoustical measurements made in the new and existing spaces are compared to the results of surveys and focus group discussions with user groups including hospital staff, patients, and visitors. While sounds of individual acoustical events were similar in the new and existing ICUs, the overall sound levels and frequency of sounds decreased in the new ICUs compared to the existing ICUs. This is because the design of the new ICU removed the sounds of nurses speaking at the nurses' station, medical equipment, and other operations from the patient areas in the new ICU. Overall, the new ICU had a quieter environment for patients, nurses, and visitors as indicated by the acoustical measurements, surveys, and focus group discussions.

11:30

2aAA9. Designing personalized soundscape for care facilities. Arezoo Talebzadeh (Ghent Univ., 126 Tech Ln. Ghent Sci. Park, Ghent 9052, Belgium, arezoo.talebzadeh@ugent.be) and Dick Botteldooren (Information Technol., Ghent Univ., Gent, Belgium)

Sound is a critical element in making people aware of their environment; it gives people a sense of place and time. Soundscape relies not only on the objective quality of sound by quantifying the sound level but also on the subjective quality of the auditory environment based on people's perceptions. When a sonic environment is unfamiliar, it adds to the anxiety of those who receive the sound. People with dementia may have difficulty understanding time and space; they may live in long-term care (LTC) facilities or have to relocate to LTC to reduce care responsibilities from their families. Sensory perception in these facilities is unfamiliar: light, sound, temperature, and smells may be strange. Unfamiliar sensory stimuli add to residents' anxiety and annoyance, making them agitated and disturbed and decreasing their sleep quality. Designing a pleasant and personalized soundscape helps reduce BPSD (behavioural and psychological symptoms of dementia) and improves sleep quality. This study employed a sound system that uses a customized algorithm to play sounds at prescheduled moments throughout the day. The soundscape is composed according to the resident's daily activities and is delivered in their room, using familiar sounds like bird sounds and raindrops. The goal is to implement soundscape into the design of LTC facilities to enhance the quality of life for residents and their caregivers.

Session 2aAB

Animal Bioacoustics, Acoustical Oceanography, and Underwater Acoustics: Whitlow Au Memorial Session I

Kelly Benoit-Bird, Cochair
Monterey Bay Aquarium Research Institute (MBARI), 7700 Sandholdt Road, Moss Landing, CA 95003

Marc Lammers, Cochair
Hawaii Institute of Marine Biology, 46-007 Lilipuna Rd., Kaneohe, HI 96744

Wu-Jung Lee, Cochair
Applied Physics Laboratory, University of Washington, 1013 NE 40th St., Seattle, WA 98105

Chair's Introduction—8:20

Invited Paper

8:25

2aAB1. Launching underwater bioacoustics in Asia by Whitlow W. L. Au. Tomonari Akamatsu (Ocean Policy Res. Inst., The Sasakawa Peace Foundation, 1-15-16 Toranomon, Tokyo, Minato-ku 105-8524, Japan, akamatsu.tom@gmail.com)

Whitlow W. L. Au combined biosonar science with engineering. His studies influenced so many students in Asia no matter what their specialities, such as physiology, morphology, and physics. The epoch-making book "The Sonar of Dolphins" is an ideal example of his interdisciplinary approach. Whit educated young researchers in various ways even he did not notice. He led discussion among scientists that motivated current and new generations to conduct underwater and terrestrial bioacoustic research in the lab, in the forest and in the ocean. Quite a few attendees were from Asia at the series of biosonar symposiums in 1991 Moscow, in 1994 Harderwijk, and in 1998 Algarve, but the symposium held in 2009 Kyoto accommodated nearly 200 international specialists and half of them were Asian. An example of his influence was the towed and fixed passive acoustic monitoring in the Yangtze River, China, which revealed a range-wide distribution of finless porpoises collaborating with an international team. These techniques were applied later in India, Thailand, Malaysia, Nepal, Taiwan, Hongkong, and many other areas including Japan for the impact assessment of offshore wind farm developments. We owe him all the outcomes of underwater biosonar studies in Asia.

Contributed Papers

8:50

2aAB2. Tribute to Whitlow Au. Ronald A. Kastelein (Res., SEAMARCO, Julianalaan, 46, Harderwijk 3843CC, Netherlands, rk@seamarco.nl)

Whitlow Au has played a large role in my life both as a friend, as editor of manuscripts I submitted to JASA and a book on bycatch of harbor porpoises, as co-authors of manuscripts we submitted together to JASA, as author of manuscripts Whit submitted to sensory system books of which I was a co-editor. Whit also proposed me as a Fellow of the Acoustical Society of America. We collaborated on several research project, which were conducted in pools at the cetacean rehabilitation center of Dolfinarium Harderwijk, the Netherlands, in a large floating pen at Neeltje Jans, the Netherlands, at the Hawaii Institute for Marine Biology, Oahu, USA, in a large fish pool at Jacobahaven, and at the Netherlands and at SEAMARCO Research Institute, Wilhelminadorp, the NL. The publications in Whit and I are co-authors are used as points of departure of the talk. The studies are described in chronological order, and the gist of each study is given.

9:05

2aAB3. Spatio-temporal patterns of blue and fin whale sound production identified in National Oceanic Atmospheric Administration-National Park Service Ocean Noise Reference Station Network recording sites in the Pacific Ocean. Emma Pearson (Univ. of Colorado Boulder, Boulder, CO 80309, emma.pearson@colorado.edu) and Carrie Wall (Cooperative Inst. for Res. in Environ. Sci., Univ. of Colorado, Boulder, CO)

The ability of sound to travel long distances, especially in lower frequencies, makes it an effective way for organisms to communicate in marine environments. Passive acoustic monitoring (PAM) is an essential method to study low-frequency sound in marine ecosystems. The NOAA-NPS Ocean Noise Reference Station Network (NRS) project is a long-term study that utilizes PAM to monitor low-frequency (<2kHz) soundscapes throughout the U.S. exclusive economic zone. While these data have been analyzed to understand large scale patterns, assessment of specific sound sources is an area where these datasets can provide new insights. This study used a call energy index (CEI) to assess blue whale and fin whale sound production from three NRS recording sites in the Gulf of Alaska, off the Olympic coast, and near the Channel Islands between 2014 and 2020. We present the diel and seasonal patterns detected for both species and compare them to the decidecade bands assessed for the recording sites. Understanding the song production and migration patterns for endangered blue whales and fin whales is essential for effective conservation efforts. By utilizing PAM and efficient detection methods such as CEI, researchers gain the ability to process large amounts of bioacoustic data and better understand the migratory behaviors of endangered marine species.

9:20

2aAB4. Bottlenose dolphin whistle repertoires: Size and stability over time. Julie Oswald (Univ. of St. Andrews, Scottish Oceans Inst., East Sands, St. Andrews, Fife KY16 8LB, United Kingdom, jno@st-andrews.ac.uk), Peter Sugarman (Humans and Dolphins Talking, LLC, Bellevue, WA), Elizabeth L. Ferguson (Ocean Sci. Analytics, San Diego, CA), Sam F. Walmsley, and Vincent M. Janik (Univ. of St. Andrews, St. Andrews, United Kingdom)

Bottlenose dolphins possess vocal learning abilities that influence the development of individually distinctive signature whistles. While signature whistles have been studied in detail, little is known about other whistle types in bottlenose dolphin communication or the size of their whistle repertoires. We made 24-hour acoustic recordings from a group of 13 bottlenose dolphins at Oceanogràfic (Valencia, Spain) for two months to determine the whistle repertoire size of this group and investigate whether learning leads to changes in existing whistle types over time. We extracted fundamental frequency contours from 50 randomly chosen whistles per day ($n = 3{,}119$ whistles) and categorised them using ARTwarp (96% vigilance level), resulting in 701 whistle types. The whistle type discovery curve did not plateau after two months, indicating that we did not capture the entire repertoire. Three analytical methods were used to estimate repertoire size (curve-fitting, a capture-recapture model, and the coupon collectors method). These produced repertoire size estimates between 888 and 1358 whistle types. Whistle types appeared and disappeared over time; however, exact randomization tests showed no significant differences between the observed patterns of appearance and disappearance and time-randomized permutations. Our results suggest that these dolphins possess a large whistle repertoire that is stable over time.

9:35

2aAB5. Assessing intra-individual consistency in humpback whale song production using animal-borne acoustic recorders. Julia Zeh (Dept. of Biology, Syracuse Univ., 107 College Pl, Syracuse, NY 13210, jzeh01@syr.edu), Marc Lammers (Hawaiian Islands Humpback Whale National Marine Sanctuary, Kihei, HI), Adam Pack (Departments of Psych. and Biology, Univ. of Hawai'i at Hilo, Hilo, HI), and Susan Parks (Dept. of Biology, Syracuse Univ., Syracuse, NY)

Studying individual variation in vocal behavior can provide insight into its functions, stability, and mechanisms. Collecting such data at the scale of the individual can be facilitated using animal-borne tags. Here, we use archival suction-cup acoustic recording tags to investigate intra-individual variation in male humpback whale song production. Humpback whale song is a complex and hierarchically structured vocal sequence of 4-7 repeated phrases that are comprised of different units. Repeated songs are termed song sessions. To investigate how consistent song production is within the song session of an individual whale, we deployed suction cup-attached acoustic recording tags on humpback whales in the Hawaiian Islands Humpback Whale National Marine Sanctuary during the 2018–2022 breeding seasons. We analyzed tag data from nine whales to assess intra-individual variation in song structure and syntax as well as differences in the acoustic properties of song units across each song iteration. Each tag contained between 4 and 24 song iterations, and song session recordings varied between 45 minutes and 5 hours. Across individuals, the most variable song iterations occurred at the beginning of a song session. All individuals showed variation in syntax and unit production throughout a session; however, some singers were more consistent than others.

9:50–10:05 Break

10:05

2aAB6. Integrating acoustics and video imaging to illuminate life in the mesopelagic. Kelly Benoit-Bird (Monterey Bay Aquarium Res. Inst. (MBARI), 7700 Sandholdt Rd., Moss Landing, CA 95003, kbb@mbari.org)

The complementary strengths of acoustic and *in situ* visual sampling for understanding ocean life have been evident from the earliest days of each technique. The "deep scattering layer," for example, was discovered using military sonars in the 1940s. This layer, thought at first to be the seafloor, was confirmed to be biological via *in situ* visual observations. This early example of combining these two powerful approaches highlights their complementary strengths (and weaknesses). Since then, acoustics and imaging have made great advances in both technology and application. Yet, they have been used together to address biological questions only infrequently, noting substantive contributions by Whitlow Au. Building on this foundation, we have been integrating acoustics and imaging to understand life in the mesopelagic. Used in a synergistic fashion, these tools allow us to describe the behavioral responses that animals exhibit to platforms and their lights, to measure the species-specific broadband "acoustic signatures" of pelagic animals needed to interpret remotely collected data, make measurements of animal size with a single camera, and examine how animal distributions are affected by the environment. In combining acoustics and imaging, there is power to observe biological phenomena in the ocean not fully resolvable with either technique alone.

10:20

2aAB7. Cetacean acoustic monitoring across the Hawaiian archipelago: Building on Whitlow Au's legacy. Marc Lammers (Hawaiian Islands Humpback Whale National Marine Sanctuary, 46-007 Lilipuna Rd., Kaneohe, HI 96744, lammers@hawaii.edu), Eden J. Zang (Hawaiian Islands Humpback Whale National Marine Sanctuary, Kihei, HI), Anke Kügler (Univ. of Hawai'i at Mānoa, Honolulu, HI), Jonathan Martinez (Papahānaumokuākea Marine National Monument, Albuquerque, NM), Karlina Merkens (NOAA Fisheries Pacific Islands Fisheries Sci. Ctr., Honolulu, HI), and Leila Hatch (Stellwagen Bank National Marine Sanctuary, Scituate, MA)

The Hawaiian islands are home to more than 20 species of cetaceans and are the principal breeding ground of the north Pacific humpback whale population. The archipelago stretches more than 2500 km from Hawaii Island to Kure Atoll, creating a significant challenge for monitoring the occurrence and distribution of cetaceans across such a vast range. To meet this challenge, the National Oceanic and Atmospheric Administration and the U.S. Navy have been engaged in a three-year effort to monitor the marine soundscape of the Hawaiian archipelago known as the SanctSound Project. Bottom-moored acoustic recorders were deployed at multiple locations across the Hawaiian Islands Humpback Whale National Marine Sanctuary and the Papahānaumokuākea Marine National Monument to examine the occurrence of humpback whales and odontocetes based on the relative prevalence of their acoustic signaling. Anthropogenic sound sources were also studied to understand how these co-occur with cetaceans. Substantial spatial and temporal variability was observed in the prevalence of whale song and dolphin acoustic activity across locations with high cetacean presence sometimes overlapped with elevated anthropogenic activity. This work helps expand our understanding of how cetaceans use the archipelago and builds on the legacy of Whitlow Au, who pioneered cetacean acoustic monitoring in Hawaii.

10:35

2aAB8. Insights into acoustic behavior of false killer whales. Pina Gruden (Cooperative Inst. for Marine and Atmospheric Res., Hawai'i Inst. of Geophys., 2525 Correa Rd., Honolulu, HI 96822, pgruden@hawaii.edu), Yvonne Barkley (Cooperative Inst. for Marine and Atmospheric Res., Honolulu, HI), and Jennifer L. McCullough (NOAA Fisheries, Pacific Islands Fisheries Sci. Ctr., Honolulu, HI)

The Hawaiian Archipelago is home to three distinct populations of false killer whales (*Pseudorca crassidens*), including one currently listed as endangered. These delphinids are known to interact with fishing gear, leading to whale mortality or injury. Hence, it is critical to assess the abundance

of these populations typically achieved through visual-based sighting surveys. However, these surveys are complicated by a number of biases and uncertainties specific to this species. Passive acoustics could aid in monitoring of their population status, but the knowledge is limited about the patterns in their acoustic repertoire and behavior, which hinders our ability to derive reliable acoustics-based abundance estimates. Here, we discuss insights into the acoustic behaviour of false killer whales in the wild, gained by simultaneous tracking of both narrowband whistles and broadband echolocation clicks from towed hydrophone arrays. The results indicate a diverse acoustic behaviour between different subgroups within the same encounter, where 23.8% of subgroups (N total = 408) only echolocate, 18.9% only whistle, and 57.3% emit both types of vocalizations. Such increased understanding of false killer whale vocal behavior can contribute information from passive acoustic data for management and conservation purposes.

10:50

2aAB9. Autumn acoustic behavior of right whales in Southern New England waters. Susan Parks (Dept. of Biology, Syracuse Univ., 107 College Pl., Syracuse, NY 13244, sparks@syr.edu), Julia Zeh (Dept. of Biology, Syracuse Univ., Syracuse, NY), K. Alex Shorter (Mech. Eng., Univ. of Michigan, Ann Arbor, MI), Heather Foley, Lisa Conger, and Danielle Cholewiak (Northeast Fisheries Sci. Ctr., NOAA Fisheries, Woods Hole, MA)

North Atlantic right whales (*Eubalaena glacialis*) are an endangered species of baleen whale found in high human use areas off the East Coast of the United States. Conservation efforts for this species include the use of passive acoustic monitoring to detect sounds produced by right whales to determine when they are present in areas of interest. Right whale acoustic behavior is known to vary by age, sex, and behavioral state, with differing call types and call rates found across different habitats and seasons. There are currently plans for development of offshore wind energy installations off the East Coast of the United States, including lease areas south of Massachusetts in areas known to be frequented by right whales. These development plans necessitate a better understanding of right whale acoustic behavior in this region to best inform passive acoustic monitoring efforts for right whales. In this study, we analyzed ~38 h of data from eight suction cup archival acoustic biologging tags attached to North Atlantic right whales in October 2021 in Southern New England waters south of Nantucket. The call types and call rates by behavioral state will be discussed with the primary observed behaviors including foraging and social surface behaviors.

TUESDAY MORNING, 24 MAY 2022 GOVERNORS SQUARE 14, 8:55 A.M. TO 12:00 NOON

Session 2aBAa

Biomedical Acoustics: New Developments in Lung Ultrasound I

Libertario Demi, Cochair
Information Engineering and Computer Science, University of Trento, Via Sommarive 9, Trento, 38123, Italy

Marie Muller, Cochair
MAE, North Carolina State University, 911 Oval Drive, Engineering Building III, Raleigh, NC 27606

Chair's Introduction—8:55

Invited Paper

9:00

2aBAa1. Relating lung pathology to ultrasound imaging: An experimentally validated simulation approach. Oleksii Ostras (Biomedical Eng., Univ. of North Carolina at Chapel Hill and North Carolina State Univ., Chapel Hill, NC), Igor Shponka (Dnipro State Med. Univ., Dnipro, Ukraine), and Gianmarco Pinton (Biomedical Eng., Univ. of North Carolina at Chapel Hill and North Carolina State Univ., 116 Manning Dr., Mary Ellen Jones Rm. 9212A, Chapel Hill, NC 27599, gia@email.unc.edu)

Lung ultrasound (LUS) imaging can be highly sensitive to disease. However, lung imaging depends on reverberation that occurs at the lung interface, which is complex and upends the conventional time-space relationship in delay-and-sum beamforming resulting in images that require the interpretation of artifacts. Establishing a clear link between ultrasound images and underlying alveolar or fibrotic state of the lung could improve the diagnostic accuracy and clinical deployment of lung ultrasound and potentially establish LUS as a gold-standard imaging modality. Here, it is shown how histology-derived acoustical maps of the lung, Visible Human maps of the

abdomen, and Fullwave simulations of ultrasound propagation can accurately model the multiple scattering physics at the lung interface. Lung B-mode images are generated based on the first principles of propagation and multiple scattering and they are compared to clinical imaging. *In silico* modifications of the aeration/porosity and the fluid-to-tissue ratio in the lung parenchyma are related to the corresponding changes in B-mode images. Additionally, the patterns of superficial/subpleural air inclusions were analyzed and mapped to corresponding B-mode image markers (white lung, single and multiple B-lines, A-lines). This establishes a validated framework for quantitative imaging of lung disease and the development of LUS-specific beamforming.

Contributed Papers

9:30

2aBAa2. Investigating the link between intensity of lung ultrasound vertical artifacts and penetration depth of ultrasound waves, *in silico* study. Federico Mento (Dept. of Information Eng. and Comput. Sci., Univ. of Trento, Via Sommarive, 9, Povo, Trento, Trento (TN) 38123, Italy, federico.mento@unitn.it) and Libertario Demi (Dept. of Information Eng. and Comput. Sci., Univ. of Trento, Trento, Italia, Italy)

Lung ultrasound (LUS) is nowadays widely applied for lung surface evaluation. In particular, LUS is often based on the analysis of vertical artifacts, which correlate with various pathologies but whose genesis is not fully comprehended yet. To better understand the phenomena causing these artifacts' appearance, numerical simulations can represent a powerful tool. Specifically, we present a simulation study (k-Wave MATLAB toolbox) that investigates whether a link exists between the intensity of these artifacts and the penetration depth of ultrasound waves through the lung parenchyma. The size of the simulated numerical domain is 4 cm $\times$ 2 cm, with the simulated lung surface located at a depth of 2 cm. Different alveolar structures were modeled by varying the alveolar diameter (from 40 to 800 μm), the inter-alveolar axial-spacing (from 40 to 790 μm) and inter-alveolar lateral-spacing (from 10 to 100 μm). For each configuration, Gaussian pulses (with bandwidth equal to 0.5 MHz at -6 dB) having different center frequencies (from 1 to 5 MHz) were transmitted without focusing (plane wave imaging). Results highlight how the artifacts' intensity seems to be independent from waves' penetration depth.

9:45

2aBAa3. Ultrasound vibro-elastography for assessing water content in a lung sponge phantom model. Xiaoming Zhang (Mayo Clinic, 200 1st ST SW, Rochester, MN 55905, zhang.xiaoming@mayo.edu) and Jinling Zhou (Mayo Clinic, Rochetser, MN)

Pulmonary edema is common in patients with congestive heart failure. Extravascular lung water (EVLW) correlates to the disease prognosis but the assessment is challenging. The current standard method, computed tomography (CT), poses a significant logistic burden and exposes patents to ionizing radiation. Non-invasive lung ultrasound (LUS) is introduced to subjectively evaluate B-lines for EVLW but results in significant inter-observer variability. We have developed ultrasound vibro-elastography (USVE) to safely measure lung surface wave speed quantitatively. In this abstract, we presented the technique for analyzing water content in a lung phantom model. A sponge is used as the lung phantom. Twelve water contents were injected in the phantom for different EVLW values. The mass density of the phantom was measured for each water content. Wave propagation was generated at five frequencies between 100 Hz and 300 Hz in the phantom, and the surface wave speeds were collected. The viscoelasticity was derived from the wave speed dispersion with frequency. The obtained surface wave speeds showed no clear correlation with different water content. The elasticity and viscosity, however, were found increased with water content. This research pinpoints the likelihood and possible direction for USVE to assess EVLW for patients with pulmonary edema.

Invited Papers

10:00

2aBAa4. Investigating response to treatment of pulmonary fibrosis in rats using ultrasound multiple scattering. Roshan Roshankhah (Mech. and Aerosp. Eng., North Carolina State Univ., Raleigh, NC), John Blackwell (Surgery, Univ. of North Carolina at Chapel Hill, Chapel Hill, NC), Hong Yuan (Radiology, Univ. of North Carolina at Chapel Hill, Chapel Hill, NC), Stephanie A. Montgomery (Pathol. and Lab Medicine, Univ. of North Carolina, Chapel Hill, NC), Thomas M. Egan (Surgery and Biomedical Eng., Univ. of North Carolina at Chapel Hill, Chapel Hill, NC), and Marie Muller (MAE, North Carolina State Univ., 911 Oval Dr., Eng. Bldg. III, Raleigh, NC 27606, mmuller2@ncsu.edu)

Lung alveoli constitute a complex distribution of strong ultrasound scatterers, leading to multiple scattering (USMS). Conventional ultrasound cannot be utilized to produce images that would accurately render lung structure. Pulmonary fibrosis affects lung microstructure by thickening alveolar walls, which changes wave diffusion and scattering patterns by modifying the distribution and size of scatterers. We present a method for the quantitative approach of structural changes in lung parenchyma based on diffusion of ultrasound waves, relying on measurement of the scattering mean free path (SMFP). We quantify severity of lung damage due to bleomycin-induced fibrosis in rats, and to monitor response to Nintedanib treatment by comparing the SMFP in 6 control (normal) lungs, 6 fibrotic lungs, and 6 fibrotic lungs from rats treated with Nintedanib. We observed significant differences in SMFP among control lungs (483 ± 50 μm), fibrotic lungs (1433 ± 612 μm), and lungs from Nintedanib-treated rats (835 ± 149 μm) (mean $\pm$ sd). Strong correlations were observed between SMFP and fibrosis severity score on inflated *ex vivo* CT lung images ($p = 0.076$, $r = 0.43$), as well as between SMFP and modified Ashcroft score of inflation-fixed lungs stained with H&E and Sirius red ($p = 0.008$, $r = 0.61$). This suggests SMFP may be useful to monitor response to treatment of pulmonary fibrosis.

10:30–10:45 Break

10:45

2aBAa5. *In vivo* assessment of pulmonary edema and fibrosis in rats using separation of single and multiple scattering contributions and quantitative ultrasound. Theresa H. Lye (Riverside Res., 156 William St., Fl. 9, New York, NY 10038, tlye@riversideresearch.org), Roshan Roshankhah, Aidan Turnbull (Mech. and Aerosp. Eng., North Carolina State Univ., Raleigh, NC), John Blackwell, Thomas M. Egan (Surgery and Biomedical Eng., Univ. of North Carolina at Chapel Hill, Chapel Hill, NC), Marie Muller (Mech. and Aerosp. Eng., North Carolina State Univ., Raleigh, NC), and Jonathan Mamou (Riverside Res., New York, NY)

The application of spectral quantitative ultrasound (QUS) and envelope statistical methods to lung ultrasound is complicated due to the presence of multiple scattering (MS). In this study, the relationship between QUS assessment of lung disease and scattering regime was explored by applying QUS to lung ultrasound data, where the single scattering (SS) and MS components were separated. Data were acquired by a Verasonics Vantage ultrasound scanner from 15 healthy rats, 19 edematous rats, and 20 fibrotic rats. SS and MS components of the data were separated by a random matrix theory approach. Spectral QUS and envelope statistical parameters were then estimated from the SS and MS data separately. Initial results were obtained using the healthy and pulmonary edema data, where the extracted QUS parameters were correlated to the wet-to-dry (W/D) weight ratio, a gold standard measurement of edema. Several QUS parameters were significantly correlated to W/D ratio for both the SS and MS data, and the highest correlation coefficient was 0.53 using the SS data and 0.48 using the MS data. These results demonstrate the utility of QUS for assessing lung disease and provide further insight on the effects of SS and MS on spectral QUS and envelope statistical parameters.

Contributed Papers

11:15

2aBAa6. Localizing pulmonary nodules for surgical resection using ultrasound multiple scattering. Roshan Roshankhah (Mech. and Aerosp. Eng., North Carolina State Univ., 1840 ENTREPRENEUR Dr., Raleigh, NC 27606, rroshan2@ncsu.edu), John Blackwell, Thomas M. Egan (Surgery and Biomedical Eng., Univ. of North Carolina at Chapel Hill, Chapel Hill, NC), and Marie Muller (Mech. and Aerosp. Eng., North Carolina State Univ., Raleigh, NC)

Using conventional ultrasound to image pulmonary nodules is elusive due to multiple scattering in highly heterogeneous lung tissue. It is possible to leverage multiple scattering as a source of contrast between nodules and healthy lung parenchyma, because lung nodules do not contain air-filled alveoli. We developed a method based on the separation of multiple and single scattering using singular value decomposition. When combined with a depression detection algorithm, this allows us to render a map of the regions exhibiting less multiple scattering, associated with the presence of nodules. These techniques allowed for localization of 55 out of 59 nodules placed in ten lungs, as small as 5 mm diameter. In this study, we are demonstrating the feasibility of the method to locate nodules with respect to a surgical stapler by merging the rendered nodule map with a B-mode image of the surgical device. Using Verasonics Vantage scanner with array transducer, inter-element responses matrices were acquired from *ex vivo* pig/dog lungs, in which artificial nodules were implanted. We show that the relative position of nodule to the stapler can be measured. By measuring the distance of the nodule to the surgical stapler, it is possible to resect the nodule with safe margins.

11:30

2aBAa7. Quantifying severity of lung fibrosis in rodents using random matrix theory. Azadeh D. Cole (MAE, North Carolina State Univ., Eng. Bldg. III (EB3) 3141, Raleigh, NC 27695, adashti@ncsu.edu), John Blackwell (Surgery and Biomedical Eng., Univ. of North Carolina at Chapel Hill, Chapel Hill, NC), Stephanie A. Montgomery (Dept. of Pathol. and Lab. Medicine, Univ. of North Carolina at Chapel Hill, Chapel Hill, NC), Thomas M. Egan (Surgery and Biomedical Eng., Univ. of North Carolina at Chapel Hill, Chapel Hill, NC), and Marie Muller (MAE, North Carolina State Univ., Raleigh, NC)

We investigate random matrix theory (RMT) as a tool to detect and quantify pulmonary fibrosis in rodents *in vivo*. Highly scattering structures such as lung alveoli result in specific characteristics in the distribution of singular values of the inter-element response matrix (IRM). When multiple scattering dominates (healthy lung), the distribution of singular values of the IRM is expected to follow a quarter circle law. However, when single scattering regime dominates, the singular values distribution is closer to a Henkel distribution. We propose to exploit this feature to detect pulmonary fibrosis and quantify its severity. Two metrics are defined to describe the singular value distribution: the expected value *E(x)*, which is the weighted average of all singular values, and, the singular value with the highest probability. A 128-element linear transducer operating at 7.8 MHz and a Verasonics scanner were used to collect IRMs from 6 normal and 18 rat lungs with bleomycin-induced fibrosis *in vivo*. Significant correlations were observed between *E(x)* ($r = -0.46$, $p = 0.02$) and ($r = 0.52$, $p = 0.01$) with the severity of fibrosis independently assessed by histology. These preliminary results show the potential of RMT metrics *E(x)* and to quantify structural changes in the lung parenchyma.

11:45

2aBAa8. Using ultrasound multiple scattering to determine severity of pulmonary edema. Thomas M. Egan (Surgery and Biomedical Eng., Univ. of North Carolina at Chapel Hill, 3040 Burnett-Womack Bldg., Chapel Hill, NC 27599-7065, thomas_egan@med.unc.edu), Roshan Roshankhah (Mech. and Aerosp. Eng., North Carolina State Univ., Raleigh, NC), John Blackwell (Surgery, UNC at Capel Hill, Chapel Hill, NC), Hong Yuan (Radiology, UNC at Chapel Hill, Chapel Hill, NC), Stephanie A. Montgomery (Lab and Animal Medicine, UNC at Chapel Hill, Chapel Hill, NC), and Marie Muller (Mech. and Aerosp. Eng., Biomedical Eng., North Carolina State Univ., Raleigh, NC)

Conventional ultrasound (US) cannot quantitatively evaluate lung tissue because of US multiple scattering (USMS). Pulmonary edema causes vertical artifacts -B-lines- which provide qualitative information about alveolar flooding. We showed that Scattering mean free path (SMFP), a measure of the density of air-filled alveoli, is longer in edematous than normal lungs. Here, we show that SMFP correlates with wet:dry weight ratio of lung tissue (W/D) and CT-scan assessment of edema. Ischemia-reperfusion injury (IRI) was created in lungs of anesthetized rats. The left lung hilum was clamped for 20, 40, or 60 minutes ($n = 6$/group) then reperfused for 60 minutes before ligating the apical portion of each lung to measure W/D, removing inflated lung blocks, then measuring SMFP and backscatter frequency shift (BFS) with a Verasonics scanner and L11-4v US probe. Six lung blocks were removed from healthy rats as Controls. Inflated lung blocks had *ex vivo* CT scans, followed by inflation-fixed histology. By logistic regression, there was a correlation between SMFP and W/D in 18 edematous and 12 Control lungs ($r^2 = 0.27$, $p < 0.004$) and a significant correlation between edema extent by CT and SMFP and W/D. BFS was larger, and histology confirmed edema in IRI lungs. SMFP may be useful to quantify lung edema.

Session 2aBAb

Biomedical Acoustics: For a Few Bubbles More: Recent Developments in Medical Ultrasound I

John S. Allen, Cochair
Mechanical Engineering, University of Hawaii Manoa, Holmes 302, 2540 Dole Street, Honolulu, HI 96822

Mark Borden, Cochair
Biomedical Engineering, University of Colorado, 1111 Engineering Drive, Boulder, CO 80309-0427

Chair's Introduction—9:00

Invited Papers

9:05

2aBAb1. Applying ultrasound phase-change contrast agents to guide therapeutic intervention in type 1 diabetes. Richard K. Benninger (Bioengineering, Univ. of Colorado Anschutz Med. Campus, Denver, CO, richard.benninger@cuanschutz.edu)

Ultrasound is a widely deployable and cost-effective clinical imaging modality. However, conventional microbubble contrast agents are restricted to the vasculature. Sub-micron nanodroplet (ND) phase-change agents can be vaporized into micron-sized bubbles, serving as a microbubble precursor. In principle, the enhanced permeabiltiy and retention effect can be used with sub-micron NDs to detect injured tissues. Methods to detect type1 diabetes (T1D) progression prior to clinical diagnosis are needed. T1D results from autoreactive T-cells infiltrating the islets of Langerhans, destroying insulin-producing beta-cells. Overt disease takes years to present and at diagnosis there is substantial beta-cells loss. Therapeutic intervention to preserve beta-cell mass is hampered by an inability to follow pre-symptomatic T1D progression and tracking whether therapeutic interventions are impacting disease progression. We will present ultrasound imaging of phase-change nanodroplet (ND) contrast-agent accumulation within the islet. ND accumulation is dependent on immune infiltration, therefore tracks pre-symptomatic T1D development and progression to diabetes. Measurement of ND accumulation detected pre-symptomatic T1D earlier and with greater sensitivity compared to existing measurements of circulating autoantibodies. This provides an opportunity to guide early therapeutic treatments to prevent T1D.

9:30

2aBAb2. Ultrasound-stimulated, drug-loaded bubbles for cancer therapy. Naomi Matsuura (Biomedical Eng./Mater. Sci. & Eng., Univ. of Toronto, 164 College St., Rm. 407, Toronto, ON M5S 3G9, Canada, naomi.matsuura@utoronto.ca)

Non-invasive, focused ultrasound in combination with microbubbles have previously been used for targeted drug delivery. Focused ultrasound-stimulated microbubbles can increase local tumour blood vessel permeability such that co-injected chemotherapeutics can diffuse preferentially into tumour tissue. Focused ultrasound and microbubbles have also been used to initiate other local bioeffects, for example, to damage tumour vessels for mechanically induced vascular disruption therapy, which can profoundly increase the efficacy of a variety of anticancer drugs. In this talk, the development of new acoustically-active bubbles that are capable of carrying and releasing therapeutic concentrations of hydrophobic and hydrophilic chemotherapeutic drugs will be introduced. Agents that are entirely composed of FDA-approved components are preferred as they may permit expedited translation into clinical use. Trade-offs among size, stability, drug-loading, and *in vivo* performance between bubble variants will be overviewed. Challenges and opportunities identified through recent efforts towards achieving cancer therapy potentiation in small animal cancer models will also be discussed.

9:55

2aBAb3. Acoustic droplet vaporization and gas embolotherapy. Joseph L. Bull (Biomedical Eng.eering, Tulane Univ., 6823 St. Charles Ave., Lindy Boggs Ctr., Ste. 201, New Orleans, LA 70118, jbull@tulane.edu)

We present an overview of our research group's work in acoustic droplet vaporization and gas embolotherapy, including bubble and droplet dynamics, bioeffects, targeting of droplets, localized drug delivery, and selective occlusion of blood flow to tumors. In these applications, transvascular liquid perfluorocarbon droplets are injected intravenously and, subsequently, vaporized with ultrasound to selectively form vascular micro- and nano-bubbles that are used for therapy. The resulting bubbles are approximately 125 times the volume of the droplets from which they originate. Embolization of tumors with this methodology involves droplets that are sufficiently large to produce bubbles that will lead to occlusion. Drug-loaded droplet may be used without occlusion if they are sufficiently small. We have used a combination of theoretical, computational, and experimental approaches to elucidate the behaviors and mechanisms involved in acoustic droplet vaporization. In a murine model of hepatocellular carcinoma, we have demonstrated that the combination of gas embolization and chemotherapy can result in complete tumor regression.

10:20–10:35 Break

10:35

2aBAb4. Bubble-based propulsion of engineered particles for drug delivery and immunotherapy. Wyatt Shields (Chemical and Biological Eng., Univ. of Colorado Boulder, Jennie Smoly Caruthers Biotechnology Bldg., 3415 Colorado Ave. #D218, Boulder, CO 80303-1904, charles.shields@colorado.edu), Andrew Goodwin, Taylor R. Ausec, and Jin G. Lee (Chemical and Biological Eng., Univ. of Colorado Boulder, Boulder, CO)

Biological barriers, such as mucus and cellular endothelia, obstruct the transport of drugs to diseased tissues. In this talk, I will highlight our recent work to overcome these barriers using engineering particles that undergo bubble-based propulsion in response to ultrasound. I will focus on two innovations. First, I will share our collaborative effort to synthesize hydrophobically modified mesoporous silica nanoparticles (MSNs) stabilized with phospholipids that entrap hydrophobic drugs. When insonated with high-intensity focused ultrasound, the MSNs undergo cavitation to rapidly propel through dense media and release drugs. Second, I will share our effort to fabricate microparticles with arbitrary three-dimensional shapes using two-photon lithography. These microparticles are capable of entrapping air bubbles of a defined size, leading to their efficient, frequency-dependent propulsion when stimulated by ultrasound. By modifying the surfaces of these particles, they can robustly attach to epithelial cell layers and release small molecule drugs for extended durations. By way of outlook, I will discuss how these technologies have potential to stimulate immune cells against tumor progression and to abate inflammation in auto-immune diseases.

Contributed Papers

11:00

2aBAb5. Biomimetic lung surfactant nanodrops for ultrasound imaging. Mark Borden (Biomedical Eng., Univ. of Colorado, 1111 Eng. Dr. UCB 427, ECME 245, Boulder, CO 80309, mark.borden@colorado.edu), Alec Thomas, Kang-Ho Song, Awaneesh Upadhyay (Biomedical Eng., Univ. of Colorado, Boulder, CO), Virginie Papadopoulou (Biomedical Eng., Univ. of North Carolina, Chapel Hill, NC), David Ramirez, Richard K. Benninger (Bioengineering, Univ. of Colorado, Denver, CO), Matthew R. Lowerison (Elec. and Comput. Eng., Univ. of Illinois, Urbana, IL), Pengfei Song (Elec. and Comput. Eng., Univ. of Illinois Urbana-Champaign, Urbana, IL), and Todd Murray (Biomedical Eng., Univ. of Colorado, Boulder, CO)

Nanodrops comprising a perfluorocarbon liquid core can be acoustically vaporized into echogenic microbubbles for ultrasound imaging. Packaging the microbubble in its condensed liquid state provides some advantages, including *in situ* activation of the acoustic signal, longer circulation persistence, and the advent of expanded diagnostic and therapeutic applications in pathologies, which exhibit compromised vasculature. One obstacle to clinical translation is the inability of the limited surfactant present on the nanodrop to encapsulate the greatly expanded microbubble interface, resulting in ephemeral microbubbles with limited utility. In this study, we examine a biomimetic approach to stabilizing an expanding gas surface by employing the lung surfactant replacement, Beractant. Lung surfactant contains a suite of lipids and surfactant proteins that provides efficient shuttling of material from bilayer folds to the monolayer surface. We hypothesized that Beractant would improve stability of acoustically vaporized microbubbles. To test this hypothesis, we characterized Beractant surface dilation mechanics and revealed a novel biophysical phenomenon of rapid interfacial melting, spreading, and re-solidification. We then harnessed this unique spreading capability to increase the stability and echogenicity of microbubbles produced after acoustic droplet vaporization for *in vivo* ultrasound imaging. Such biomimetic lung surfactant-stabilized nanodrops may be useful for applications in ultrasound imaging and therapy.

11:15

2aBAb6. Effectiveness of 3 MHz ultrasound in *ex vivo* scleral delivery of macromolecules of different sizes. Hanaa H. Almogbil (School of Eng. and Appl. Sci., George Washington Univ., Sci. & Eng. Hall, 800 22nd St. NW, Washington, DC 20052, Hanaa@gwu.edu), Fadi P. Nasrallah (Retina Consultants, Washington, DC), and Vesna Zderic (School of Eng. and Appl. Sci., George Washington Univ., Washington, DC)

Therapeutic ultrasound offers a novel approach for enhancing scleral delivery of macromolecules for treatment of various ocular diseases. Our previous *in vitro* diffusion cell studies showed that 3 MHz ultrasound could enhance scleral delivery of Avastin (MW: 149 kDa). We tested similar ultrasound parameters in an *ex vivo* whole eye rabbit model by using fluorescently labeled FITC-dextran of various sizes (40, 70, and 150 kDa) to mimic the sizes of the current macromolecules drugs for treatment of retinopathies. 3 MHz ultrasound was applied for 5 min on the sclera of the eye submersed in a macromolecule solution, and then the eyes were left in the solution for another 55 min while in the 34.6 °C water bath. Ultrasound was not turned on for the sham-treated group. Temperature modeling was performed to assess the safety of 5-min ultrasound application in the eye. The fluorescence intensity values of FITC-150 group were statistically different ($p < 0.05$) between the ultrasound-treated (1809.4, $n = 8$) and their sham-treated group (1185.8, $n = 8$). However, the FITC-70 and -50 groups lacked any appreciable difference between their ultrasound-treated (1313.6, $n = 8$) and (1646.3, $n = 6$), and their sham-treated groups (1431, $n = 8$) and (1121.8, $n = 6$), respectively. Thermal simulations demonstrated a temperature rise of 4.6 °C, while experimentally the maximum temperatures was 4.4 °C.

11:30

2aBAb7. Feasibility of cavitation nucleation in the crystalline lens. Alice Ganeau (LabTAU - INSERM U1032, Lyon, France), Maxime Lafond (LabTAU - INSERM U1032, 151, cours Albert Thomas, Lyon 69424, France, maxime.lafond@inserm.fr), Olfa Ben Moussa (Laboratoire de Biologie, Ingénierie et Imagerie de la Greffe de Cornée (BiiGC), Université Jean Monnet, Saint-Etienne, France), Charles Mion (LabTAU - INSERM U1032, Lyon, France), Sylvain Poinard, Frédéric Mascarelli (Laboratoire de Biologie, Ingénierie et Imagerie de la Greffe de Cornée (BiiGC), Université Jean Monnet, Saint-Etienne, France), Stefan Catheline (LabTAU - INSERM U1032, Lyon, France), Gilles Thuret, Philippe Gain (Laboratoire de Biologie, Ingénierie et Imagerie de la Greffe de Cornée (BiiGC), Université Jean Monnet, Saint-Etienne, France), and Cyril Lafon (LabTAU - INSERM U1032, Lyon, France)

Presbyopia is the age-related stiffening of the crystalline lens, reducing near vision. We propose ultrasonic cavitation to interact with the lens structure and restore its flexibility. Here, we present early results on the feasibility of the technique. First, the cavitation threshold was determined: the focus of a 4-elements transducer was placed in the lens nucleus in a porcine eye, and signals emitted in the focal area were acquired using an imaging array and reconstructed using frequency-domain passive cavitation imaging (PCI) with Richardson-Lucy deconvolution. Inharmonic emissions inside the lens were calculated, and we evidenced two distinct populations: low and high amount of signal inside the lens, when cavitation was absent or present, respectively. Peak negative pressure of 23 MPa for 20 μs pulses at 1.1 MHz provided a clear chance of nucleating cavitation with our system. Exposures were performed after 10 s of cavitation using a PRF of 250 Hz. Cavitation clouds were clearly visible in B-mode images for both 6-months old and 3-years old porcine eyes. The absence of cataract was confirmed on transparency images, and histology revealed no damages to the lens structure. These preliminary results are encouraging towards the application of cavitation in the crystalline lens.

11:45

2aBAb8. Optimization of cavitation-mediated mRNA delivery to cancer and non-cancer cells *in vitro* and *in vivo*. Alexander Martin (Inst. of Biomedical Eng., Univ. of Oxford, IBME, Old Rd. Campus Res. Bldg., Oxford OX3 7DQ, United Kingdom, alexander.martin@eng.ox.ac.uk), Cameron Smith, Robert Carlisle, Robin O. Cleveland (Inst. of Biomedical Eng., Univ. of Oxford, Oxford, United Kingdom), and Constantin Coussios (Inst. of Biomedical Eng., Univ. of Oxford, Headington, Oxford, Oxfordshire, United Kingdom)

The use of mRNA provides several compelling benefits. Cavitation-mediated delivery of mRNA has yet to be fully explored, especially in the context of achieving delivery of non-encapsulated, "free" mRNA. We present data that explore the relationship between cavitation and the transfer of, and protein expression from, free, fluorescently labelled mRNA. The protein expression exhibited in various cell lines *in vitro* was further tested *in vivo*, using sub-micron sized cavitation nuclei and the effects on transfection efficiency and expression time were examined. Acoustic exposure parameters were carefully selected, using a focused 0.5 MHz ultrasound transducer: pressure was varied from 0.3 to 2.1 MPa and 100–50 000 cycles at 5% duty cycle were used to achieve different cavitation regimes. A difference in expression time and transfection rate was observed between different cavitation nuclei and acoustic parameters. A luciferase reporter gene assay was used to characterise events *in vitro*. Expression due to cavitation-mediated delivery was further examined *in vivo* using the *In Vivo* imagining system and quantified using excised homogenised tumour tissue. These results provide us with a measurement window for maximum expression, exploration of how time points change across cell lines and lay the foundations for cavitation-mediated delivery and transfection of free mRNA.

TUESDAY MORNING, 24 MAY 2022 — DIRECTORS ROW H, 9:00 A.M. TO 11:30 A.M.

Session 2aMU

Musical Acoustics and Education in Acoustics: Strategies for Online and Hybrid Teaching of Musical Acoustics

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

Taffeta Elliott, Cochair
CLASS, New Mexico Inst. of Mining and Technology, 801 Leroy Pl, Class Dept, Socorro, NM 87801

Chair's Introduction—9:00

Invited Papers

9:05

2aMU1. Acoustics instruction in a pandemic world. Gordon P. Ramsey (Phys., Loyola Univ. Chicago, Phys. Dept., Loyola University Chicago, Chicago, IL 60660, gramsey@luc.edu)

COVID-19 has required a change in instruction at all levels. Fortunately, the technology exists to enable us to give quality education to our students. In acoustics, the wealth of existing online materials makes it practicable to motivate and enlighten our students to the beauty of our subject. Some of the key elements used in our courses can be presented effectively using auxiliary materials and web technology. Like many instructors, I provide PowerPoints with sound bites, text readings with discussion topics, and detailed supplements. Regular discussions are important to keep up motivation. "In class" labs can be done either live or online via Zoom or equivalent. Analysis and reports are done as homework. Exam security is an issue that can be minimized by alternate means. Assessments include short and frequent quizzes, essays and discussion writeups, lab reports, homework for practice, and a final project combining musical and physical aspects of the course. This project has been a more accurate assessment than a final exam. There is a wealth of web materials that are available on various sites. This talk will provide an overview of these elements as they apply to musical acoustics courses and strategies used to implement them.

9:25

2aMU2. Implementing a flipped classroom model to teach an introductory musical acoustics course. Andrew C. Morrison (Natural Sci., Joliet Junior College, 1215 Houbolt Dr., Joliet, IL 60431, amorriso@jjc.edu)

Because of the COVID-19 pandemic and our institution's response to the emergency, our musical acoustics course was delivered exclusively online for a full academic year, followed by a semester where only the lab component was in-person. A library of videos was developed to provide this course online. The videos contained the lecture content for the entire course. A flipped class model was adopted upon returning the course to a face-to-face modality. Students were assigned to watch the videos of class lecture material at home instead of expecting to hear that material presented in the classroom. Classtime was exclusively reserved for laboratory activities, answering student questions, and applying concepts covered in the videos to problems in preparation for class assessments. The benefits and challenges of this course delivery method will be addressed.

9:45

2aMU3. Collaborative online assignments for an introductory course in musical acoustics. Andrew A. Piacsek (Phys., Central Washington Univ., 400 E. University Way, Dept. of Phys., Ellensburg, WA 98926-7422, andy.piacsek@cwu.edu)

Hands-on activities are a valuable component of any introductory course in physical science, especially a course in musical acoustics that attracts students who are not majoring in a physical science. Tasks that require active engagement, both physical and cognitive, have been shown to improve retention of new concepts [*Am. J. Phys.* **80**, 478 (2012)]. Such tasks also provide an opportunity for students to improve collaboration skills and to learn from each other. The development of a well-vetted library of interactive online physics simulations, such as PhET (https://phet.colorado.edu), and recent improvements in online collaboration platforms have enabled collaborative active learning to occur in a structured way outside the classroom. This presentation will describe three collaborative tasks for a general education course in musical acoustics at Central Washington University that were adapted for online and hybrid versions of the course. Each task makes use of freely available software. Examples of how the collaboration can be structured using Zoom and Gather.town will be provided.

10:05–10:20 Break

10:20

2aMU4. Teaching a graduate course on the acoustics of musical instruments over Zoom from my house during COVID-19. Daniel A. Russell (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg, University Park, PA 16802, dar119@psu.edu)

During Spring 2021, I taught a graduate course, Acoustics of Musical Instruments, to a blended enrollment of resident and distance students. Course content focused on wind (woodwind and brass) and stringed instruments with heavy reference to published literature. This was my third time teaching this course, but the first time teaching it entirely over Zoom from my house with all students also participating only on Zoom. It was also the first time teaching without access to laboratory equipment that would have allowed for interactive classroom demonstrations and out-of-class student projects. Although several students were accomplished musicians and had access multiple musical instruments in their homes, very few students had any access to test equipment that would have allowed for meaningful measurements or experimental projects. To offset this lack of access, I purchased several musical instruments for demonstrations and simple experiments that I could perform from my house and shifted the emphasis for student deliverables to written literature review summaries of published papers. This presentation will describe simple hands-on activities students carried out, demonstrations I was able to perform from my house with limited data analysis capabilities, and the benefits of having graduate students do literature reviews.

10:40

2aMU5. In-group identity of a video narrator has no effect on instructional persuasiveness for undergraduates at a minority-serving technical university. Taffeta Elliott (CLASS, New Mexico Inst. of Mining and Technol., 801 Leroy Pl, Class Dept., Socorro, NM 87801, taffeta.elliott@nmt.edu) and Susmi Sharma (Neurosci., Univ. of Minnesota, Minneapolis, MN)

Identifying with upperclassmen may help undergraduates transition towards effective academic skills in college. We statistically compared two identity-related models of how video instruction could support students in adopting advantageous study habits in core math, science, and engineering programs. Each model relied on the same video animation and script about evidence-based study strategies for efficient learning (SSEL), presented in a first-semester required Composition class. SSEL in the in-group model were endorsed by a STEM-major upperclassman who introduced her experience improving her studies at the institution, before she narrated the video. The expert model version of the video was narrated by a trained voice actor who declared that cognitive psychologists espoused the same SSEL based on research. Volunteer participants were 136 undergraduates at a public technical 4-year Hispanic-Serving Institution (HSI). Video models were randomly assigned to each fall 2021 Composition class. Study habits were assessed on a 19-item pretest survey in the 4th week, just before students watched the video. They were assessed again using a parallel survey as a post-test in the antepenultimate week of the semester. Students reported significantly more SSEL in the posttest versus pretest. There was no significant effect of model, possibly due to student interest in science.

Contributed Papers

11:00

2aMU6. Experimenting with technologies in introductory musical acoustics classes. Jack A. Dostal (Dept. of Phys., Wake Forest Univ., PO Box 7507, Wake Forest Univ., Winston Salem, NC 27109, dostalja@wfu.edu)

I have relied on a variety of programs and platforms to teach my introductory Physics of Music course online. Now that my students are back in person, some of the online/hybrid course modifications still work well for in-person delivery. I will describe the changes that managed to "stick" as well as some that did not. These include ways I have used conferencing platforms, free/open source programs, course learning management systems (LMS), and streaming services in my class.

11:15

2aMU7. An observational protocol to measure learning in a problem-based learning vibrations course. Andrew Wright (Systems Eng., Univ. of Arkansas at Little Rock, 2801 South University Ave. Little Rock, AR 72204, abwright@ualr.edu) and Amanda Nolen (School of Education, Univ. of Arkansas at Little Rock, Little Rock, AR)

We have incorporated problem-based learning into several courses in an undergraduate engineering curriculum, including dynamics and vibrations. Students' prior knowledge (both formal and informal) were assessed through concept inventories to determine what prior knowledge supported or impeded the acquisition of new content in these courses. A point-and-click classroom observation tool was developed and refined over the span of two consecutive academic semesters to capture the students' interactions as they engaged in authentic learning activities in engineering. We identified patterns of discourse and interactions among the students and instructor, and, in vibrations, we identified a gap in the curriculum from the pre-requisite course, differential equations. The concept inventories and revised observational protocol that we have developed will enable instructors to assess the quality of the prior knowledge that students bring into a course and to analyze the manner in which students and instructor engage around the course content. We are developing bridge curriculum to improve transference of the theoretical knowledge in differential equations and the applied skills in vibration. This project spans across the beginning of the COVID-19 pandemic and includes observations on the interventions (successful and unsuccessful) where we attempted to continue student learning through a disrupted environment.

TUESDAY MORNING, 24 MAY 2022 PLAZA BALLROOM F, 9:00 A.M. TO 11:50 A.M.

Session 2aNS

Noise, Rockets, Wind Turbines, and Other Topics on Noise

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

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

Contributed Papers

9:00

2aNS1. Spectral analysis of Firefly Alpha launch noise. Matthew G. Yancey, Griffin Houston (Phys. and Astronomy, Brigham Young Univ., Provo, UT), Grant W. Hart (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), Logan T. Mathews, Michael S. Bassett (Phys. and Astronomy, Brigham Young Univ., Provo, UT), J T. Durrant (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, taggart.durrant@gmail.com), and Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

On 2 September 2021, the first Firefly Alpha rocket launched from Vandenberg Space Force Base in California, USA. Acoustic data were collected at two locations 30 m and 60 m away from the launch pad, at another eight locations located on an arc with a radius of about 300 m, as well as an additional location roughly 8 km away. Around 14 s into the launch, an anomaly occurred, resulting in the premature shutdown of one of the four first-stage engines. By analyzing the overall levels and spectral characteristics of the rocket throughout the different phases of its flight, the effects of the engine shutdown on overall level and spectral peak frequency are discussed. Based on estimated engine parameters, the measured effects are compared to models for clustered nozzles.

9:15

2aNS2. Outdoor acoustic measurements of a rocket engine hot fire test. Peter Kerrian (ATA Eng., Inc., 13290 Evening Creek Dr. S, San Diego, CA 92128, peter.kerrian@ata-e.com), John McShane, Parthiv Shah, Michael Yang (ATA Eng., Inc., San Diego, CA), Dan Larson, and Bill Murray (Ursa Major Technologies, Berthoud, CO)

The primary focus of this paper is to present the results from outdoor acoustic measurements from a rocket engine hot fire test. The two objectives of the test were to quantify the overall sound pressure level at large

distances from engine to quantify the impact of the events on the surrounding community, and to better understand the source distribution in the engine plume. Microphone arrays were positioned around the property to be able to capture the sound propagation, estimate source levels, and conduct phased array-based plume source localization.

9:30

2aNS3. Acoustical measurement and analysis of an Atlas V launch without solid rocket boosters. Carson F. Cunningham (Dept. of Phys. and Astronomy, Brigham Young Univ., Brigham Young Univ., Provo, UT 84602, carsonfcunningham@gmail.com), Mark C. Anderson, Levi T. Moats, Kent L. Gee, Grant W. Hart (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), Lucas K. Hall (California State Univ., Bakersfield, CA), and Steven C. Campbell (Ball Aerosp., Beavercreek, OH)

In September 2021, an Atlas V rocket without solid rocket boosters was launched from Vandenberg Space Force Base, California, carrying the NASA/USGS Landsat 9 satellite. In this launch configuration, the plumes from the RD-180 engine's two nozzles are unobstructed, providing the opportunity to analyze the sound generated by a liquid-fuel rocket engine with an azimuthally asymmetric nozzle geometry. Acoustical data were collected at various locations surrounding the launch pad, ranging from a few hundred meters to several kilometers. This paper discusses an overview of the measurement logistics, the types of analyses that may be performed using these data, and an initial analysis of the azimuthal variability of the overall sound pressure level and spectra.

9:45

2aNS4. Comparison of rocket launch data to empirical models for overall level and peak directivity angle. Michael S. Bassett (Dept. of Phys. and Astronomy, Brigham Young Univ., 562 N 200 E Apt 15, Provo, UT 84606, MichaelScottBassett@physics.byu.edu), Samuel A. Olausson, Griffin Houston, Logan T. Mathews, Mark C. Anderson, J T. Durrant, Grant W. Hart, and Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

With increasing frequency in global rocket launches, it is important to assess the effects of liftoff noise. Understanding of the fundamental physics of the sound radiation from rocket exhaust plumes remains limited. Some empirical predictive models, derived from plume parameters, cover aspects of the sound radiation, such as convective Mach number, peak directivity angle, and overall sound pressure level in the direction of maximum emission. These are evaluated using high-fidelity noise data from a number of different launches from Vandenberg Space Force Base. Initial results of the model comparisons will be presented.

10:00–10:20 Break

10:20

2aNS5. Field validation of octave band sound modeling for wind turbines. Dana Lodico (RSG, 1801 Broadway, Denver, CO 80202, dana.lodico@rsginc.com), Ken Kaliski (RSG, White River Junction, VT), Emma Butterfield, Hugo Rost (RSG, Denver, CO), and Shawn Fitzgerald (RSG, White River Junction, VT)

The Sugar Creek Wind Project is a 57-turbine wind farm in Logan County, Illinois, with a capacity of up to 202 megawatts. Preconstruction sound modeling of the final Project layout was conducted in July 2019. Following construction of the Project, postconstruction attended sound measurements were made at 38 sites in the vicinity of Project turbines in September, October, and November 2021. Measured postconstruction sound levels were compared to the octave band sound level limits applicable under the Project's Conditional Use Permit. A comparison of the modeled and background-adjusted measured octave band sound levels for the Project indicate that with the proper modeling parameters, octave band sound levels can be conservatively predicted.

10:35

2aNS6. Attended sound monitoring of wind turbines. Emma I. Butterfield (Resource Systems Group, Inc., 1801 N Broadway, Ste. 505, Denver, CO 80202, emma.butterfield@rsginc.com), Dana Lodico (Resource Systems Group, Inc., Denver, CO), Ken Kaliski, Shawn Fitzgerald (Resource Systems Group, Inc., White River Junction, VT), and Hugo Rost (Resource Systems Group, Inc., Denver, CO)

Long-term unattended sound monitoring of wind turbines allows the selection of optimal data from a long period of time in which many meteorological conditions occur. However, in some cases, attended monitoring is required, either as a specific requirement of the permitting or because many measurement sites are required, making long-term monitoring impractical. Short-term attended monitoring of wind turbines introduces several complexities to ensure that the data are valid and representative of the target conditions. To illustrate this, we describe the challenges of attended octave band sound measurements made at 38 sites in the vicinity of wind turbines in Illinois in September, October, and November 2021. The attending monitoring was completed at night to reduce the background sound levels and involved the cooperation of residences, turbine operators, and the weather. Successful navigation of these challenges led to completion of the sound monitoring at all locations and acceptance of the study by the County.

10:50

2aNS7. Directional processing in assessment of wind turbine noise. Alexander Sutin (STAR Ctr., Stevens Inst. of Technol., 1 Castle Point Terrace, Hoboken, NJ 07030, asutin@stevens.edu), Hady Salloum, Alexander Sedunov, and Nikolay Sedunov (STAR Ctr., Stevens Inst. of Technol., Hoboken, NJ)

Assessments of wind turbine generator (WTG) noise are required to comply with the US Environmental Agency and local governments and avoid legal action that may result of non-compliant operation. Current methods for WTG noise measurements require the comparison of long-term sound data recorded before and after a WTG installation. These measurements must be conducted during several months for various wind speeds and environmental conditions. Such measurements are complicated due to the presence of multiple sources other than noise not from WTG, including intermittent and fluctuating sources. The application of directional processing can help localize and identify the sources responsible for increased sound levels at any given moment and to estimate the contribution of the other than WTG sources to the whole sound level. Stevens has developed various acoustic systems with several sensors that provide directional processing of sounds that were previously applied for detection, tracking and classification of small boats, divers, small aircraft, and drones. We propose modifications of such systems for directional sensing of WTG noise at long distances in the frequency band relevant for WTG noise measurement.

11:05

2aNS8. Acoustic optimization of a contra-rotating propeller rig. Fabio Casagrande Hirono (School of Sci., Eng. and Environment, Univ. of Salford, The Crescent, Salford M5 4WT, United Kingdom, F.CasagrandeHirono@salford.ac.uk), Antonio Torija Martinez, and Andrew Elliott (School of Sci., Eng. and Environment, Univ. of Salford, Manchester, United Kingdom)

With the advance of unmanned aerial vehicles as a viable alternative for urban transportation, it is imperative that these vehicles have a low impact on community noise. Contra-rotating propeller technologies offer interesting advantages over traditional single-propeller equivalents, such as improved aerodynamic performance and redundancy in case of failure, but can be exceedingly noisy when not carefully designed. This work presents a parametric study for the acoustic optimization of a custom-made contra-rotating propeller rig. The rig consists of two commercially available motors mounted on a rotating stand built in-house and mounted inside an anechoic chamber. A far-field microphone arc is used to collect acoustic pressure data, and a load cell is used to measure total thrust. We investigate the effect

of variations in the number of blades (between 2 and 6 on both rotors) and axial spacing (between 0.1 and 1 rotor diameter) on the radiated noise, while adjusting the rpm to maintain constant thrust. Overall sound pressure levels, far-field spectra, and directivities (overall SPL, tones, and broadband) are assessed to determine trade-offs and optimal choices in designing a contra-rotating propeller rig.

11:20

2aNS9. Use of an acoustic camera together with modeling of a unique crypto-mining facility. Iliana Albion-Poles (Wave Eng., 1100 W Littleton Blvd., #420, Littleton, CO 80120, ilianaap@waveengineering.us)

Crypto-currency mining facilities have become more and more common over the last few years, but there is no standard for construction or noise mitigation. Many noise issues are encountered after the facilities are complete and operating. Mining facilities can contain thousands of "miners" which are quite loud. Often, the manufacturers cannot supply reliable sound data. The miners have extreme power and ventilation requirements, which introduce additional noise sources and paths. The use of modeling software is often a great tool to predict the noise of a future project, rank order multiple noise sources, or to determine what mitigation strategies will be most effective. Using an acoustic camera and standard sound level measurements together with a noise model of an existing facility can better pinpoint the dominant noise sources and facilitate the type and placement of noise mitigation measures. This presentation will show how the multiple technologies were used together to identify the problem noise sources and paths and to evaluate mitigation measures.

11:35

2aNS10. Chiller compressor sound emission analysis. Hezekiah George (Bradley Univ., P. O. Box 502245, St. Thomas 00805, VI, hgeorge@mail.bradley.edu), Jingdou Wang, and Becky Bryant (Trane Technologies, Davidson, NC)

Building occupants rely on heating, ventilation, and air conditioning (HVAC) units for comfort. Chillers are HVAC units that serve the purpose of cooling a building. Because chillers are often placed on top of or near the building, the noise from the chiller can adversely affect the occupants. A crucial component of all chillers, the compressor, is responsible for the majority of the noise produced by the chiller. In addition to designing for cooling efficiency, engineers must also consider how quiet and acoustically efficient a chiller is. In this project, we aim to develop an enclosure to house the compressor, therefore, reducing the magnitude of noise that is emitted. We begin by studying an empty cavity model in an acoustical software and observing its modes, resonant frequencies, and corresponding pressure distributions. Next, we placed a sound source within the cavity and then analyzed the acoustics frequency response of the cavity without and with a sound-absorbent material. We then proceeded to analyze a FEM model of the compressor and its dynamic frequency response due to the bearing force. After further analysis of the compressor sound emission, simulations will be made for the compressor with the designed enclosure and sound-absorbent material.

Session 2aPAa

Physical Acoustics, Computational Acoustics, Signal Processing in Acoustics, and Noise: Progress in Sonic Boom Modeling, Processing, and Analysis on Community Response

Joel B. Lonzaga, Cochair
Structural Acoustics Branch, National Aeronautics and Space Administration, 2 N. Dryden St. (MS 463), Hampton, VA 23681

Aaron B. Vaughn, Cochair
Structural Acoustics Branch, NASA Langley Research Center, 1 NASA Drive, Hampton, VA 23666

Philippe Blanc-Benon, Cochair
CNRS, 36 Avenue Guy de Collongue, Ecully, 69134, France

Chair's Introduction—7:55

Invited Papers

8:00

2aPAa1. Implementation of a sonic boom propagation algorithm with the augmented Burgers equation. Jacob J. Jäschke (Inst. of Air Transportation Systems, Hamburg Univ. of Technol., Blohmstraße 20, Hamburg 21079, Germany, jacob.jaeschke@tuhh.de) and Bernd Liebhardt (Inst. of Air Transportation Systems, German Aerosp. Ctr. (DLR), Hamburg, Germany)

For the purpose of sonic boom carpet determination for an optimized routing of supersonic aircraft around landmasses, a sonic boom propagation code was implemented at the Institute of Air Transportation Systems. The implementation is based on the well-known augmented Burgers equation for stratified atmospheres and extends the sole geometric ray tracing that has been used at the institute to determine the lateral cutoff distance of sonic booms so far. For the quantification of sonic boom signatures on ground, algorithms for the computation of noise metrices, namely, the perceived level of noise Stevens Mk VII (PL) and the sound exposure level (SEL) with several frequency weightings are also implemented. This work describes the implementation of the sonic boom propagation algorithm and the methods used for loudness calculation briefly. For the evaluation of the implemented code, pre-defined cases from the Third Sonic Boom Prediction Workshop are simulated, and the computed results are compared to the published results of that workshop. The similarities and differences are discussed in detail, such that the general implementation of the propagation code can be verified, whereas potential inaccuracies, especially for the off-track determination, can be detected and quantified.

8:20

2aPAa2. Three-dimensional wavefront modeling of secondary sonic booms. Joseph A. Salamone (Eng., Boom Supersonic, 12876 East Adam Circle, Centennial, CO 80112, joe.salamone@boom.aero)

There is an abundance of previous work in the literature regarding the Mach cone of a supersonic vehicle and its associated ground intercepts identifying where and when the primary sonic boom reaches the ground. Likewise, there is excellent past research that models where the secondary sonic boom reaches the ground due to the reflection and refraction of the sonic boom emanating from above and below a supersonic vehicle. This work will provide a bridge between the two aforementioned aspects of past research. The Mach cone for both above the aircraft and below the aircraft after it reaches the ground will be extended to model the three-dimensional wavefront evolution of the secondary sonic boom as it propagates to the ground. The secondary sonic boom wavefronts modeled in this work will only consist of rays that refract in the vicinity of the stratosphere. The present work will show several cases of the secondary sonic boom wavefront propagation in three dimensions for a variety of aircraft headings and geographic locations around the world with representative upper air temperature and wind conditions.

8:40

2aPAa3. Coastal buffer distances and secondary sonic booms. Victor W. Sparrow (Penn State, 201 Appl. Sci. Bldg., University Park, PA 16802, vws1@psu.edu)

This presentation describes how secondary sonic booms have influenced our understanding of appropriate coastal buffer distances for supersonic aircraft. The International Civil Aviation Organization issued its Circular 126 in 1975 after the completion of test flights for the supersonic aircraft Concorde but before it began routine operations. This guidance was to prepare everyone for Concorde's introduction. Particularly Chapter 5 recommended that a supersonic aircraft over the ocean should slow down to subsonic speeds at least 45

nautical miles before reaching the coastline, providing a coastal buffer distance to keep the sonic boom off of land. However, when Concorde began regular transatlantic service, the phenomenon of secondary sonic booms was better appreciated. Air France and British Airways changed their operations to provide much larger coastal buffer distances on the order of 150 nautical miles. Increasing the coastal buffer distances was successful in enabling routine Concorde operations. Implications for the introduction of future civilian supersonic aircraft are described. [Work funded by the FAA through ASCENT Project 57. Any opinions, findings, conclusions or recommendations expressed in this material are those of the author and do not necessarily reflect the views of the FAA.]

9:00

2aPAa4. Reflection of sonic boom over realistic urban areas. Didier Dragna, Ariane Emmanuelli, Sebastien Ollivier (LMFA, Ecole Centrale de Lyon, Ecully, France), and Philippe Blanc-Benon (LMFA, Ecole Centrale de Lyon, 36 Ave. Guy de Collongue, Ecully 69134, France, philippe.blanc-benon@ec-lyon.fr)

This study aims at analyzing sonic boom reflection over realistic urban areas. For this purpose, numerical simulations are performed. The 2D Euler equations are solved using high-order finite-difference methods, following Emmanuelli *et al.* [*J. Acoust. Soc. Am.* 149, 2437–2450 (2021)]. The local climate zone classification of urban morphologies [Steward and Oke, Bull. Am. Meteorol. Soc. 93, 1879–1900 (2012)] is used to generate ten typical geometries. Two booms, an N-wave and a low-boom wave, are considered. The pressure waveforms along the profiles are analyzed first. In addition to the geometric arrivals, low-frequency oscillations, associated with resonance modes of the street canyons, are noted. They induce an important increase in the duration of the signals. Their frequency and amplitude depend on the street width and on the boom wave. The noise levels are then investigated. Depending on the urban geometry, two behaviors are highlighted. For a sparse arrangement of buildings, the evolution of the levels is similar than for isolated buildings, with an amplification on the front facade and a reduction on the rear facade. For a dense arrangement, it is more complex due to the multiple reflections on the buildings. Finally, the levels follow a comparable evolution for both boom waves.

9:20

2aPAa5. A spatial averaging of sonic boom waveforms. Joel B. Lonzaga (Structural Acoust. Branch, National Aeronautics and Space Administration, 2 N. Dryden St. (MS 463), Hampton, VA 23681, joel.b.lonzaga@nasa.gov)

Sonic booms generated by conventional supersonic aircraft are influenced by turbulence in the atmospheric boundary layer through which they propagate. The turbulence effects lead to random variability of the sonic boom waveforms measured on the ground, complicating the prediction of such waveforms. While some of the waveforms have been described as N-waves with spikes immediately after the shocks, others measured a few hundreds of feet away tend to be rounded. The current paper presents an analysis method that uses a spatial averaging of an ensemble of measured waveforms to obtain a mean waveform and higher-order statistical moments. The statistics obtained are anticipated to be useful in characterizing the waveforms generated by any supersonic vehicle. They will also be useful for validating theoretical and semi-empirical models used for predicting turbulence effects on sonic booms. This paper discusses the advantages and disadvantages of spatial averaging in the time domain and the frequency domain. Preliminary results indicate that the mean waveforms have rise times larger than those obtained using a sonic boom propagation model that does not account for the turbulence effects, suggesting that turbulence effects may not be ignored.

Contributed Paper

9:40

2aPAa6. Analysis of sonic booms from Falcon 9 booster landings. J T. Durrant (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, taggart.durrant@gmail.com), Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), Mark C. Anderson, Logan T. Mathews (Phys. and Astronomy, Brigham Young Univ., Provo, UT), and Grant W. Hart (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

One drawback to the ability to land boosters of orbital launch vehicles, such as the Falcon 9, is the associated reentry sonic boom. Because of the Falcon 9 booster shape and landing orientation, the observed waveform at the ground contains three shocks (a triple boom), rather than the two associated with a traditional N-wave. To assess the impact of these sonic booms, acoustic data from a Falcon-9 booster landing at Vandenberg Space Force Base are analyzed. The data were collected in Lompoc, CA, at locations 7-14 km away from the landing site. Waveform and spectral characteristics are examined and various metrics, including A-weighted Sound Exposure Levels (ASEL), are calculated. These metrics from the reentry sonic booms are compared with metrics calculated from the launch noise.

9:55–10:10 Break

10:10

2aPAa7. Investigation of measured sonic boom variability and ambient noise coherence across a 400-ft linear array. Mark C. Anderson (Dept. of Phys. and Astronomy, Brigham Young Univ., D-70 ASB, Provo, UT 84602, anderson.mark.az@gmail.com), Kent L. Gee, J T. Durrant, Logan T. Mathews (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), Alexandra Loubeau, and William Doebler (NASA Langley Res. Ctr., Hampton, VA)

In preparation for the NASA X-59 flight test campaign, it is relevant to understand the variability of measured sonic boom metrics over short distances. One of the measurement stations at the 2019 NASA Carpet Determination in Entirety Measurements (CarpetDIEM) campaign consisted of a 400-ft linear array of seven microphones beneath and perpendicular to the flight track. The purpose of this array was to assess variability in CarpetDIEM boom characteristics over the array aperture. Analysis shows that all candidate metrics can vary by more than 6 dB across the array and that this variability appears to be normally distributed about a mean value for the array. Confidence intervals for several boom metrics are discussed. The ambient noise coherence across the array is also discussed in relation to the sonic boom metrics. [Work supported by NASA Langley Research Center through the National Institute of Aerospace.]

10:30

2aPAa8. Regression analysis of meteorological, aircraft trajectory, and ambient effects on sonic boom metric variability. J T. Durrant (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, taggart.durrant@gmail.com), Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), Mark C. Anderson, Mylan R. Cook (Phys. and Astronomy, Brigham Young Univ., Provo, UT), Alexandra Loubeau, and William Doebler (NASA Langley Res. Ctr., Hampton, VA)

NASA's Low-Boom Flight Demonstration mission is a major step toward commercial, overland supersonic flight. Certifying low-boom aircraft will require accurate measurements, including understanding uncertainty due to variations in meteorology, aircraft trajectory, and measurement environment. This work builds on preliminary work done with Least Absolute Shrinkage and Selection Operator (LASSO) regression [Durrant *et al.*, J. Acoust. Soc. Am. 150, A259 (2021)] to analyze the variability present in NASA's Quiet Supersonic Flights 2018 (QSF18) dataset. With the variability quantified and several factors having been identified as potential contributors through LASSO regression, least-squares regression is used on sets of these potential contributors to determine coefficient weights for each factor, including meteorological, aircraft trajectory, and ambient noise. Sonic boom metrics are then predicted using the results and the accuracy is compared to PCBoom predictions. Because the ambient noise is significant, the same regression techniques are used to examine which variables from a geographic information systems (GIS) dataset are correlated with the ambient noise. These techniques are likely to be useful for noise monitor placement planning and data interpretation during community testing of low-boom aircraft. [Work supported by NASA Langley Research Center through National Institute of Aerospace.]

10:50

2aPAa9. Review of noise metric sensitivities for analysis of quiet supersonic overflight. Alexandra Loubeau (NASA Langley Res. Ctr., MS 463, Hampton, VA 23681, a.loubeau@nasa.gov)

Six different noise metrics (PL, ASEL, BSEL, DSEL, ESEL, and ISBAP) are currently used to quantify sonic boom levels from overflight of supersonic aircraft. A previous meta-analysis using laboratory subjective data identified these metrics, each of which correlated well with human perception of low-boom sounds both indoors and outdoors. Because the analysis did not identify a single metric that was significantly superior, no internationally agreed-upon metric has been chosen for the quiet supersonic aircraft noise certification procedures currently under development. Other analyses of metric sensitivities using existing empirical and simulation datasets, however, have shown significant differences between metrics. Variability of metrics due to macro-atmospheric effects and atmospheric turbulence perturbations have been explored, as well as variability in measurements due to microphone setup configurations, array layouts, and ambient noise effects. This paper reviews these prior studies, summarizes the current understanding of metric sensitivities, and discusses how they may impact future downselection of metrics for certification procedures.

11:10–11:40
Panel Discussion

Session 2aPAb

Physical Acoustics, Biomedical Acoustics, and Structural Acoustics and Vibration: Vortex Beams and Radiation Torques I

Likun Zhang, Cochair
National Center for Physical Acoustics and Department of Physics and Astronomy, University of Mississippi, University, MS 38655

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

Chair's Introduction—7:55

Invited Papers

8:00

2aPAb1. Designing an array for acoustic manipulation of kidney stones. Mohamed A. Ghanem (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington, 1013 ne boat St., Seattle, WA 98105, mghanem@uw.edu), Adam D. Maxwell (Dept. of Urology, Unviersity of Washington, Seattle, WA), Oleg A. Sapozhnikov, and Michael R. Bailey (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington, Seattle, WA)

Remote acoustic manipulation of objects in three dimensions has been achieved through acoustic radiation forces imparted by designed acoustic beam traps. This technology can be used as a noninvasive medical tool to control foreign objects in the body. Success of this technology requires the ability to transmit complex beam shapes through the tissue while maintaining stable and robust acoustic traps. Vortex beams and other acoustic beam traps were synthesized using a 1.5 MHz, focused, multi-element array to trap and manipulate glass spheres mimicking stones. Measurements of the radiation forces agreed with calculations to within $<10\%$. The strength of the trapping stability and steering range of the acoustic traps were achieved by controlling the pulsing directionality to eliminate rotational instability, geometrical shape of the beam relative to target size, and acoustic power of the beam. Spheres were noninvasively moved along predefined paths in the urinary bladders of pigs without morphological injury. To further improve the technology of remote manipulating of kidney stones, simulations and experiments were used to define a design and parameters of a 256-element array to move 2–5 mm stones distances up to 3 cm in a kidney. [Work supported by NIH P01-DK043881 and APL SEED Fellowship.]

8:25

2aPAb2. Low frequency (<1 MHz) transducers for intravenous sonothrombolysis. Xiaoning JIang (NC State, 911 Oval Dr., Raleigh, NC 27695, xjiang5@ncsu.edu), Bohua Zhang, Huaiyu Wu (NC State, Raleigh, NC), and Chengzhi Shi (GWW School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

Venous embolism is a great challenge in the treatment of a wide range of diseases, including cardiovascular diseases and cancers. Ultrasound-mediated thrombolysis (sonothrombolysis) presents a promising treatment of venous embolism. In this paper, miniature ultrasound transducers with frequencies ranging from 250 kHz to 800 kHz were developed for microbubble/nanodroplet-mediated intravenous sonothrombolysis tests. Relative low frequency (e.g., <1 MHz) is preferred because of the findings that cavitation plays an important role in sonothrombolysis, and microbubble-associated cavitation events can be enhanced at low frequencies. Small aperture (e.g. <2 mm $\times$ 2 mm) low-frequency transducers are usually known with high electrical impedance (e.g., $>1000\ \Omega$), which is not preferred for efficient electromechanical transduction. To address this challenge, multilayer acoustic stacks were designed, fabricated, and characterized. The measured electrical impedance of these stacks is mostly closer to 50 Ω, and the peak negative pressure is higher than 1 MPa. These small aperture transducers were then used in *in vitro*- thrombolysis tests with unretracted and retracted blood clots, showing promising lysis rates. Other techniques such as laser ultrasound transducers, vortex wave generation, and magnetic microbubbles were also investigated for thrombolysis tests.

8:50

2aPAb3. Amplitude and time-dependence of ultrasonic radiation force on modulation frequency computed from specular reflection contributions. Philip L. Marston (Phys. & Astronomy, Washington State Univ., Washington State University, Phys. & Astronomy Dept., Pullman, WA 99164-2814, marston@wsu.edu), Timothy D. Daniel, Auberry R. Fortuner (Phys. & Astronomy, Washington State Univ., Bremerton, WA), Ivars P. Kirsteins (NUWCDIVNPT, Newport, RI), and Ahmad T. Abawi (HLS Res., San Diego, CA)

Ever since the late 1970s, there has been interest in the modulated radiation pressure of double sideband suppressed carrier (DSSC) ultrasound, because the associated radiation pressure oscillates at a single low frequency. Recent research considers situations, where the radius of curvature of the illuminated objects greatly exceeds the wavelength at the implicit carrier frequency of the ultrasonic illumination [Marston *et al.*, *J. Acoust. Soc. Am.* 149, 3042–3051 (2021); 150, 25–28 (2021)]. Furthermore, the objects of interest, cylinders, and spheres in water are taken to be sufficiently highly reflecting that the radiation force can be shown to be dominated by contributions associated with specular reflection. In such cases, it is helpful to geometrically analyze the amplitude and time dependence of the oscillatory radiation force for perfectly reflecting objects, where the relevant integrals of the stress projections can be carried out analytically. The results are expressed using Bessel and Struve functions. They explain how the amplitude and phase of the force depends on the DSSC modulation frequency in agreement with partial-wave-series based calculations. The approach is relevant in other situations. [Work supported by the U. S. Office of Naval Research.]

Contributed Papers

9:15

2aPAb4. Modeling the effects of modulated radiation pressure using a new acoustic lens. Ahmad T. Abawi (HLS Res., San Diego, CA, abawi@hlsresearch.com), Philip L. Marston (Phys. & Astronomy, Washington State Univ., Pullman, WA), Ivars P. Kirsteins (NUWCDIVNPT, Newport, RI), Auberry R. Fortuner, Sterling M. Smith, and Timothy D. Daniel (Phys. & Astronomy, Washington State Univ., Pullman, WA)

Modulated radiation pressure (MRP) is a method of utilizing the acoustic radiation force by using a double-sided suppressed carrier modulated (DSB-SCM) signal. This produces a modulated signal that oscillates at twice the modulation frequency. This process facilitates target detection and classification by using a high-carrier frequency to probe the target and shake it with the low-modulation frequency, which can be designed to be commensurate with the vibrational modes of the target. A frequency sweep can estimate the target resonant frequencies, while a physical scan along its length can estimate its mode shapes. Recently, a series of experiments were conducted at Washington State University with a newly designed and fabricated acoustic lens with a one-meter focal length. These experiments demonstrated that the first few bending modes of scaled targets like a solid aluminum cylinder and an open-ended aluminum cylindrical shell could be excited by MRP at a standoff distance of one meter and target mode shapes were measured using a physical scan. In this presentation, we show modeling results that confirm experimental findings: (1) estimation of target resonant frequencies using a frequency sweep and (2) estimation of target mode shapes using a physical scan.

9:30

2aPAb5. Quantifying the efficiency of the acoustic radiation force from two-wave beating for structural resonance excitation of a thin spherical shell. Steven R. Craig (GWW School of Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., NW, Atlanta 30318, Georgia, scraig32@gatech.edu), Karim G. Sabra, and Chengzhi Shi (GWW School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

The structural echoes obtained from low-frequency ($\sim$kHz) excitation of resonant underwater targets can serve as a basis for automated target recognition. Acoustic radiation force (ARF) can provide a means to excite those structural echoes with compact transducing mechanisms as it relies on high-frequency transducers to generate locally low-frequency excitation via the classical beating effect. However, a key question is quantifying the efficiency of ARF excitation mechanisms to excite those structural echoes when compared to using direct low-frequency pulses. Here, we develop a time-domain finite element framework that compares the ARF and direct low-frequency excitation methods to excite classical resonances scenarios such as thickness resonances of a thin plate or the so-called "mid-frequency enhancement echo" for elastic spherical shells. These canonical target objects provide a straightforward procedure to investigate the ARF excitation efficiency and establishes the numerical framework to investigate more complex targets with a wider range of environmental parameters.

9:45–10:00 Break

Invited Paper

10:00

2aPAb6. Born approximation of acoustic radiation force and torque. Thomas S. Jerome (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX) and Mark F. Hamilton (Appl. Res. Labs., The Univ. of Texas at Austin, Walker Dept. of Mech. Eng., The University of Texas at Austin, Austin, TX 78712-1591, hamilton@mail.utexas.edu)

The Born approximation provides a simple method for calculating acoustic radiation force and torque on compressible objects of arbitrary shape, which may be inhomogeneous. The specific acoustic impedance throughout the object must not differ significantly from that of the background fluid. Also, the incident field must not be too similar to a progressive plane wave, because the approximation is associated with radiation forces due to time-averaged energy density gradients in the incident field. Comparisons with a complete theory for radiation force and torque exerted by standing plane waves on compressible spheroids, based on expansions of the incident and scattered fields in spheroidal wave functions, indicate that the Born approximation is accurate for impedance contrasts up to at least 30% and for spheroids with sizes on the order of a wavelength. This presentation will review the derivation, validation, and applications of the Born approximation. Closed-form solutions for spheres and finite cylinders are presented and for selected distributions of material inhomogeneity. Applications include acoustofluidic separation of biological targets, and acoustic forces on nonspherical objects near interfaces. The approximation also serves as a useful benchmark for other methods of calculation. [T.S.J. was supported by an ARL:UT McKinney Fellowship in Acoustics.]

Contributed Papers

10:25

2aPAb7. Born approximation of acoustic radiation force used for acoustofluidic separation. Chirag A. Gokani (Appl. Res. Labs., The Univ. of Texas at Austin, Appl. Res. Labs., The University of Texas at Austin, Austin, TX 78713-8029, chiragokani@gmail.com), Thomas S. Jerome, Michael R. Haberman, and Mark F. Hamilton (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Acoustofluidic separation often involves biological targets with specific acoustic impedance similar to that of the host fluid, and with dimensions on the order of the acoustic wavelength. This parameter range, combined with the use of standing waves to separate the targets, lends itself to use of the Born approximation for calculating the acoustic radiation force. Considered here is the configuration analyzed by Peng *et al*. [J. Mech. Phys. Solids **145**, 104134 (2020)], in which two intersecting plane waves radiated into the fluid by a standing surface acoustic wave exert a force on a eukaryotic cell modeled as a multilayered sphere. The angle of intersection is determined by the velocity of the surface wave and the sound speed in the fluid. The acoustic field in this case is a standing wave parallel to the substrate and a traveling wave perpendicular to the substrate. For all parameter values considered by Peng *et al*., including spheres several wavelengths in diameter, the Born approximation of the acoustic radiation force parallel to the substrate is in good agreement with a full theory based on spherical wave expansions of the incident and scattered fields. [C.A.G. and T.S.J. were supported by ARL:UT McKinney Fellowships in Acoustics.]

10:40

2aPAb8. Acoustic radiation force and torque on a spheroid near an interface. Blake E. Simon (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, blakesimon8@utexas.edu) and Mark F. Hamilton (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Acoustic radiation force and torque on a spheroid of arbitrary size near a rigid or pressure release interface are calculated using expansions of the pressure field in terms of both spherical and spheroidal wave functions. The spheroidal expansion coefficients are obtained by satisfying the boundary conditions at the interface and at the surface of the spheroid. Conditions at the interface are satisfied using a virtual (image) spheroid, while satisfying conditions at the spheroid surface require the use of addition theorems for spheroidal wave functions. The radiation force and torque are expressed as a summation of terms involving products of the coefficients in spherical wave expansions of the incident and scattered fields [Ilinskii *et al*., J. Acoust. Soc. Am. **144**, 568–576 (2018)]. Far-field asymptotes are used to relate the spheroidal and spherical harmonic expansion coefficients analytically. Results from the present analytical model are compared with those from a finite-element model. The effect of the interface on the radiation force and torque relative to that on a spheroid in a free field is discussed. [B.E.S. is supported by the Applied Research Laboratories Chester M. McKinney Graduate Fellowship in Acoustics.]

Invited Paper

10:55

2aPAb9. Acoustic vortex beam reflection, refraction, and orbital Hall effect. Likun Zhang (National Ctr. for Physical Acoust. and Dept. of Phys. and Astronomy, Univ. of Mississippi, 145 Hill Dr., University, MS 38655, zhang@olemiss.edu), Xu-dong Fan, Zheguang Zou, and Robert L. Lirette (National Ctr. for Physical Acoust. and Dept. of Phys. and Astronomy, Univ. of Mississippi, Oxford, MS)

Some of our recent work concerned the interactions of twisted wave fronts of acoustic vortex beams with boundaries and inhomogeneities of media. Experimental measurements and numerical simulations were conducted to investigate vortex beam reflection from a water-air interface and vortex beam refraction when travelling in an inhomogeneous medium or through an acoustic metasurface. The results reveal spatial evolution of the wave fields and the carried angular momenta. In particular, the investigation observed a reversal of orbital angular momentum in the reflected wave field [Zou *et al*., Phys. Rev. Lett. 125(7), 074301 (2020)], and a shift of beam center and an asymmetric vortex pattern in the refracted wave field in the direction transverse to the plane of incidence [Fan *et al*., Phys. Rev. Res. 1(3), 032014 (2019); 3(1), 013251 (2021)]. The shift relates to conservation of angular momenta and coupling of the twisted wave fronts with the refraction and is termed as acoustic orbital Hall effect.

Contributed Paper

11:20

2aPAb10. Implementation of an ultrasonic fluid manipulation experiment on the International Space Station. Robert L. Lirette (National Ctr. for Physical Acoust. and Dept. of Phys. and Astronomy, Univ. of Mississippi, 502 Park Ln., Oxford, MS 38655, rlirette@go.olemiss.edu) and Likun Zhang (National Ctr. for Physical Acoust. and Dept. of Phys. and Astronomy, Univ. of Mississippi, University, MS)

The International Space Station (ISS) provides a unique opportunity to investigate acoustic radiation force interactions with a minimal interference from gravity. There are many design considerations to make when creating an experiment that is suited for use on board the ISS. NASA's primary concerns with all experiments implemented in the ISS are with the safety of the crew and the success of the mission. Acoustic experiments are surprisingly rare on the ISS and present additional unique design challenges. This is especially true for experiments operated under water. Our work implements a high-intensity acoustic vortex beam under water for single droplet tractoring and capture operated inside the Microgravity Science Glovebox (MSG) on the ISS. The experiment payload consists of a signal generator, an amplifier module, and a sealed water chamber. A secondary fluid immiscible with water is injected into the chamber for acoustic manipulation, and the results of the experiment are recorded as high-definition video.

Invited Paper

11:35

2aPAb11. Acoustic angular momenta and their applications. Chengzhi Shi (GWW School of Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr NW, Atlanta, GA 30332, chengzhi.shi@me.gatech.edu)

Acoustic angular momentum is a physical quantity characterizing the rotation of pressure and local particle velocity fields. The acoustic angular momentum was first observed in vortex waves and was found to have many interesting physics including pseudospin based topological effects, spin-momentum locking, and trapping potential for contactless particle manipulations. In this presentation, we will introduce the categorization of acoustic angular momentum into spin and orbital angular momenta. While airborne sound is a longitudinal wave that was long thought to be spinless, we observed the acoustic spin in some special cases. The spin-momentum locking and spin induced torque for particle manipulation were experimentally demonstrated. In addition, the application of acoustic orbital angular momentum in high-speed communication will be discussed. These novel physical acoustic properties have many more applications ranging from underwater exploration to biomedical engineering.

TUESDAY MORNING, 24 MAY 2022 PLAZA BALLROOM D, 8:30 A.M. TO 11:55 A.M.

Session 2aPP

Psychological and Physiological Acoustics and Speech Communication: Age-Related Changes in Mechanisms of Speech Perception

Matthew J. Goupell, Chair
Hearing and Speech Sciences, University of Maryland-College Park, 7251 Preinkert Drive, 0141 Lefrak Hall, College Park, MD 20742

Chair's Introduction—8:30

Invited Paper

8:35

2aPP1. Effects of age on brainstem coding of speech glimpses in interrupted noise. William J. Bologna (Speech-Lang. Pathol. & Audiol., Towson Univ., 8000 York Rd., Towson, MD 21252, wbologna@towson.edu), Michelle R. Molis, Brandon M. Madsen (National Ctr. for Rehabilitative Auditory Res., VA Portland Medical System, Portland, OR), and Curtis J. Billings (Commun. Sci., Idaho State Univ., Pocatello, ID)

Difficulty understanding speech in fluctuating backgrounds is common among older adults. Previous research suggests that this age-related decline may be due to increased susceptibility to forward masking and difficulty incorporating short "glimpses" of speech into a coherent auditory object. These findings may be linked via a common brainstem mechanism, such that brief glimpses of speech surrounded by noise are not faithfully represented in the neural code that reaches the cortex. This hypothesis was tested using electrophysiological recordings of the envelope following response (EFR) elicited by glimpses of speech-like stimuli varying in duration (42, 70, and 210 ms) and separated by silence or intervening noise. Responses from adults aged 23–73 years indicated that older adults produced weaker neural responses than younger adults. Across all ages, declines in EFR strength were noted with shorter glimpses of speech and with the addition of intervening noise. However, the extent of these declines was not associated with participant age. These results suggest that the EFR is sensitive to factors commonly associated with glimpsing but may not entirely reflect age-related changes that affect speech recognition in fluctuating backgrounds. [Work supported by NIH TL1TR002371, R01DC015240, R01DC012313, and VA-RR&D 5I01RX002139.]

Contributed Paper

9:10

2aPP2. Comparing the monaural effects of reverberation on speech intelligibility in noise for hearing-impaired and normal-hearing listeners. Raphael Cueille (ENTPE, 3 rue Maurice Audin, Vaulx-en-Velin 69120, France, raphael.cueille@entpe.fr), Mathieu Lavandier (ENTPE, Lyon, France), and Nicolas Grimault (CRNL, Lyon, France)

Reverberation has several detrimental effects on speech intelligibility in noise. They have been extensively studied with normal-hearing listeners. These effects are often simultaneous, and it is hard to disentangle the corresponding difficulties experienced by hearing-impaired listeners in noisy reverberant rooms. In this study, two monaural effects of reverberation were investigated for normal-hearing and hearing-impaired listeners with linear amplification (NAL-RP) to compensate for audibility. One is the temporal smearing of the target speech, which makes it less intelligible. The other is the temporal smearing of the noise masker, which decreases the dip listening benefit associated with envelope modulations in the noise. These two effects were studied separately and in combination, by applying reverberation either on the target speech, on the noise masker or on both. The results were analyzed using Bayesian ANOVAs and t-tests. Overall, the detrimental influence of reverberation was found similar for the hearing-impaired and normal-hearing listeners, indicating that a linear amplification of the signals seemed sufficient to compensate for the potential differences experienced by normal-hearing and hearing-impaired listeners.

Invited Papers

9:25

2aPP3. Susceptibility to signal distortion in older listeners with hearing loss. Varsha H. Rallapalli (Commun. Sci. & Disord., Northwestern Univ., 2240 Campus Dr., Evanston, IL 60208, varsha.rallapalli@northwestern.edu), Pamela E. Souza (Northwestern Univ., Evanston, IL), and Kathryn Arehart (Univ. of Colorado Boulder, Boulder, CO)

Older adults show significant individual variability in hearing aid benefit. Our recent work has identified factors contributing to this individual variability including age, degree of hearing loss, and working memory. Specifically, older adults with greater hearing loss and poorer working memory are disadvantaged (poorer speech recognition) by signal distortions arising from adverse listening conditions and aggressive hearing aid signal processing settings. In this presentation, we focus on the range of signal distortions (quantified by a cepstral correlation metric) due to hearing aid signal processing such as amplitude compression, frequency lowering, and digital noise reduction, combined with different amounts and types of background noise. Across studies, we characterized individual susceptibility to signal distortions with a comprehensive set of conditions ranging from laboratory simulations to realistic listening conditions. Earlier studies considered omnidirectional settings and co-located speech and noise conditions, whereas recent work has focused on clinically realistic conditions such as wearable hearing aids, microphone directionality, and spatially-separated speech and multi-talker babble. We will discuss the clinical implications of these findings, particularly the impact on customizing hearing aid fittings to maximize benefit for older individuals. [Work is funded by NIDCD.]

10:00–10:15 Break

10:15

2aPP4. Assessing age-related temporal processing deficits with single word contrasts in cochlear-implant users. Matthew J. Goupell (Hearing and Speech Sci., Univ. of Maryland-College Park, 7251 Preinkert Dr., 0141 Lefrak Hall, College Park, MD 20742, goupell@umd.edu), Zilong Xie (Hearing and Speech, Univ. of Kansas Medical Ctr., Kansas City, KS), Casey R. Gaskins, Samira Anderson, and Sandra Gordon-Salant (Hearing and Speech Sci., Univ. of Maryland-College Park, College Park, MD)

For listeners with acoustic hearing, aging and hearing loss are associated with temporal processing deficits. For those with enough hearing loss, a cochlear implant (CI) becomes the preferred intervention. However, a CI degrades acoustic information by delivering primarily temporal envelope information through modulated electrical pulse trains. Therefore, if older CI listeners experience temporal processing deficits like acoustic-hearing listeners, they are compelled to understand speech primarily through a process that is affected by aging. We will present temporal processing data from a series of experiments that use word contrasts that vary primarily in a temporal dimension (silence duration: Dish-Ditch; voice-onset time: Buy-Pie). Participants include CI listeners and normal-hearing listeners presented with a CI simulation via a channel vocoder and range in age from younger (<45 yrs) to older (>65 yrs). Results show that CI listeners and NH listeners presented with CI simulations need longer temporal cues to discriminate temporally based word contrasts. In addition, such deficits are differentially worse at higher stimulation levels for the older CI listeners compared to the younger CI and all NH listeners. In conclusion, aging may lead to temporal processing deficits in CI listeners. These results have implications for CI programming and directions for technological improvements.

10:50

2aPP5. Aging effects on listening effort in cochlear-implant users. Kristina DeRoy Milvae (Hearing and Speech Sci., Univ. of Maryland-College Park, 7251 Preinkert Dr., College Park, MD 20742, kmilvae@umd.edu), Jordan C. Abramowitz (Hearing and Speech Sci., Univ. of Maryland-College Park, College Park, MD), Stefanie E. Kuchinsky (Audiol. and Speech Pathol., Walter Reed National Military Med. Ctr., Bethesda, MD), and Matthew J. Goupell (Hearing and Speech Sci., Univ. of Maryland-College Park, College Park, MD)

Cochlear-implant (CI) users often report that listening to speech can be exhausting. Greater listening effort can occur with a degraded acoustic signal even in individuals with normal hearing, but age-related changes in auditory processing and individual differences in cognitive abilities may exacerbate the effort required to process a degraded signal. Therefore, we examined the effects of aging and individual differences in cognitive abilities on listening effort (measured with pupillometry) in two experiments. In the first experiment, we measured how changes in signal degradation (spectral resolution) affect listening effort in younger and older adults with normal hearing recalling CI-simulated (vocoded) sentences. In the second experiment, we examined how memory load and cognitive abilities (individual differences in working memory scores) affect listening effort in CI users recalling digit strings. It was hypothesized that increased effort would be observed with greater signal degradation, memory load, and age, and that individual differences would relate to working memory scores. Preliminary results are consistent with the hypothesized effects. These results support the clinical relevance of listening effort in older CI users and provide motivation for development of clinical practices to reduce listening effort in this population.

Contributed Papers

11:25

2aPP6. Listening effort over time: Single-trial electrophysiological measures of listening effort in middle-aged adults. Cynthia Hunter (Speech-Language-Hearing, Univ. of Kansas, 1000 Sunnyside Ave. DHDC 3000, Lawrence, KS 66045, c.hunter@ku.edu)

Listening effort refers to the use of working memory and/or attentional resources to understand speech. In the current study, it was hypothesized that electrophysiological measures of listening effort, specifically oscillatory power in the alpha (8–12 Hz) and theta (4–8 Hz) frequency bands, would be affected by signal-to-noise ratio (SNR), reflecting increased effort in more difficult listening conditions. Furthermore, changes in effort across trials were assessed as interactions with trial number. Although most studies of effects of aging on listening effort have compared young and older adults, impacts of aging on hearing and cognition begin in middle-age. Here, middle-aged adults ($n = 11$; age range = 37—65 years) with and without hearing loss listened to spoken sentences in a background of multi-talker babble at a range of SNRs relative to an individualized SNR corresponding to the speech recognition threshold. Each electrophysiological measure tracked with SNR, such that alpha power decreased for lower (poorer) SNRs, whereas theta power increased at lower SNRs. Notably, the linear effect of SNR on alpha power emerged only at later trials. Overall, results support use of these measures as neural markers of listening effort during speech perception in middle-aged adults but indicate that alpha power measures of listening effort may emerge gradually over an episode of listening, perhaps reflecting listening-related fatigue, in contrast to theta power effects, which may emerge more quickly.

11:40

2aPP7. Cortical encoding and cognition in reverberant speech perception in older adults. Ramesh Kumar Muralimanohar (Dept. of Speech, Lang., and Hearing Sci., Univ. of Colorado Boulder, 20954 SW Edgemont St., Beaverton, OR 97003, muralima@colorado.edu), Lauren Charney (Dept. of Speech-Lang. Pathol. and Audiol., Towson Univ., Towson, MD), Marcin Wróblewski (School of Audiol., Pacific Univ., Hillsboro, OR), Michelle R. Molis (NCRAR, VA RR&D, Portland, OR), and Curtis J. Billings (Dept. of Commun. Sci. & Disord., Idaho State Univ., Pocatello, ID)

Older listeners are more susceptible than younger listeners to noise and reverberation when communicating in everyday environments. This study investigated the effects of hearing loss and cognitive processing ability on older individuals' ability to perceive reverberant speech. The effects of reverberation were examined using three types of measures (1) electrophysiology: cortical auditory evoked potentials (CAEPs), (2) cognition: working memory and inhibitory control aspect of executive function, and (3) speech perception: syllable discrimination and sentence intelligibility. These assessments were carried out in two age- and sex-matched groups of ten older adults (>50 years) with either normal hearing (thresholds ≤ 25 dB HL ≤ 4 kHz) or with mild-to-moderate hearing loss. First, the effect of reverberation on the cortical encoding in each group was determined; then, the effects of hearing loss and cognitive function on syllable and sentence perception were examined. Preliminary analyses showed that increasing reverberation negatively impacts speech perception and alters the amplitude and timing of peaks in evoked responses. Differential effects of reverberation across group will also be discussed. The outcomes of this study will help design clinically applicable models to assess individual susceptibility to environmental interference. [Work supported by an Early Clinical Investigator Grant from Medical Research Foundation of Oregon to R.K.M.]

Session 2aSAa

Structural Acoustics and Vibration and Physical Acoustics: Willis Coupling in Acoustic and Elastic Media

Stephanie G. Konarski, Cochair
US Naval Research Laboratory, 4555 Overlook Ave. SW, Washington, DC 20375

Caleb F. Sieck, Cochair
Code 7160, U.S. Naval Research Laboratory, 4555 Overlook Ave. SW, Washington, DC 20375

A. J. Lawrence, Cochair
Walker Department of Mechanical Engineering, University of Texas at Austin, 204 E. Dean Keeton Street, Stop C2200, Austin, TX 78712-1591

Chair's Introduction—8:15

Invited Paper

8:20

2aSAa1. Polarizability of electro-momentum coupled scatterers. Matthew A. Casali (Walker Dept. of Mech. Eng. & Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Samuel P. Wallen, Benjamin M. Goldsberry (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Michael R. Haberman (Walker Dept. of Mech. Eng. & Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, haberman@utexas.edu)

The macroscopic response of acoustic metamaterials with sub-wavelength asymmetry is described with coupled constitutive relations for the acoustic pressure and momentum density. This coupling leads to momentum that is dependent on the strain rate and pressure that depends on the local acceleration. The coupled constitutive form has become known as the Willis form because they were first predicted by John Willis [Wave Motion, **3**(1), 1–11 (1981)]. The subwavelength behavior of the Willis material building-blocks can be described by a polarizability matrix relating the monopole and dipole scattering moments to the local pressure and velocities fields when off-diagonal terms are non-zero [Phys. Rev. B. **96**(10), 104303 (2017)], providing a metric for the design of microscale structures leading to macroscopically observable Willis coupling. More recently, Pernas-Solomon and Shmuel showed that heterogeneous piezoelectric media with subwavelength asymmetry can lead to coupling between pressure, momentum density, and electric displacement fields, a material response known as electro-momentum coupling [J. Mech. Phys. Solids, **34**, 103770 (2020)]. This work will present a polarizability description of electro-momentum coupled scatterers and the means to calculate the polarizabilities that couple local pressure, velocity, and electric fields with the aim of creating metamaterials for acoustic sensing. [Work supported by DARPA.]

Contributed Papers

8:40

2aSAa2. Willis elastic metasurfaces for unidirectional manipulation of flexural waves. Katerina Stojanoska (Dept. of Mech. Eng., Rowan Univ., 201 Mullica Hill Rd., Glassboro, NJ 08028, stojan29@students.rowan.edu) and Chen Shen (Dept. of Mech. Eng., Rowan Univ., Glassboro, NJ)

Exceptional points label the region of the parameter space of a system in which two or more eigenvalues of a non-Hermitian Hamiltonian and their associated eigenvectors coalesce and become degenerate. Owing to the rich physics and engineering potential, there has been a recent interest in developing such acoustic or elastic media that display exceptional points. In this work, we present the design of a non-Hermitian planar elastics metasurface exhibiting Willis coupling for the control of flexural waves. The metasurface contains a series of unit cells with asymmetrically loaded piezoelectric and metallic blocks. Loss is further added to the system by negative capacitance piezoelectric shunting, which leads to zero reflection from one side of the metasurface. Asymmetric reflection properties are achieved when approaching the exceptional point of the system. Numerical simulations show that the proposed metasurface can realize asymmetric modulation of flexural waves propagation on the beam, manifested by unidirectional focusing of the incident waves. The proposed elastic metasurface has a planar profile, and the amount of loss can be conveniently tuned by the shunted piezo, making it useful for the manipulation of flexural waves where asymmetric control is desired.

8:55

2aSAa3. Highly efficient wavefront control through bianisotropic (Willis coupling) metasurfaces. Junfei Li (Elec. and Comput. Eng., Duke Univ., Durham, NC, junfei.li@duke.edu), Xiuyuan Peng (Elec. and Comput. Eng., Duke Univ., Durham, NC), Ailing Song (School of Mech. and Power Eng., East China Univ. of Sci. and Technol., Shanghai, China), and Steven Cummer (Elec. and Comput. Eng., Duke Univ., Durham, NC)

Recent advances in metasurfaces have shown that efficient wavefront transformation can be obtained by carefully controlling the bianisotropic response (Willis coupling) in the metasurfaces. However, applying such a scheme directly for complicated wavefront transformations will lead to nonreciprocal or nonlocal response, which is difficult to realize physically. Here

we identify the local power conservation requirements and introduce several strategies to meet such requirements. Our strategies include designing surface waves into the scattered field, designing power-flow conformal metasurfaces, and combining coordinate transformation with bianisotropic metasurfaces. Based on such schemes, we develop a series of bianisotropic metasurfaces design approachs for near-perfect arbitrary wavefront control. These passive and reciprocal design approaches enable experimental demonstration of highly efficient, arbitrary wavefront transformation.

9:10

2aSAa4. Willis coupling in strictly one-dimensional bulk media. Michael B. Muhlestein (Cold Regions Res. and Eng. Lab., U.S. Army Eng. Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755, Michael.B.Muhlestein@usace.army.mil)

Willis coupling, which couples the constitutive equations of an acoustical material, has been applied to acoustic metasurfaces with promising results. However, less is understood about Willis coupling in bulk media. In this talk, a variational homogenization method is used analyze the source and interpretation of Willis coupling in one-dimensional bulk media without any hidden degrees of freedom, or strictly one-dimensional media. As expected from previous work, Willis coupling is shown to arise from geometric asymmetries, but is further shown to depend greatly on the measurement position. In addition, a discussion of the predicted material properties, including Willis coupling, of macroscopically inhomogeneous media is presented.

9:25

2aSAa5. Extraction of dynamic effective material properties for 2D acoustic Willis media. Stephanie G. Konarski (Code 7165, US Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375, stephanie.konarski@nrl.navy.mil) and Caleb F. Sieck (Code 7165, US Naval Res. Lab., Washington, DC)

Acoustic Willis media is characterized by coupling between the kinetic and potential energy through the pressure-strain and momentum density-velocity relationships. The introduction of additional constitutive properties allows more appropriate characterization of complex unit cells, including effects related to physical asymmetry, finite phase and multiple scattering, and nonreciprocal biases. Similar to traditional composites or other metamaterials, homogenization is a valuable modeling tool that simplifies numerical study of Willis media. However, homogenization schemes must be altered to account for even and/or odd Willis coupling parameters. In this work, we extend a 2D homogenization approach to account for the Willis coupling using the modified dynamic equations. With this homogenization scheme, finite layers of N unit cells and N + 1 unit cells are utilized with different propagation angles and incident directions to isolate the response across a single unit cell. Numerical simulations with the finite element method will then be presented to validate the homogenization model and demonstrate the effective material properties obtained for an asymmetric unit cell. [This work was supported by the Office of Naval Research.]

Session 2aSAb

Structural Acoustics and Vibration, Engineering Acoustics, and Physical Acoustics: Acoustic Metamaterials I

Christina Naify, Cochair
Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758

Kayla Petrover, Cochair
NSWC, 9500 MacArthur Blvd., Bethesda, MD 20817

Bogdan-Ioan Popa, Cochair
Univ. of Michigan, 2350 Hayward St., Ann Arbor, MI 48109

Alexey Titovich, Cochair
Carderock Div., Naval Undersea Warfare Center, 9500 MacArthur Blvd., West Bethesda, MD 20817-5700

Chair's Introduction—10:00

Invited Papers

10:05

2aSAb1. Energy harvesting using an acoustic metamaterial. Amr M. Baz (Mech. Eng., Univ. of Maryland, 2137 Eng. Bldg., College Park, MD 20855, baz@umd.edu)

Acoustic energy is harvested using a piezoelectric-based metamaterial. The considered metamaterial harvester consists of a multi-cell array of acoustic cavities, which are provided with piezoelectric boundaries in a one-dimensional arrangement. Once impacted by an incident parasitic acoustical energy, these boundaries convert this acoustical energy into useful electrical energy. An *ANSYS* finite element model is developed to model the dynamics of the multi-field energy harvesting, predict the harnessed power, and optimize the performance of the piezoelectric-based metamaterial when coupled with an external load. The predictions of the *ANSYS* model are validated against the predictions of a lumped-parameter model of the harvester, which is based on the equivalent electrical analog of the harvester. Excellent agreement is observed between the predictions of *ANSYS* and the lumped-parameter models. The predictions of the models are validated experimentally using a prototype of the harvester consisting of five cells each of which is manufactured from acrylic cylinders provided with piezoelectric bimorphs This arrangement enables harnessing the energy by the two sides of the bimorph in order to maximize the extracted power of the harvester. The frequency characteristics of the output power of the harvester are determined in relation to the bandgap characteristics of the periodically structured metamaterial. The presented work lays down the foundation for two and three-dimensional metamaterial energy harvesters.

10:25

2aSAb2. Acoustic metamaterials with independently tunable mass, damping, and stiffness. Vinod Ramakrishnan (Mech. and Aerosp. Eng., Univ. of California, San Diego, San Diego, CA) and Michael J. Frazier (Mech. and Aerosp. Eng., Univ. of California, San Diego, 9500 Gilman Dr., MC-0411, San Diego, CA 92093-0411, mjfrazier@ucsd.edu)

Over the past decades, metamaterials—whose engineered internal architecture grants unusual or extraordinary macroscopic response—have garnered increasing attention from researchers as the desire to shape material behavior beyond natural limitations (e.g., chemistry) arises within several areas of materials science and engineering. The bulk of reported acoustic metamaterial architectures are passive such that their properties and functions are fixed at fabrication. Nevertheless, a tuning capacity is desirable to expand the range of response, in general, and to allow for adaptation in the face of changing service requirements. Despite the diversity of proposed tuning strategies, most target the stiffness parameter alone, leaving the inertial and dissipative properties unaffected. In this presentation, we present a novel implementation of (geometric) bi-stability and kinematic amplification to independently tune the value and distribution of the effective mass, stiffness, and viscous damping within acoustic metamaterials, which impacts the dynamic response. Through analytical and numerical investigations of a 1D system, we show that the corresponding frequency band structure depends on the specific configurations of bi-stable elements within the unit cell. As the number of bi-stable elements per unit cell increases, so to do the number of unique dynamic responses to which to tailor the system. The proposed strategy significantly expands the property set available for tuning acoustic metamaterial performance post-fabrication.

10:45

2aSAb3. Symmetry-enforced gapless surface states in three-dimensional acoustic gyroid structures. Yuning Guo (Dept. of Mech. Eng., Univ. of Colorado Boulder, 427 UCB, 1111 Eng. Dr., Boulder, CO 80309, yuning.guo@colorado.edu), Matheus Rosa, and Massimo Ruzzene (Dept. of Mech. Eng., Univ. of Colorado Boulder, Boulder, CO)

The discovery of topological gapless phases challenges the perception that topological features necessarily require a bandgap, expanding the understanding of topological phases of matter in various realms including electric, photonic, and phononic systems. The progress on 3D topological gapless states in elastic and acoustic systems is still in its early stages of formulation and design. We here investigate 3D acoustic gyroid crystals supporting symmetry-enforced gapless surface states in minimal surface-based structures. The inherent chirality and morphology of gyroid surfaces enable the implementation of 3D acoustic crystals hosting symmetry-enforced Dirac points and topologically gapless surface states. The associated fourfold degeneracy is protected by the nonsymmorphic space group featuring a combination of screw symmetry and glide reflections. The presence of gapless surface arcs relies on band structure calculations conducted using finite element simulations, while preliminary experimental results on additively manufactured samples validate their occurrence in the proposed gyroid surfaces. With the continuous development in additive manufacturing techniques, the presented surface-based framework provides a platform to explore a variety of topological wave physics phenomena in 3D load-bearing, continuum materials of potential engineering relevance, among which superior acoustic absorption may be particularly promising.

11:00

2aSAb4. Anomalous polarization in asymmetric gyroid structures. Saranchana Keattitorn (Aerosp. Eng., Wichita State Univ., 1845 Fairmount St., Wichita, KS 67260, sxkeattitorn@shockers.wichita.edu), Maria Carrillo-Munoz, and Bhisham Sharma (Aerosp. Eng., Wichita State Univ., Wichita, KS)

Recent studies show that structures with triply periodic minimal surfaces (TPMS) provide enhanced mechanical, acoustical, and energy abortion performance. Previously, we have shown that breaking the symmetry of the gyroid lattice—one of the most used TPMS geometry—results in the creation of directional and polarized bandgaps. Here, we focus on the effect of breaking symmetry on the effective wave speeds of the gyroid structure. We analyze the wave speeds of different asymmetric gyroid lattices using the finite element analysis approach. Our analysis shows that certain gyroid asymmetries result in the transverse waves propagating faster than the longitudinal waves in particular direction. Our research shows that breaking the symmetry leads to previously unobserved anomalous polarization of elastic waves in asymmetric gyroids.

11:15

2aSAb5. Phononic crystal with rigid scatterers: Infinite density versus infinite elasticity. Dmitrii Shymkiv (Phys., Univ. of North Texas, PO Box 311427, Denton, TX 76203, dmytroshymkiv@my.unt.edu) and Arkadii Krokhin (Phys., Univ. of North Texas, Denton, TX)

Models involving materials with extreme parameters, while usually unreachable in practice, play important role in different areas of physics, for example, black body and ideal gas in thermodynamics, perfect conductor, and ideal diamagnetic in electrodynamics. In acoustics, the concept of rigid (or hard) scatterer is widely used to study propagation of sound in heterogeneous media with high-acoustic contrast between the constituents. Here, we report the results obtained for band structure calculations of phononic crystals with rigid scatterers. A scatterer with infinite acoustic impedance is modeled by approaching either the mass density (ρ) or the elastic modulus (λ) to infinity. It is shown, using the plane-wave expansion method, that in both cases the dispersion equation contains singular matrices with elements related to the form factor of the crystal lattice. However, this singularity does not affect calculations of the band structure in the case $\lambda \rightarrow \square$. Unlike this, in the limiting case of infinite density, the dispersion equation becomes meaningless. We explain the mathematical reason of this drastic difference and propose a regularization numerical procedure. Our general results are illustrated by a particular case of elastic superlattice when the dispersion relation is available analytically.

11:30

2aSAb6. Enhanced energy trapping in a nonreciprocal phononic crystal cavity. Jyotsna Dhillon (Phys., Univ. of North Texas, 425 Bernard St., Apt 1116, Denton, TX 76201, jyotsnadhillon@my.unt.edu), Ezekiel Walker (Echonovus, Inc., Denton, TX), Arup Neogi, and Arkadii Krokhin (Phys., Univ. of North Texas, Denton, TX)

Acoustic energy trapping using defect modes in the band gap frequencies of acoustic metamaterials has been widely explored. Unlike this extensively used mechanism, the present work demonstrates the use of nonreciprocity in the transmission band to trap energy inside a phononic crystal cavity. A phononic crystal with broken parity and time reversal symmetry can be used to generate linear nonreciprocal transmission of the ultrasound waves. A gradient induced differential dissipation (GIDD) based passive non-reciprocal phononic crystal with asymmetric scatterers was employed to create three configurations of cavities. The parity of the present system is broken by the asymmetric shape of the scatterers, and the time reversal symmetry is naturally broken by viscous dissipation. The cavity configurations were based on the orientations of the asymmetric scatterers in part of the crystal that allowed utilization of non-reciprocity involved in only one of the cavities' configurations. Enhancement of energy trapping at a frequency of 622.5 kHz was observed experimentally for the cavity utilizing nonreciprocity compared to other cavity configurations. Experimental results were further confirmed and comprehended using finite element method based computational outcome. This energy trapping device is linear, robust, and allows sound energy trapping without an external energy source.

Session 2aSC

Speech Communication: Race, Racialization, and Racism in Speech Perception

Kristin J. Van Engen, Cochair
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Benjamin Munson, Cochair
University of Minnesota, 115 Shevlin Hall, Minneapolis, MN 55455

Chair's Introduction—9:00

Contributed Paper

9:05

2aSC1. Variability in the perception of race through voices and faces. Alayo Tripp (Dept. of Speech-Language-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, tripp158@umn.edu), Gisela Smith, and Benjamin Munson (Dept. of Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN)

Although perception of race is known to influence speech intelligibility, speech has often been studied without careful attention to this variability. It has been common to study language in an audio-only modality, which fails to capture the multimodal natures of language and person perception. We have developed a set of audiovisual study materials consisting of participants producing individual sentences selected from two different sets of standard materials. The corpus we have created is intentionally diverse with respect to racial self-report. However, there is extraordinary diversity in how persons who share a racial identification may present. The complex and variable relationship between physical features and the construction of social status demands that we reject reliance on self-reports when characterizing the diversity of our sample. In two experiments, participants viewed faces and listened to voices of people who contributed to the audiovisual corpus. The results allow us to explicitly characterize how the participants in the corpus may be racialized by observers, independent of their self-report. We argue that examining cognitive processes, which give rise to racialized and gendered perceptions, is foundational to advancing our understanding of speech perception in realistic environments with high degrees of variability in personal appearance and linguistic behavior.

Invited Papers

9:20

2aSC2. Socio-ethnic expectations of race in speech perception. Yolanda F. Holt (Commun. Sci. and Disord., East Carolina Univ., 300 Moye Bv 3310-X HSB, MS 668, Greenville, NC 27834, holty@ecu.edu) and Tessa Bent (Speech Lang. and Hearing Sci., Indiana Univ., Bloomington, IN)

The social construct of race, as a group of individuals to be visually or auditorily discriminated between, or from, or against is a human construction much like speech itself. This talk will share the analyses of a speech production and a speech perception experiment designed to evaluate how familiar and unfamiliar listeners parse the auditory spectral information contained in words to assign the speech token to a racial category. Familiar North Carolina listeners (n = 28) and unfamiliar Indiana listeners (n = 44) heard ten hVd words produced by three male and three female, Black and White talkers from two geographically distant dialect regions in east and west North Carolina (n = 24). Listener groups assigned most speech tokens to the correct racial category with greater than chance accuracy. Statistically significant differences in listener use of socio-ethnically aligned vowels as produced in the words *hid, heyd, head*, mid and low front vowels the back vowel as produced in *whod* and the diphthongs as produced in *hoyd, hide* and *howed* were noted. Results will be discussed in the context of socio-ethic talker group membership and patterns of both listener accuracy and listener error in assigning talkers to racial groups.

9:40

2aSC3. No news is good news: Social priming and the intelligibility of American-accented English. Drew J. McLaughlin (Psychol. & Brain Sci., Washington Univ. In St Louis, 1 Brookings Dr., St. Louis, MO 63130, drewjmclaughlin@wustl.edu) and Kristin J. Van Engen (Psychol. and Brain Sci., Washington Univ. in St. Louis, St. Louis, MO)

Social information, such as a speaker's race, can affect the perception of speech. In some cases, these *social primes* facilitate perception, while in others they can lead to reduced speech intelligibility. For example, a picture of an East Asian face may facilitate perception of Mandarin Chinese-accented English but interfere with perception of American-accented English. Using a large sample recruited online, we aimed to test whether there is a link between implicit racial associations and the effects of social primes on the perception of American-accented English. We examined priming for a White versus an East Asian prime and for a White versus a Latina prime. In

both cases, the difference between priming conditions was non-significant. Subjects also completed two implicit association tests, examining associations of these races/ethnicities with the constructs American versus Foreign and Good versus Bad. Individual differences in implicit associations did not predict social priming for either cohort. However, in some cases they did predict overall performance on the speech perception task. We discuss how these findings diverge from prior work on multiple accounts, and how consideration of the racial and linguistic diversity of listeners' interpersonal networks may improve our understanding of individual differences in social priming susceptibility.

10:00–10:15 Break

10:15

2aSC4. Perception in context: How racialized identities impact speech perception. Ethan Kutlu (Psychol. and Brain Sci., Univ. of Iowa, 111 Phillips Hall (16 N Clinton), Iowa City, IA 52242, ethankutlu@gmail.com)

When available, listeners use visual cues in speech perception [Zheng and Samuel (2017)]. However, it is not clear whether racialized identities impact listeners' judgments, and if so, to what extent everyday experiences contribute to this. American, British, and Indian English varieties were paired with white and South Asian faces to test whether listeners' intelligibility and accentedness judgments vary as a function of the faces that they saw and the varieties that they were listening to. A prior norming study was used to assess that sentences in all varieties had similar intelligibility. Listeners in a low-diverse environment (i.e., Gainesville, USA) versus a high-diverse environment (i.e., Montreal, Canada) were recruited. Racial diversity in listeners' social network and their language diversity [i.e., language entropy, Gullifer and Titone (2020)] were measured. Results showed that listeners' ability to transcribe sentences (i.e., intelligibility) decreased and their accentedness judgments increased for all English varieties when speech was paired with South Asian faces. Furthermore, these effects were modulated by participants' social network diversity and their geographic context [Kutlu *et al*. (2021); (2022)]. We discuss the holistic impacts of the racialization of different language varieties and the role of multilingual and diverse context effects on speech perception.

10:35

2aSC5. The role of perceived ethnicity in speech processing: Insights from diverse populations and methods. Adriana Hanulikova (Univ. of Freiburg, Germany, Platz der Universität 3, Freiburg 79098, Germany, adriana.hanulikova@germanistik.uni-freiburg.de)

In this talk, I will discuss behavioral and electrophysiological studies that examine the extent to which perceived talker identity influences speech comprehension and evaluation. Two frequently discussed theoretical accounts lead to different predictions. Bias-based accounts assume conscious misunderstanding of a standard variety in the case of a speaker classification as nonnative, resulting in negative ratings and poorer comprehension. Exemplar-based models suggest that such effects arise only when a contextual cue to the social identity is misleading, i.e., when ethnicity and speech clash with listeners' expectations. To address these accounts, and to assess ethnicity effects across different groups and methods, diverse non-university populations (172 monolinguals, 58 bilinguals, age range 12–92) were primed with photographs of Asian and white European women and asked to repeat and rate utterances spoken in three accents (Korean-accented German, a regional German accent, standard German), all embedded in background noise. In three electrophysiological studies, students (106 monolinguals, age range 18–30) passively listed to utterances in two accents (Turkish-accented and standard Dutch) and were either primed or not with photographs of a Turkic or a Dutch woman. The findings suggest that theoretical contradictions are a consequence of methodological choices, which reflect distinct aspects of social information processing.

Contributed Papers

10:55

2aSC6. Race, racialization, and racism in speech perception: Past, present, and future. Benjamin Munson (Speech-Language-Hearing Sci., Univ. of Minnesota, 115 Shevlin Hall, Minneapolis, MN, munso005@umn.edu) and Alayo Tripp (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN)

Individuals make inferences about people's race from the way they speak [Purnell *et al*. (1999)]. Moreover, people's assumptions about individuals' race shape the way that they perceive the form and content of the messages that people produce [Rubin (1992); Staum Casasanto (2008); McGowan (2015)]. In this talk, we step back from empirical studies on this topic and consider what it means to perceive a person's race. Our work situates the perception of talker race in the framework developed by Tripp and Munson (2021) to explain social perception and language perception more generally. We argue that racialization involves distinguishing groups of persons, which are abstractly defined, with perception measures depending upon the racial categories participants believe are perceptible, real, desirable, powerful and normative. The parsing of individuals into different imaginary person-kinds emerges from beliefs about the significance of differences between body-minds. Individuals' perception reveals an interplay among personally held ideologies, identities, and awareness of societally dominant narratives. We speculate on how these processes might have been at play in previous studies on the influence of race on speech perception. We provide a research agenda for robustly assessing the perception of person-kinds when studying how perceived race interacts with the perception of speech.

11:10

2aSC7. An analytical framework and model for de-racializing vocal perception. Nina S. Eidsheim (Musicology, Univ. of California, Los Angeles, 445 Charles E. Young Dr. East, 2686 Schoenberg Music Bldg., Los Angeles, CA 90095, neidsheim@ucla.edu)

Based on 20 years of research into vocal techniques, vocal styles, pedagogy, and history with reference to opera, popular music, and music synthesis software, I have developed a framework that seeks to explain racialized perceptions of the singing voice. On one hand, my model accounts for voice as an ever-developing instrument affected by age, hormones, environment, culture, and vocal training (whether through formal voice lessons or everyday encounters' largely tacit feedback). On the other hand, it also accounts for perception as equally dynamic and culturally dependent. It recognizes that the voice and its perception together constitute a "thick event," and that the complexity of the vocal signal and the ways in which listeners interact with voices are so numerous and so complex that a speaker's race cannot be defined by the signal. In other words: Voice is not singular; it is collective. Voice is not innate; it is cultural. Voice's source is not the singer; it is the listener. By applying this analytical model to musical case studies, this paper argues that voices are racialized when they are believed to sound a person's essence or true identity.

11:25–11:50
Panel Discussion

Session 2aSP

Signal Processing in Acoustics, Underwater Acoustics, Animal Bioacoustics, Physical Acoustics, and Engineering Acoustics: Cognitive SONAR

David A. Hague, Cochair
Naval Undersea Warfare Center, 1176 Howell St., Newport, RI 02841

Wu-Jung Lee, Cochair
Applied Physics Laboratory, University of Washington, 1013 NE 40th St., Seattle, WA 98105

Ananya Sen Gupta, Cochair
Department of Electrical and Computer Engineering, University of Iowa, 103 S Capitol Street, Iowa City, IA 52242

Chair's Introduction—8:30

Invited Papers

8:35

2aSP1. Toward a computational theory for the sensory world of bat biosonar. Rolf Müller (Mech. Eng., Virginia Tech, ICTAS II, 1075 Life Sci. Cir, (Mail Code 0917), Blacksburg, VA 24061, rolf.mueller@vt.edu), Sounak Chakrabarti (Mech. Eng., Virginia Tech, Blacksburg, VA), and Liujun Zhang (Elec. and Comput. Eng., Virginia Tech, Blacksburg, VA)

A sensory map of the world is a key component of any cognitive system. For cognitive systems navigating the physical world, this map is typically assumed to be a representation of the three-dimensional geometries in the environment. For vision-based maps, this is not an overly difficult goal to accomplish, since images have already two dimensions and hence only depth has to be inferred from additional clues. For bats using biosonar to navigate complex natural environments, such as dense vegetation, reconstruction of scatterer geometry is likely an ill-posed problem under the constraints of the biosonar systems (e.g., on beamwidth and spatial sampling). In these cases, biosonar echoes are "clutter," i.e., signals that must be regarded as unpredictable due to lack of knowledge. Deep learning offers an opportunity to explore the information content of these echoes and hence understand the sensory map of bat biosonar on small and large scales. In addition, the biosonar systems of many bat species that live in dense vegetation are highly active senses, where time-variant signal transformations of the emitted pulses and the returning echoes are controlled in a perception-action loop. Again, deep learning can be used to understand the information that is encoded in this loop.

8:55

2aSP2. Model-aided deep learning-based target detection for channel matrix-based cognitive radar/sonar. Touseef Ali (School of Elec., Comput., and Energy Eng., Arizona State Univ., Arizona State University, P.O. Box 875706, GWC 208, Tempe, AZ 85287-5706, tali4@asu.edu), Akshay S. Bondre (School of Elec., Comput., and Energy Eng., Arizona State Univ., Tempe, AZ), and Christ D. Richmond (Elec. and Comput. Eng., Duke Univ., Durham, NC)

Data driven based approaches to signal processing including deep neural networks (DNN) have shown promise in various fields. Such techniques tend to require significant training for good convergence. Model-based approaches, however, provide practical solutions often with insightful and intuitive interpretations. A hybrid approach that employs data driven techniques aided by knowledge from model-based approaches may help reduce required training and improve convergence rates. This work investigates the potential of deep learning techniques to detect targets while accelerating the learning process via use of expert/domain knowledge for the channel matrix based cognitive radar/sonar framework. The channel matrices characterize responses from target and clutter/reverberation. We will leverage our previous work [Ali *et al.* (2021); (2022)] on model-based adaptive detection approaches for cognitive radar/sonar. We compare the detection performance of model aided deep learning-based algorithms with that of traditional model-based techniques using receiver operating characteristic (ROC) curves from Monte Carlo simulations. We also study and compare the robustness of these techniques by changing the signal-to-interference plus noise ratio (SINR), the number of targets and clutter sources, and the amount of available training data. The effects of choice of hyperparameters and loss functions are also studied.

Contributed Paper

9:15

2aSP3. Deep reinforcement learning for cognitive active-sonar employment. Justin R. McMillan (Appl. Res. in Acoust. LLC, 305 S Main St., Madison, VA 22727, justin.mcmillan@ariacoustics.com), Jonathan Botts (Appl. Res. in Acoust. LLC, Madison, VA), and Jason E. Summers (Appl. Res. in Acoust. LLC, Washington, DC)

We introduce a framework to leverage deep reinforcement learning (RL) for active sonar employment, wherein we train an RL agent to select waveform parameters, which maximize the probability of single-target detection. We first simulate raw sonar returns of targets and clutter in reverberation and noise using a physics-based sonar-simulation model, the Sonar Simulation Toolkit (SST), then process the resulting signatures into network inputs via an in-house signal and information processing model of an archetypal anti-submarine warfare (ASW) processing chain. We demonstrate that the trained RL agent is able to appropriately select between continuous wave (CW) and hyperbolic frequency modulated (HFM) waveforms depending on target trajectory, as well as select an optimal bandwidth and pulse length trade-off (when constrained by a constant time-bandwidth product), when presented with sonar returns from a reverb-limited or noise-limited environment.

Invited Papers

9:30

2aSP4. Adaptive transmit waveform design for active cognitive sonar using multi-tone sinusoidal frequency modulation. David A. Hague (Naval Undersea Warfare Ctr., 1176 Howell St., Newport, RI 02841, david.a.hague@gmail.com)

Cognitive sonar systems leverage information gathered from earlier sensing interactions with the acoustic environment to adapt its system parameters for optimal performance. Of the many system parameters, an active cognitive sonar system could adapt, the acoustic signal projected into the medium, also known as the transmit waveform, has a profound impact on system performance as many of the physical characteristics of the acoustic environment are contained in the return echo signal. This research examines utilizing multi-tone sinusoidal frequency modulated (MTSFM) waveforms as an adaptive transmit waveform model for use in active cognitive sonar systems. The MTSFM waveform's frequency and phase modulation functions are composed of a finite set of weighted sinusoidal harmonics. The weights for each harmonic are utilized as a discrete set of design coefficients. Adjusting these coefficients results in constant amplitude, spectrally compact FM waveforms that possess a wide variety of performance characteristics, which in the past required a diverse set of waveform types to achieve. The adaptability of the MTSFM waveform model enables a cognitive sonar system to generate waveforms that are finely tuned for the novel scenarios and environments that it may encounter.

9:50

2aSP5. Are universal beamformers passive cognitive sonar systems? John R. Buck (Elec. and Comput. Eng., Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., Dartmouth, MA 02747, jbuck@umassd.edu), Kathleen E. Wage (George Mason Univ., Fairfax, VA), and Andrew C. Singer (Elec. and Comput. Eng., Univ. of Illinois, Urbana, IL)

Cognitive sonars dynamically tune system parameters to improve performance in pursuit of specific goals. Most research on cognitive sonar focuses on active sonar, varying the transmitted waveform and ping rate. Although passive sonar also faces the challenges of pursuing goals in varying and unknown environments, little research has been done exploring cognitive approaches to passive beamforming for sonar arrays. Universal beamformers provide one implementation of a cognitive approach to passive sonar. Practical adaptive beamformers generally regularize the sample covariance matrix before estimating the array weights. Universal beamformers blend the array weights across a family of beamformers competing on different choices for the regularization parameters. The blend of array weights is performance driven based, including the largest portion from the beamformers best suited for the current environment. Universal are "doubly adaptive" in that each competing beamformer is adapting its array weight vector in response to the data observed at the sensor array, the universal algorithm is then meta-adapting the blend of these array weight vectors used to process the data based on the performance of each competing beamformer. We will present examples of beamformers, which are universal over dominant subspace dimension and beam pattern notch width for moving interferers. [Work supported by ONR 321US.]

10:10–10:25 Break

Contributed Papers

10:25

2aSP6. Machine learning approach to prediction of real-time ocean sound speed profile. Madusanka Madiligama (National Ctr. for Physical Acoust. and Dept. of Phys. and Astronomy, Univ. of Mississippi, Dept. of Phys. and Astronomy, 108 Lewis Hall, University, MS 38677, mbabeyko@go.olemiss.edu), Zheguang Zou, and Likun Zhang (National Ctr. for Physical Acoust. and Dept. of Phys. and Astronomy, Univ. of Mississippi, University, MS)

Sound speed profile (SSP) determines sound propagation in the ocean. Variation of sound speed profiles with time and location heavily impacts underwater sonar systems used in ocean science and engineering. However, mapping sound speed profile over a vast area of the oceans in real-time is impossible by the traditional way, where sound speed is calculated from *in situ* profiling measurements of temperature, salinity, and pressure, which is expensive and, therefore, limited in sparse locations and time. Here, we aim to develop a machine learning model to predict real-time sound speed profiles anywhere in the global oceans. Our model takes advantage of the big data of long-term global temperature, salinity, and pressure measurements and further accounts for the variability of the surface ocean by inputting sea surface temperature and sea surface salinity data, which can be updated by satellite remote sensing in real-time. Here, SSPs are predicted for Pacific, Atlantic, and Indian ocean regions. The results show that the estimated SSPs had a root-mean-square error of 0.26 m/s and a coefficient

of determination of 0.99. About 99% of the estimates lie within $\pm$0.4 m/s of the SSPs obtained from *in situ* temperature and salinity profiles.

10:40

2aSP7. Automated acoustic arrival matching using a convolutional neural network approach informed by statistics of acoustic scattering from internal waves. Cristian E. Graupe (Ocean Eng., Univ. of Rhode Island, 30 Fish Rd., URI Narragansett Bay Campus Rm 13-14, Narragansett, RI 02882, graupec@uri.edu), Lora Van Uffelen (Ocean Eng., Univ. of Rhode Island, Narragansett, RI), and Bruce M. Howe (Dept. of Ocean and Resources Eng., Univ. of Hawaii at Manoa, Honolulu, HI)

A machine learning model was developed to automatically align underwater acoustic measurements taken at various depths and ranges from a transmitting source in the Philippine Sea to a reference model of long range acoustic arrival structure, simultaneously determining source-receiver range and travel-time offsets associated with multipath arrivals. Ocean sound-speed variability complicates the task as the measured arrivals may exhibit scattering not present in range-independent predictions. Monte Carlo style broadband parabolic equation simulations through random internal wave fields consistent with the Garrett-Munk internal wave energy spectrum were used to generate a large data set of simulated acoustic receptions including scattered multipath arrivals with known source-receiver ranges and imposed travel time offsets. These simulated receptions were used to train and evaluate the machine learning model for arrival pattern matching to the reference model. The inclusion of various data dimensions, such as peak amplitude and width, and contextual information, such as range and depth, were also explored as input to the model. Ranging results for the machine learning model were compared to a programmatic solution engineered for the same task.

Invited Paper

10:55

2aSP8. Sonar acoustic feature interpretation using braid manifolds. Ananya Sen Gupta (Dept. of Elec. and Comput. Eng., Univ. of Iowa, Iowa City, IA), Bernice Kubicek (Elec. and Comput. Eng., Univ. of Iowa, 103 South Capitol St., Iowa City, IA 52242, bernice-kubicek@uiowa.edu), Andrew J. Christensen, Timothy Linhardt (Elec. and Comput. Eng., Univ. of Iowa, Iowa City, IA), and Ivars Kirsteins (Naval Undersea Warfare Ctr., Newport, RI)

Autonomous target recognition has been a long-standing problem with complementary techniques based on classic subspace-based methods and recent machine cognition techniques proposed across several decades. Despite some success in both approaches, robust detection and meaningful interpretation of target features stay an open challenge. The primary bottleneck is the entanglement of multi-dimensional spectral features, some of which are intrinsic to target composition and geometry, and others originate from environmental reverberation and background clutter. This challenge is further compounded by unpredictable non-linear overlap between target features, which may themselves morph as a function of experimental conditions such as ping angle, range, frequency, and other factors. Furthermore, there is a compelling need to represent target features in a compact representation that lends itself to meaningful physical interpretation by a domain expert as well as robust interface with popular machine learning architectures. In this light, we will provide a broad overview of our recent and continuing work with braid manifolds, a morphological construct based on topological continuity of target features, that bridges this gap between physical model-based target representation and autonomous machine-learnt representation of target features. Representative results based on experimental field data on different target sizes and experimental conditions will be presented.

Contributed Papers

11:15

2aSP9. Sonar target feature representation using temporal graph networks. Andrew J. Christensen (Dept. of Elec. and Comput. Eng., Univ. of Iowa, 3100 Seamans Ctr., Iowa City, IA 52242, andrew-christensen@uiowa.edu), Ananya Sen Gupta (Dept. of Elec. and Comput. Eng., Univ. of Iowa, Iowa City, IA), and Ivars Kirsteins (NUWC, Newport, RI)

Autonomous sonar target recognition suffers from uncertainty caused by waveguide distortions to signal, unknown target geometry, and morphing target features. Typical "black-box" neural networks do not produce physically interpretable features and, therefore, are not effective in meeting these challenges. The primary objective of our work is to harness signal processing with machine learning to extract braided features that allow such physical interpretation by a domain expert. In this work, we introduce a feature extraction method using graph neural networks (GNNs) that seeks to discover braid manifolds from sonar magnitude spectra data. The approach involves representing the sonar magnitude spectra as sparse, dynamic graphs. These dynamic graphs can then be fed into a GNN as sequences of timed events to produce feature dictionaries that are resilient to environmental uncertainty and agnostic to ping direction. The ability of GNNs to learn complex systems of interactions makes them a great choice for braid-like feature discovery. To handle the evolving dynamic features of the sonar spectra graphs, a variation of a GNN, called temporal graph networks (TGNs), is used. TGNs utilize memory modules and graph-based operators to outperform previous GNN-based approaches when handling dynamic graphs. We use TGNs to model the evolution of the sonar spectra graphs and ultimately perform graph-based classification. Preliminary results performed on the Malta Plateau field experiment are presented.

11:30

2aSP10. Empirical analysis of latent space encodings for submerged small target acoustic backscattering data. Timothy Linhardt (Dept. of Elec. and Comput. Eng., Univ. of Iowa, 103 S Capitol St., Iowa City, IA 52242, tlinhardt@uiowa.edu) and Ananya Sen Gupta (Dept. of Elec. and Comput. Eng., Univ. of Iowa, Iowa City, IA)

With future sights on specialized classification methods, we work to generalize the acoustic backscattering data from sonar measurements of small targets submerged in water by learning a non-invertible mapping (encoding) to a low-dimensional vector space ($\mathbb{R}^n$). Finding the optimal dimensionality of this latent space is an important task. The encoding is accomplished by utilizing modality agnostic convolutional machine learning methods that have seen success in other signal and image processing domains. We have explored the autoencoder and its variants, the sparse autoencoder, and the variational autoencoder. Autoencoders encode input samples from a high-dimensional manifold to a lower latent vector space and then reverse the lossy mapping back to a high-dimensional manifold similar to the initial domain. The sparse autoencoder induces sparsity in the components of the latent vectors, and the variational autoencoder learns an encoding to n-dimensional gaussian distributions instead of n-dimensional vectors. The TREX13 data are the primary dataset used for training the networks used in experiments. We evaluate the change in models' accuracies as the latent dimensionality is increased as well as the models' ability to generalize to unseen data. Additionally, the PONDEX09 and PONDEX10 data are used to evaluate the models' cross-domain efficacy.

Meeting of the Standards Committee Plenary Group
to be held jointly with the meetings of the

ANSI-Accredited U.S. Technical Advisory Groups (TAGs) for:

ISO/TC 43, Acoustics,
ISO/TC 43/SC 1, Noise,
ISO/TC 43/SC 3, Underwater acoustics,
ISO/TC 108, Mechanical vibration, shock, and condition monitoring,
ISO/TC 108/SC 2, Measurement and evaluation of mechanical vibration and shock as applied to machines, vehicles, and structures,
ISO/TC 108/SC 4, Human exposure to mechanical vibration and shock,
ISO/TC 108/SC 5, Condition monitoring and diagnostics of machine systems,
and IEC/TC 29, Electroacoustics

C. J. Struck, Chair, L. A. Wilbur, Vice Chair, U.S. Technical Advisory Group for ISO/TC 43
CJS Labs, 57 States St., San Francisco, CA 94114
Northwestern University, 422 Skokie Blvd., Wilmette, IL 60091

S. J. Lind, Chair, R. D. Hellweg, Vice Chair, U.S. Technical Advisory Group for ISO/TC 43/SC 1 Noise
Air-Conditioning, Heating and Refrigeration Institute, 2111 Wilson Blvd., Suite 400, Arlington, VA 22201
Hellweg Acoustics, 13 Pine Tree Road, Wellesley MA 02482

A. Beaudry, Chair of the U.S. Technical Advisory Group for ISO/TC 43/SC 3 Noise
B. Underwater acoustics
Noise Control Engineering, Inc., 799 Middlesex Turnpike, Billerica, MA 01821

W. Madigosky, Chair of the U.S. Technical Advisory Group for ISO/TC 108 Mechanical vibration, shock, and condition monitoring
MTECH, 10754 Kinloch Road, Silver Spring, MD 20903

M. L'vov, Chair of the U.S. Technical Advisory Group for ISO/TC 108/SC 2 Measurement and evaluation of mechanical vibration and shock as applied to machines, vehicles, and structures
Siemens Energy, Inc., 5101 Westinghouse Blvd., Charlotte, NC 28273

L. Mullinix, Chair of the U.S. Technical Advisory Group for ISO/TC 108/SC 4 Human exposure to mechanical vibration and shock
Logan Mullinix Consulting, 7376 Tumblebrook Dr., New Albany, OH 43054

K. J. Culverson, Chair of the U.S. Technical Advisory Group for ISO/TC 108/SC 5 Condition monitoring and diagnostics of machine systems
2203 Fawn Dr., Dalton, GA 30720

C. Walber, U.S. Technical Advisor, C. J. Struck, Deputy Technical Advisor of the U.S. Technical Advisory Group for IEC/TC 29, Electroacoustics diagnostics of machine systems
PCB Piezotronics, Inc., 3425 Walden Avenue, Depew, NY 14043 2495
CJS Labs, 57 States St., San Francisco, CA 94114

The reports of the Chairs of these TAGs will not be presented at any other S Committee meeting

The Accredited Standards Committees S1, S2, S3, S3/SC 1, and S12, are scheduled to take place in the following sequence:

Accredited Standards Committee	Day and Date	Time
ASC S2 Mechanical Vibration and Shock	Tuesday, 14 May 2019	2:00 p.m.–3:15 p.m.
ASC S1, Acoustics	Tuesday, 14 May 2019	3:30 p.m.–4:45 p.m.
ASC S3, Bioacoustics	Tuesday, 14 May 2019	3:30 p.m.–4:45 p.m.
ASC S3/SC1, Animal Bioacoustics	Tuesday, 14 May 2019	3:30 p.m.–4:45 p.m.
ASC S12, Noise	Tuesday, 14 May 2019	5:00 p.m.–6:15 p.m.

Discussion at the Standards Committee Plenary Group meeting will consist of national items relevant to all S Committees and U.S. TAGs.

The U.S. Technical Advisory Group (TAG) Chairs for the various international Technical Committees and Subcommittees under ISO and IEC, which are parallel to S1, S2, S3, and S12 are as follows:

U.S. Technical Advisory Group (TAG)	Chair	Parallel S Committee
U.S. Technical Advisory Group for **ISO/TC 43**	C. J. Struck	ASC S1, Acoustics ASC S3, Bioacoustics
U.S. Technical Advisory Group for **ISO/TC 43/SC 1** Noise	S. J. Lind	ASC S12, Noise
ISO/TC 43/SC 3 Underwater acoustics	A. Beaudry	ASC S1, Acoustics ASC S3/SC1, Animal Bioacoustics ASC S12, Noise
U.S. Technical Advisory Group for **ISO/TC 108** Mechanical vibration shock, and condition monitoring	W. Madigosky	ASC S2 Mechanical Vibration and Shock
U.S. Technical Advisory Group for **ISO/TC 108/SC 4**	L. Mullinix	ASC S2 Mechanical Vibration and Shock
U.S. Technical Advisory Group for **ISO/TC 108/SC 5** Condition monitoring and diagnostics of machine systems	K. J. Culverson	ASC S2 Mechanical Vibration and Shock
U.S. Technical Advisor for **IEC/TC 29**, Electroacoustics	C. Walber	ASC S1, Acoustics ASC S3, Bioacoustics

TUESDAY, 24 MAY 2022 DIRECTORS ROW E, 11:00 A.M. TO 12:15 P.M.

Meeting of Accredited Standards Committee (ASC) S1 Acoustics

A. A. Scharine, Chair ASC S1
U.S. Army Research Laboratory, Human Research & Engineering Directorate
ATTN: RDRL-HRG, Building 459 Mulberry Point Road
Aberdeen Proving Ground, MD 21005 5425

R. J. Peppin, Vice Chair ASC S1
5012 Macon Road, Rockville, MD 20852

Accredited Standards Committee S1 on Acoustics. Working group chairs will report on the status of standards currently under development in the areas of physical acoustics, electroacoustics, sonics, ultrasonics, and underwater sound. Consideration will be given to new standards that might be needed over the next few years. Open discussion of committee reports is encouraged.

People interested in attending the meeting of the TAGs for ISO/TC 43 Acoustics, ISO/TC 43/SC 3, Underwater acoustics, and IEC/TC 29 Electroacoustics, take note that those meetings will be held in conjunction with the Standards Plenary meeting at 9:15 a.m. on Tuesday, 24 May, 2024.

Scope of S1: Standards, specifications, methods of measurement and test, and terminology in the field of physical acoustics, including architectural acoustics, electroacoustics, sonics and ultrasonics, and underwater sound, but excluding those aspects which pertain to biological safety, tolerance, and comfort.

Session 2pAA

Architectural Acoustics: Acoustic Comfort in Healthcare Facilities II

Lucky S. Tsaih, Cochair
Dept. of Architecture, National Taiwan Univ. of Sci. and Tech., 43 Keelung Rd., Sec. 4, Taipei, 10607, Taiwan

Joanne Solet, Cochair
Harvard Medi. School, Cambridge, MA 02138

Invited Paper

1:05

2pAA1. An overview on the impact of nursing homes' acoustic characteristics on the elderly's quality of life. Saleh Naseer (College of Design, Construction and Planning, Univ. of Florida, 2510 NE 9th St., #711, Gainesville, FL 32609, saleh.naseer@ufl.edu)

The primary motivation for this study is that the acoustical features of elderly care facilities are frequently neglected. According to the centers for disease control and prevention, around 1.5 million senior individuals are in nursing homes in the USA in 2021. Some have a lower quality of life (QOL) due to health care concerns, social issues, and the home environmental issues. This paper intends to study the current investigations on the environmental issues and specifically acoustical issues around these types of facilities. Since hearing loss is a common chronic illness among the elderly, the acoustical issues will result in stronger impacts, e.g., making it difficult to communicate with staff and other elderly. In return, improving the quality of the indoor acoustic environment can help alleviate the physical weakness of hearing loss, which directly impacts their QOL. Similarly, some of the other chronic illnesses can be alleviated by improving nursing home's acoustics quality, which can also result in an enhanced elderly's QOL. This study will provide a comprehensive overview by analyzing the methodology and results of the most relevant studies in this regard and making meaningful comparisons. This research aims to achieve an efficient research method for this type of study.

Contributed Papers

1:25

2pAA2. A qualitative approach to explore auditory and visual perception in a hospital environment. Semiha Yilmazer (Interior Architecture and Environ. Design, Bilkent Univ., Ray W Herrick Lab 177, S Russell St., West Lafayette, IN 47907, syilmaze@purdue.edu) and Zeynep M. Uğurlu (Interior Architecture and Environ. Design, Bilkent Univ., Ankara, Turkey)

This study aims to examine the effects of auditory and visual perception in a hospital environment to provide comfort for the patients. The research focused on the waiting area of an oncology polyclinic. The binaural audio recordings and 360° photographs were taken from three different locations on the site. Those locations differed from each other in their acoustic and visual settings. Regarding the hospital acoustic environment, sound sources are human activity-related and technology-based. The visual aspects show differences; the reception area has an indoor opening, the courtyard area has indoor and outdoor connections, and the corridor area has none of them. Voluntary oncology patients in three locations in the polyclinic were interviewed. The audio recordings were visualized through signal processing, and the photographs were analyzed through image processing in MATLAB in order to define acoustic and visual differences among the locations. LAeq measurements were taken within the interview hours and one-hour intervals from three areas. The conceptual framework created with semi-structured interviews showed how the acoustic and visual environments affect the patients' perception. The framework revealed audio-visual interactions in the hospital environment with its existing condition and preferred condition by the patients and proposed the suggestions to a hospital environment.

1:40

2pAA3. Case study: Acoustical design for newly renovated behavioral health suite. Peter Holst (TEECOM, 50 California St., Ste. 1500, San Francisco, CA 94111, peter.holst@teecom.com)

This presentation addresses requirements and opportunities for the acoustical design of a behavioral design suite. Examples of coordinated design elements will be presented from a recent remodel project for a single suite in the UCSF Langley Porter Psychiatric Hospital and Clinics Mt. Zion relocation. The following will be presented. Unique components: Equipment transcranial magnetic stimulation (TMS) is a noninvasive procedure that uses magnetic fields to stimulate nerve cells in the brain, used to address depression. TMS therapy treatment equipment has a loud tapping noise that can be highly annoying to occupants of adjacent rooms. Sound isolation and sound-absorbing treatments are important design elements for a room with this equipment. Unique requirements: Patient safety behavioral health suites have unique patient safety requirements that led to the design selection for acoustical tamper-proof ceiling panels and specializd door gaskets at specific room types. Design considerations: Group therapy room group therapy rooms are in-patient areas that can consist of small and large groups, sometimes involving simultaneous group activities among patients and staff. HIPAA requirements, speech intelligibility, and occupant comfort, as well as hospital operations were important considerations for the coordinated acoustical design.

1:55

2pAA4. Prefabricated partitions in a hospital clinic—Airborne sound isolation design and field measurements. Jeff Teel (Henderson Engineers, Inc., Lenexa, KS), Josh Thede (Henderson Engineers, Inc., 8345 Lenexa Dr. Ste. 300, Lenexa, KS 66214, josh.thede@hendersonengineers.com), Kevin Butler, and Elaina Bargas (Henderson Engineers, Inc., Lenexa, KS)

We will discuss the benefits, challenges, and requirements from an acoustics case study using modular prefabricated walls in healthcare. A recent hospital expansion/renovation project successfully used multi-trade prefab construction extensively in administrative, outpatient, inpatient, and surgical suite prep/recovery areas. This session will cover the acoustic and technical issues related to prefab partitions encountered during design and construction and post-construction acoustics observations and noise isolation class measurements.

2:10–2:40
Panel Discussion

TUESDAY AFTERNOON, 24 MAY 2022 GOVERNORS SQUARE 17, 1:00 P.M. TO 3:25 P.M.

Session 2pAB

Animal Bioacoustics, Acoustical Oceanography, and Underwater Acoustics: Whitlow Au Memorial Session II

Kelly Benoit-Bird, Cochair
Monterey Bay Aquarium Research Institute (MBARI), 7700 Sandholdt Road, Moss Landing, CA 95003

Marc Lammers, Cochair
Hawaii Institute of Marine Biology, 46-007 Lilipuna Rd., Kaneohe, HI 96744

Wu-Jung Lee, Cochair
Applied Physics Laboratory, University of Washington, 1013 NE 40th St., Seattle, WA 98105

Chair's Introduction—1:00

Invited Paper

1:05

2pAB1. How toothed whales hunt with echolocation. Peter T. Madsen (Marine Bioacoustics Lab, Biology, Aarhus Univ., Build 1131, CF Mollers Alle, Aarhus 8000, Denmark, peter.madsen@bio.au.dk)

The late Dr. Au developed a highly influential, quantitative framework for understanding the acoustic characteristics and performance of toothed whale biosonar systems. In his honor, I will apply this framework to the question of how echolocating toothed whales use their active sensory system to find, track and subdue prey in the field. Specifically, I will present data from echo-recording Dtags on wild toothed whales to show that they can detect, discriminate and approach prey with just a few tens of clicks, but that they use several hundreds of weak, high-rate clicks in the terminal buzz phase to track movements of elusive prey for capture in a very small sensory volume. I will use captive and wild data to demonstrate that toothed whales deliberately adjust their narrow beam sonars at short target ranges to receive weak prey echo levels within a 40-dB range above their hearing thresholds, and I will go on to argue that they do so to form simpler auditory scenes with less range ambiguity. I will end the talk by providing some perspectives on how Dr. Au's emphasis on the synergy between captive and field studies is more relevant than ever if we wish to understand how toothed whales operate their biosonars under natural and Anthropocene conditions.

Contributed Papers

1:30

2pAB2. Bioacoustics and beyond: Investigating animal sensory systems and cognitive processes. Caroline M. DeLong (Psych., Rochester Inst. of Technol., 18 Lomb Memorial Dr., Rochester, NY 14623, cmdgsh@rit.edu), Irene Fobe, Evan Morrison, and Jessica Wegman (Psych., Rochester Inst. of Technol., Rochester, NY)

Whit Au served on my dissertation committee at the University of Hawaii and mentored me while I was in the Marine Mammal Research Program at the Hawaii Institute of Marine Biology (1997–2003). He guided me through the process of recording echoes from objects ensonified by dolphins and oversaw my first human listening study. When I became a professor, I continued to study dolphin bioacoustics. I also broadened the scope of my research program to accommodate the interests of my students, just like Whit did. In this talk, I will summarize several animal perception and cognition projects I have pursued with my graduate students. First, we explored how humans discriminate acoustically among bottlenose dolphin signature whistles with and without masking by boat noise. In another study, we investigated auditory rhythm perception in African penguins. Next, we researched long-term memory in river otters using multimodal stimuli. One of our current projects focuses on visual object perception and learning in olive baboons. I will always be grateful for Whit's support when I was in graduate school and I will continue to follow his model for graduate student mentoring in the future.

1:45

2pAB3. Does rotation during echolocation increase the acoustic field of view? Comparative numerical models based on CT data of a live versus deceased dolphin. Chong Wei (Ctr. for Marine Sci. and Technol., Curtin Univ., Ctr. for Marine Sci. & Technol. Curtin University GPO Box U1987, Perth, Western Australia 6845, Australia, chong.wei@curtin.edu.au), Dorian Houser (National Marine Mammal Foundation, San Diego, CA), Christine Erbe (Ctr. for Marine Sci. and Technol., Curtin Univ., Perth, Western Australia, Australia), Chuang Zhang (Xiamen Univ., Xiamen, China), Eszter Matrai (Ocean Park Hong Kong, Hong Kong, Hong Kong), James J. Finneran (SSC Pacific Code 71510, US Navy Marine Mammal Program, San Diego, CA), and Whitlow W. Au (Hawaii Inst. of Marine Biology, Univ. of Hawaii, Kaneohe, HI)

Spinning is a natural and common dolphin behavior; however, its role in echolocation is unknown. We used computed tomography (CT) data of a live and a recently deceased bottlenose dolphin together with measurements of the acoustic properties of head tissues to perform acoustic property reconstrcution. The anatomical configuration and acoustic properties of the main forehead structures between the live and deceased dolphins were compared. Finite element analysis (FEA) was applied to simulate the generation and propagation of echolocation clicks, to compute their waveforms and spectra in both near- and far-fields, and to derive echolocation beam patterns. Model results from both the live and deceased dolphins were in good agreement with click recordings from live, echolocating individuals. FEA was also used to estimate the acoustic scene experienced by a dolphin rotating 180Ð about its longitudinal axis to detect fish in the far-field at elevation angles of 0Ð – 20Ð. The results suggest that the spinning behavior provides a wider insonification area and compensates for the dolphin's relatively narrow biosonar beam and constraints on the pointing direction that are limited by head movement. The results also have implications for examining the accuracy of FEA in acoustic simulations using freshly deceased specimens.

2:00

2pAB4. Broadband echolocation: From dolphins to drones. Lars N. Andersen (Ocean Sci., Kongsberg Maritime, Strandpromenaden 50, Horten 3191, Norway, lars.nonboe.andersen@km.kongsberg.com)

Broadband echolocation signals can be used to obtain detailed information about the underwater environment. The work performed by Whitlow W. L. Au on characterizing broadband clicks used by dolphins, studying the capabilities of the dolphin echolocation system, and exploring processing of the reflected signals has been and continues to be an important inspiration to the development and exploration of new echo sounder systems used by scientists world-wide to explore the oceans. This presentation will contain examples on Whit's invaluable inspiration.

2:15

2pAB5. Is broad angular coverage necessary for echolocation-based discrimination involving aspect-dependent targets? Wu-Jung Lee (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, wjlee@apl.washington.edu), Michael Ladegaard (Biology, Aarhus Univ., Aarhus, Denmark), John R. Buck (Elec. and Comput. Eng., UMass Dartmouth, Dartmouth, MA), Peter T. Madsen, Kristian Beedholm (Biology, Aarhus Univ., Aarhus, Denmark), and Peter Tyack (Univ. of St Andrews, Falmouth, MA)

Building on the foundation of Dr. Whitlow Au's pioneer and extensive work on the sonar of dolphins, recent research has illuminated the dynamic nature of echolocation involving free-swimming toothed whales. The sophisticated orchestration of movements and biosonar outputs by toothed whales is particularly relevant in the context of developing agile autonomous sonar-guided underwater vehicles. In this study, we test the hypothesis that broad angular coverage, which provide comprehensive echo information of aspect-dependent objects, is necessary for target discrimination via echolocation. We trained a harbor porpoise to actively approach and select a sphere against a spheroid in a two-alternative forced-choice target discrimination task in a net pan. Two spheroids of different aspect ratios representing two overall difficulty levels were used with each presented at diverse orientations when paired with the sphere. We show that the porpoise can discriminate these targets without acquiring aspect-dependent spheroid echo information over broad ranges of angles. Instead, focused, rapid sampling via buzzes appears to aid discrimination in more challenging scenarios. These results suggest that echolocation-based discrimination involving aspect-dependent objects is possible based on subsets of echo signatures without more extensive angular sampling that could build toward an understanding of the target identity. [Work supported by ONR.]

2:30

2pAB6. WAU! Identifying complex objects with an acoustic flashlight? Heidi E. Harley (Div. of Social Sci., New College of Florida, New College of Florida, 5800 Bay Shore Rd., Sarasota, FL 34243-2101, harley@ncf.edu)

A Hawaiian bunker, a bearded and barefoot entrepreneur, some dolphins, and an electrical engineer: A starter recipe for one of the most productive dolphin biosonar researchers in world history? Sound unlikely? Overstated? Apocryphal? And yet… Thirty years with Whit inspired us to investigate a dolphin's use of wholistic shape matching versus "flashlight" object-parts-focused matching. Our stimuli were 3 3-object sets (Sets W, A, U). Objects *within* sets varied on shape alone made with identical PVC parts, but PVC parts varied *between* sets. In Condition Shape, experienced dolphin Calvin matched objects based on shape within sets (matching W1Snail to W1Snail) in a 3-alternative echoic matching-to-sample task: mean accuracy across 12 18-trial sessions/set: Set W: 37%, Set A: 44%, Set U: 39%, chance = 33%. In Condition Parts/NoShape, Calvin performed untrained transfer tests with same-parts/different-shape objects as sample-match pairs within alternative arrays of objects from different sets with different parts (sample W1Snail matched to W2StingRay with alternatives A2Trumpet and U2Whale). Mean accuracy in ten transfer sessions/set (WAU1 to WAU2: 47%, WAU2-WAU3: 48%, WAU3-WAU1: 39%) was above-chance and comparable to Condition Shape even though shape information was absent for matching in this parts-focused condition. Calvin's narrow echolocation beam may have emphasized object parts versus wholistic shape.

2:45–2:55 Break

2:55–3:25

Panel Discussion

Session 2pBAa

Biomedical Acoustics: For a Few Bubbles More: Recent Developments in Medical Ultrasound II

John S. Allen, Cochair
Mechanical Engineering, University of Hawaii Manoa, Holmes 302, 2540 Dole Street, Honolulu, HI 96822

Mark Borden, Cochair
Biomedical Engineering, University of Colorado, 1111 Engineering Drive, Boulder, CO 80309-0427

Invited Papers

1:00

2pBAa1. Ultrasound and microbubble-mediated blood-brain barrier disruption: Pre-clinical best practices. Nicholas Ellens (Acertara Acoust. Labs, 1950 Lefthand Creek Ln., Longmont, CO 80501, nellens@acertaralabs.com)

The combination of trans-cranial ultrasound and microbubbles has been shown to open the blood-brain barrier safely and reversibly. The potential for neurological therapies and functional imaging with this approach is enormous. The future success of this procedure requires robust pre-clinical research and carefully conducted clinical studies that lower the cost of the therapy without compromising patient safety. This talk focuses on the hardware necessary for successful pre-clinical studies in mice and rats. It will explore the pros and cons of off-the-shelf, turnkey systems; custom lab-developed systems; and repurposed clinical systems. It will discuss best practices for animal handling. Toward clinical translation, the talk will conclude with an overview of some of the regulatory hurdles faced by those trying to move these therapies into patients. With this, it will provide recommendations for things to consider in upstream experimental design.

1:25

2pBAa2. Optimal control of the nonspherical oscillations of encapsulated microbubbles. Fathia F. Arifi (Dept. of Mech. and Aerosp. Eng., Univ. of Colorado, Colorado Springs, Colorado Springs, CO) and Michael L. Calvisi (Dept. of Mech. and Aerosp. Eng., Univ. of Colorado, Colorado Springs, 1420 Austin Bluffs Parkway, Colorado Springs, CO 80918, mcalvisi@uccs.edu)

Encapsulated microbubbles (EMBs) were originally developed as contrast agents for ultrasound imaging but are more recently emerging as vehicles for intravenous drug and gene delivery. Ultrasound can excite nonspherical oscillations, or shape modes, that can enhance the acoustic signature of an EMB and also incite rupture, which promotes drug and gene delivery at targeted sites. Therefore, the ability to control shape modes can improve the efficacy of both the diagnosis and treatment mediated by EMBs. This work uses optimal control theory to determine the ultrasound input that maximizes a desired nonspherical EMB response (e.g., to enhance scattering or rupture), while minimizing the total acoustic input in order to enhance patient safety and reduce unwanted side effects. The optimal control problem is applied to a model of an EMB that accounts for small amplitude shape deformations. This model is solved subject to a cost function that maximizes the incidence of rupture or acoustic echo while minimizing the acoustic energy input. The optimal control problem is solved numerically through pseudospectral collocation methods using commercial optimization software. Single frequency and broadband acoustic forcing schemes are explored and compared. The results show that broadband forcing significantly reduces the acoustic effort required to incite EMB rupture relative to single frequency schemes. Furthermore, the acoustic effort required depends strongly on the shape mode that is forced to become unstable.

Contributed Papers

1:50

2pBAa3. Effect of microbubble and acoustic radiation force parameters on $\alpha_v\beta_3$–microbubble targeting. Jair I. Castillo (Biomedical Eng., Univ. of Colorado Boulder, 1111 Eng. Dr. UCB 427, ECME 245, Boulder, CO 80309, jair.castillo@colorado.edu), J. A. Navarro-Becerra, and Mark Borden (Mech. Eng., Univ. of Colorado Boulder, Boulder, CO)

Ultrasound molecular imaging (USMI) employs targeted microbubbles (MBs) decorated with specific ligands to enhance the local visualization of vascular structures. It has been demonstrated that cyclo Arg-Gly-Asp (cRGD) targeted MBs in combination with acoustic radiation force (ARF) can bind to $\alpha_v\beta_3$ integrins providing enhanced images of the tumor vasculature. Therefore, the goal of this work was to study and elucidate the role of ARF and MB parameters on the targeting of cRGD-MBs using an *in vitro* $\alpha_v\beta_3$-coated microvessel phantom. Labeled polydisperse and 2-μm size- isolated MBs decorated with cRGD with buried-ligand architecture (BLA) were pumped at 1 ml/h in an $\alpha_v\beta_3$-coated microvessel (D = 200 μm) and pushed using ARF. At higher monodisperse MB concentrations, the specific attachment follows a linear behavior independent of the frequency applied. However, at lower MB concentrations, the attachment becomes linearly dependent on the frequency. Finally, attachment of monodisperse population of RGD-MBs increased 8- and 3- fold than polydisperse MBs. In conclusion, we elucidated that the use of a monodisperse population of MBs improves the molecular specificity for $\alpha_v\beta_3$ integrins.

2:05

2pBAa4. Inactivation of gram positive or gram negative microbes using different histotripsy regimes. Pratik Ambekar (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, pambek@uw.edu), Yak-Nam Wang (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Tatiana D. Khokhlova (Dept. of Medicine, Univ. of Washington, Seattle, WA), Daniel Leotta, Ekaterina Kuznetsova, Matthew Bruce, and Thomas Matula (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Abscesses are infected walled-off collections of pus and bacteria. They can affect any part of the body. Current treatment is typically limited to antibiotics, catheter drainage and hospitalization, or surgical wash-out. Although bacteria can develop drug resistance, they remain susceptible to thermal and mechanical damage. Histotripsy generates localized cavitation without heating and represents a potential new noninvasive treatment modality. Successful histotripsy treatments were demonstrated with gram-negative *E-coli* in both *in vitro* and *in vivo* studies [Brayman *et al*., UMB 44, 1996–2008 (2018); Matula *et al*., UMB 47, 603–619 (2020)]. Interestingly, preliminary data showed that boiling histotripsy was less effective than cavitation histotripsy in killing bacteria. In this talk, we will describe more in-depth studies that compare cavitation histotripsy (low number of cycles, high-pulse repetition rate) with boiling histotripsy (high number of cycles, low-pulse repetition rate) on inactivation rates of microbes. [Work funded in part by NIH NIBIB No. R01EB019365.]

2:20

2pBAa5. Ultrasound contrast agents: From buckling dynamics to swimming. Georges Chabouh (LiPhy, CNRS, 140 rue de la physique, Grenoble 38000, France, georges.chabouh@univ-grenoble-alpes.fr), Catherine Quilliet, Benjamin Dollet, Philippe Marmottant, and Gwennou Coupier (LiPhy, CNRS, Grenoble, France)

Cleverly engineered microswimmers have been of increasing scientific interest, as they show great promise in various biomedical applications. In this study, we propose a novel mechanism of propulsion in fluids at the microscale, using a buckling mechanism activated by pressure waves. We considered an *in vivo*-friendly hollow elastic shell of micrometric size composed of a lipidic membrane enclosing a gas bubble. Such microshells are approved for clinical use as diagnostic ultrasound contrast agents (UCAs). We experimentally investigate the buckling dynamics of microshells upon an increase of external pressure. The effect of the driving frequency is studied as well as that of the size and mechanical properties of the microshells. We evidence a non-zero displacement upon a complete cycle of deflation and re-inflation of the microshells, which includes buckling events. The proposed propulsion mechanism whose direction is controlled in the shell reference frame can be an answer to the problem of directivity accounted in the acoustic radiation force technique used in ultrasound molecular imaging and drug delivery.

2:35

2pBAa6. Instantaneous phase difference of nonlinear coupled bubbles from nonstationary forcing. Rintaro Hayashi (Mech. Eng., Univ. of Hawaii at Manoa, 95-030 Waihonu St., Mililani, HI 96789, rintaro@hawaii.edu) and John S. Allen (Mech. Eng., Univ. of Hawaii Manoa, Honolulu, HI)

The attraction and repulsion regimes of acoustically forced bubbles follow from their phase coupling. The direct evaluation of phase difference has received less attention than the secondary Bjerknes force, though it has important implications for nonlinear, nonstationary forcing regimes. We numerically investigate the response of two nonlinearly coupled bubbles undergoing radial oscillations and translation from chirp forcing. The corresponding scattered acoustic responses are analyzed using the empirical mode decomposition (EMD) and Hilbert transform in terms of respective instantaneous amplitude, frequency, and phase. This allows for determination of the phase differences from the coupling between the two bubbles. The phase coupling results agree with previous analytical theory in the linear acoustic limit. Attraction and repulsion regimes for nonstationary forcing can be classified with respect to the instantaneous phase differences unlike the time averaged secondary Bjerknes force. In the nonlinear regime, the chirp direction (forward, backward) results in significantly different radial and translational responses. This result can be explained with respect to influence of the softening jump discontinuity (backbone curve) on the phase coupling. Notably a backward chirp results in overall greater amplitude response and thus more translation during attraction. Implications for optimal waveforms are highlighted for potential applications.

2:50–3:05 Break

3:05

2pBAa7. A novel ultrasound-assisted laser technique to remove atherosclerotic plaques. Rohit Singh (Dept. of Mech. Eng., Univ. of Kansas, 1530 W15th St., Learned Hall, Rm. 5109, Lawrence, KS 66045, rohitsingh@ku.edu) and Xinmai Yang (Univ. of Kansas, Lawrence, KS)

Atherosclerosis is a medical condition in which arteries hardens and narrows due to the buildup of fat, cholesterol, calcium, and other substance. It can affect arteries of heart, brain, arms, legs, pelvis, and kidney, resulting in ischemic heart disease, carotid artery disease, peripheral artery disease, and chronic kidney disease. Laser based treatment techniques like laser angioplasty and carotid artery laser endarterectomy can be used to treat most common atherosclerosis disease like ischemic heart disease and carotid artery disease. Excimer laser coronary angioplasty (ELCA) is a clinical technique to remove the arterial plaque. However, the use of ELCA remains low due to higher risk of complication and its low efficacy in removing plaques as compared with balloon angioplasty. In this study, we developed a novel technology that uses ultrasound along with laser to assist the removal of atherosclerotic plaque. Spatio-temporally synchronized pulsed laser and ultrasound was used. The preliminary experiments were performed on pig belly fat sample to find the optimal ultrasound and laser parameters. The final experiment was performed on the plaque samples collected from patient during carotid endarterectomy procedure. Our results show that the addition of ultrasound can effectively reduce the needed laser power, which could result in improved treatment safety and efficiency.

3:20

2pBAa8. Optical breakdown in holmium:YAG laser lithotripsy. Yuri A. Pishchalnikov (Applaud Medical, Inc., 953 Indiana St., San Francisco, CA 94107, yuri.pishchalnikov@applaudmedical.com), William M. Behnke-Parks (Applaud Medical, Inc., San Francisco, CA), and Marshall L. Stoller (Dept. of Urology, Univ. of California San Francisco, San Francisco, CA)

Optical breakdown produces shock waves and cavitation and is typically observed with the Q-switched lasers. In contrast, the free-running mode holmium:YAG laser lithotripters produce long (>150 μs) laser pulses that were reported to break urinary stones via thermal decomposition of stones and explosive vaporization of interstitial water without optical breakdown. Here, we show evidence of optical breakdown in holmium:YAG laser lithotripsy using fine temporal and spatial resolution of ultrahigh-speed video microscopy. Images were collected with a Shimadzu HPV-X2 camera at rate up to 10 million frames per second. Infrared- and visible-light photodetectors (10-ns rise time) resolved temporal profiles of infrared laser pulses and visible sparks indicative of optical breakdown. Hydrophone measurements showed shock waves produced by optical breakdown and subsequent collapses of vapor bubbles, capable of breaking glass microscope slides with a single laser pulse. Optical breakdown was observed with the laser fiber tip in contact with synthetic and whole surgically retrieved urinary stones. These observations suggest that the previously unappreciated optical breakdown with holmium:YAG lasers can be an additional acoustic mechanism of action in laser lithotripsy, currently the most used treatment modality for urinary stones, and in other laser procedures. [Work supported by NIDDK of NIH under Award No. R43DK129104.]

2p TUE. PM

3:35

2pBAa9. Application of Koopman operator theory to the control of nonlinear bubble dynamics. Andrew J. Gibson (Dept. of Mech. and Aerosp. Eng., Univ. of Colorado, Colorado Springs, 1420 Austin Bluffs Parkway, Colorado Springs, CO 80918, agibson7@uccs.edu), Xin C. Yee, and Michael L. Calvisi (Dept. of Mech. and Aerosp. Eng., Univ. of Colorado, Colorado Springs, Colorado Springs, CO)

Microbubbles are widely used in biomedicine for ultrasound contrast imaging and are a promising vehicle for targeted drug and gene delivery. The ability to control the oscillations of bubbles through the applied ultrasound can improve the effectiveness of these treatments, for example, by enhancing the acoustic echo or inciting bubble rupture at precise locations. Koopman operator theory has gained interest in the past decade as a framework for rigorously transforming nonlinear dynamics on the state space into linear dynamics on abstract function spaces, which preserves the underlying nonlinear dynamics of the system. These spaces can be approximated purely through machine learning and data-driven methodologies, which then enables the application of classical linear control strategies to strongly nonlinear systems. Here, we use a Koopman linear quadratic regulator (LQR) to control the nonlinear dynamics of spherical bubbles, as described by the well-known Rayleigh-Plesset equation, with two novel objectives: (1) stabilization of the bubble at a nonequilibrium radius and (2) simple harmonic oscillation at amplitudes large enough to incite nonlinearities. Control is implemented through both broadband and single-frequency transducers that are modulated by the Koopman LQR controller. We repeat these results using Koopman model predictive control (MPC), which allows for the implementation of constraints. This work is a step towards controlling nonspherical shape modes of encapsulated microbubbles.

3:50

2pBAa10. Analysis of cavitation induced stresses on blood vessel wall during photo-mediated ultrasound therapy using finite-element based numerical models. Rohit Singh (Dept. of Mech. Eng., Univ. of Kansas, 1530 W15th St., Learned Hall, Rm. 5109, Lawrence, KS 66045, rohitsingh@ku.edu) and Xinmai Yang (Univ. of Kansas, Lawrence, KS)

Photo-mediated ultrasound therapy (PUT) is a novel technique, which utilizes spatio-temporally synchronized pulsed laser and ultrasound to produce enhanced cavitation inside blood vessels. Vascular bio-effects during PUT, such as reduced perfusion rate and rupture of blood vessel, are thought to be due to shear and circumferential stresses induced on vessel wall. We have developed a 2D axisymmetric and a 3D finite-element method (FEM)-based models in COMSOL Multiphysics to calculate the stresses induced on vessel wall during PUT. The 2D axisymmetric model simulation showed that drastic increase in bubble size with PUT application resulted in higher shear and circumferential stresses as compared with ultrasound application only in the beginning phase of PUT. The 3D model simulation showed increase in stresses when a bubble moves towards the vessel wall, and found relatively higher increase in shear stress as compared to circumferential stress. The 2D results were imperative in understanding of PUT inception resulting in stronger stresses on vessel walls, which were otherwise almost zero for ultrasound-only. The 3D results demonstrated that the stresses on vessel walls further increased during the subsequent phase of PUT, enough to induce vascular bio-effects.

4:05

2pBAa11. Investigating the resonance response of a system of two ultrasound-driven lipid encapsulated microbubbles. Hossein Yusefi (Phys., Concordia Univ., Montreal, QC, Canada) and Brandon Helfield (Phys., Concordia Univ., 7141 Sherbrooke St. West, L-SP 365.04, Montreal, QC H4B 1R6, Canada, brandon.helfield@concordia.ca)

Ultrasound-stimulated microbubbles are clinical imaging agents and being developed for therapeutic applications. Here, we aim to understand the behaviour of two individual phospholipid-encapsulated microbubbles in close proximity to each other, typically the case given clinical doses. We developed a finite element model to study the radial resonance response of each microbubble within a two-microbubble system from 1–8 MHz with bubble diameters ranging from 2 to 4 μm, bubble center-to-center distances h = 8–24 μm, and peak-negative pressures of 30–45 kPa. For two identical microbubbles, our results show the frequency of maximum response (f_{MR}) decreases (7%–10%) and the amplitude of maximum response (A_{MR}) increases (9%–11%) as the microbubbles approach one another. For a two-bubble system of different microbubble sizes, the larger microbubble shows no change in f_{MR} and a slight shift of A_{MR} (2–3%). However, the smaller microbubble exhibits an increase in f_{MR} (7–11%) and a significant decrease of A_{MR} (38–52%). Furthermore, when in very close proximity (h = 8 μm), smaller bubbles exhibit a secondary resonance peak corresponding to the f_{MR} of the larger bubble with amplitudes comparable to its primary resonance peak. Our work suggests that microbubble resonance behaviour is greatly affected by the presence of nearby bubbles, which has implications in imaging and therapy.

4:20

2pBAa12. Combining ultrasound and endovascular laser for thrombolysis. Rohit Singh (Dept. of Mech. Eng., Univ. of Kansas, 1530 W15th St., Learned Hall, Rm. 5109, Lawrence, KS 66045, rohitsingh@ku.edu), Janggun Jo (Biomedical Eng., Univ. of Michigan, Ann Arbor, MI), and Xinmai Yang (Univ. of Kansas, Lawrence, KS)

Blood clot formation in vein, known as venous thromboembolism is a major disease worldwide. Ultrasound and laser-based treatment techniques have been developed to remove blood clots and recanalize occluded vessels. The applications of both technologies are hindered by their individual limitations. We tested a novel hybrid technology based on the spatio-temporally synchronized pulsed laser and ultrasound, namely, ultrasound-assisted endovascular laser thrombolysis (USELT) in an *in-vitro-* blood clot model and *in vivo* rabbit model. The *in vitro-* study demonstrated enhanced thrombolysis with USELT as compared with ultrasound-only and laser-only. In *in vivo* study, out of seven treated rabbits in USELT group, the vein was fully or partially recanalized in five rabbits and poorly recanalized in two rabbits, whereas the vein treated with ultrasound-only and laser-only did not result in recanalization in any case. The *in vitro-* and *in vivo* study successfully demonstrated the feasibility of USELT for thrombolysis, and USELT has the potential to combine the advantages of both ultrasound and laser thrombolysis techniques and overcome their limitations.

Session 2pBAb

Biomedical Acoustics: New Developments in Lung Ultrasound II

Libertario Demi, Cochair
Information Engineering and Computer Science, University of Trento, Via sommarive 9, Trento, 38123, Italy

Marie Muller, Cochair
MAE, North Carolina State University, 911 Oval Drive, Engineering Building III, Raleigh, NC 27606

Invited Papers

1:00

2pBAb1. Big things come in small packages—Clinical use of endobronchial ultrasound. A. C. Burks (Medicine, Univ. of North Carolina at Chapel Hill, 160 Dental Circle, Campus Box 7219, Chapel Hill, NC 27599, acole_burks@med.unc.edu)

Endobronchial ultrasound (EBUS) revolutionized bronchoscopy. It first became available for clinical use in 1999 as a mechanical radial probe and was initially used to define airway wall anatomy, but was supplanted by convex probe EBUS bronchoscopes in 2002–now the preferred initial approach for evaluation of mediastinal pathology. Radial EBUS has since proven invaluable for evaluating the peripheral lung and aiding biopsy guidance despite the limitations in ultrasound (US) propagation through air filled structures. Centrally, airways pass through the mediastinum in direct contact with the esophagus, heart, great vessels, and lymph nodes, making the mediastinum amenable to direct US imaging. However, beyond the lung hila, the airways are largely surrounded by air, generally preventing production of usable US images by halting penetration and reverberation of sound waves. The tissue mass that makes up the terminal airway walls, however, does propagate ultrasound waves before encountering reflective air. The ratio of tissue to air varies according to anatomic location and physiologic/disease state; identifying patterns in the varying ratio of tissue to air within the lungs allows the ability to distinguish normal from abnormal lung. This presentation will describe the clinical use of EBUS in modern bronchoscopy.

1:30

2pBAb2. Recent developments in fetal lung ultrasound. Sleiman R. Ghorayeb (Radiology and Molecular Medicine, Hofstra Northwell School of Medicine, 133 Hofstra University, Ultrasound Res. Lab, 207 Weed Hall, Hempstead, NY 11549, Sleiman.R.Ghorayeb@hofstra.edu), Matthew J. Blitz, and Luis A. Bracero (Maternal and Fetal Medicine, Northwell Health at Southside Univ. Hospital, Bayshore, NY)

Fetal ultrasound has been employed for years to help evaluate fetal growth and development and to monitor pregnancy. Furthermore, fetal lung maturity has been assessed pre-natally by examination of the amniotic fluid, usually obtained by transabdominal amniocentesis, for lecithin, lecithin/sphingomyelin ratio, or "P" factor (fluorescent polarization measurement for lipids.) Quantitative ultrasound texture analysis of fetal lung has been proposed in previous studies as a promising noninvasive method to predict fetal lung maturity, fetal lung hypoplasia, and neonatal respiratory morbidity. This technique utilizes standard fetal lung images, which are easily obtained by sonographers during routine ultrasound examinations. Additional information then can be extracted from these images by applying quantitative processing methods that characterize the tissue. This presentation discusses how to differentiate preterm ($<$37 weeks of gestation) from term ($\geq$37 weeks of gestation) fetal lungs by quantitative texture analysis of ultrasound images. This quantification is based on the extent of heterogeneity associated with lung maturity by employing a unique, novel noninvasive technique to determine the Heterogeneity Index (HI) in ultrasound images. The HI values are then compared in sonograms of immature lungs with mature lungs. Conceptual advances for the possibility of integrating this technology in handheld ultraportable systems for point-of-care ultrasound (POCUS) are also presented.

2:00

2pBAb3. Utility of lung ultrasound in following up COVID-19 patients over a 3-month period. Tiziano Perrone (Emergency Dept., Humanitas Gavazzeni Bergamo, Italy, Bergamo, Italy, tiziano.perrone@gavazzeni.it), Giulia Gori, Federica Lepore, umberto sabatini, Maria Laura Robone (Internal Medicine, IRCCS Fondazione Policlinico San Matteo di Pavia, Pavia, Italy), Elena Torri (Emergency Dept., Humanitas Gavazzeni Bergamo, Italy, Bergamo, Italy), and Antonio Di Sabatino (Internal Medicine, IRCCS Fondazione Policlinico San Matteo di Pavia, Pavia, Italy)

Lung ultrasound (LUS) examination has been shown to have a potential diagnostic and prognostic role in SARS-CoV-2 pneumonia disease. We evaluated the role of a new LUS score protocol (14 windows evaluation, graded score 0–3) in patients with SARS-CoV-2 pneumonia and the association of LUS patterns with clinical findings in acute stage and after three month from disease recovery. First, a cohort of 52 consecutive laboratory-confirmed SARS-CoV-2 patients underwent LUS examination upon the admission in an Internal Medicine ward. A total LUS score as the sum of the scores at each explored area was computed,and we investigated the association between LUS score and the clinical worsening. Then 47 patients who survived the first COVID-19 wave and who underwent a 3-stage

LUS examination (T0 “access to ER”; T1 “ward hospitalization”; T2 “post-COVID outpatient”) were enrolled for the longitudinal study. In the acute stage, we observed that a median LUS score above 24 was associated with an almost 6-fold increase in the odds of worsening. In the longitudinal observation, we seen that LUS score’s variation between T0 and T2 resulted to be statistically significant, as well a difference of LUS score between patients with or without pleural effusion, maintained over time.

2:30

2pBAb4. A standardised and evidenced-based lung ultrasound acquisition protocol and scoring system for the monitoring and stratification of COVID-19 and post-COVID-19 patients. Libertario Demi (Dept. of Information Eng. and Comput. Sci., Univ. of Trento, Via sommarive 9, Trento, Italia 38123, Italy, libertario.demi@unitn.it), Federico Mento (Dept. of Information Eng. and Comput. Sci., Univ. of Trento, Povo, Trento, Trento (TN), Italy), Antonio Di Sabatino, Anita Fiengo, umberto sabatini (IRCCS San Matteo Polyclinic Foundation, Pavia, Italy), Veronica N. Macioce (UOS Pneumologia di Codogno, Lodi, Italy), Marco Robol (Dept. of Information Eng. and Comput. Sci., Univ. of Trento, Trento, Italy), Francesco Tursi (UOS Pneumologia di Codogno, Lodi, Italy), Carmelo Sofia, Chiara di Cienzo, Andrea Smargiassi (Pulmonary Medicine Unit, Dept. of Med. and Surgical Sci., Fondazione Policlinico Universitario Agostino Gemelli IRCCS, Rome, Italy), Riccardo Inchingolo (Pulmonary Medicine Unit, Dept. of Medical and Surgical Sci., Fondazione Policlinico Universitario Agostino Gemelli IRCCS, Gemelli, Italy), and Tiziano Perrone (Emergency Dept., Humanitas Gavazzeni, Bergamo, Italy)

Lung ultrasound (LUS) is currently utilized worldwide to assess COVID-19 patients. However, imaging protocols are often defined arbitrarily, and studies on post-COVID-19 are lacking. In this work, we report on the capabilities of standardized LUS to monitor and stratify COVID-19 and post-COVID-19 patients. A validated and standardized imaging protocol based on 14 scanning-areas and a 4-level scoring system were utilized to collect and analyze data from 220 patients, 100 COVID-19 positive, and 120 post-COVID-19. Next, the capability of five imaging protocols (based on 4, 8, 10, 12, and 14 scanning-areas) to intercept the most significant LUS findings was compared. Moreover, a longitudinal-study was conducted aiming at investigating the possibility to simplify the protocol during follow-up. Results on the agreement between AI-models and LUS experts with respect to LUS data evaluation are also reported. In conclusion, a 12-areas protocol emerges as the optimal trade-off between a time-efficient and an accurate LUS examination. However, it appears not to be possible to reduce further the number of scanning-areas during follow-up. Finally, COVID-19 and post-COVID-19 data seem to show differences capable to confuse AI models that were not trained on post-COVID-19 data, supporting the hypothesis of the existence of LUS patterns specific to post-COVID-19.

3:00–3:15 Break

Contributed Papers

3:15

2pBAb5. On the impact of pixel resolution on automated scoring of lung ultrasound images from coronavirus disease 2019 patients. Umair Khan (Dept. of Information Eng. and Comput. Sci., Univ. of Trento, Via Sommarive, 9, Trento, Trento 38123, Italy, umair.khan@unitn.it), Federico Mento (Dept. of Information Eng. and Comput. Sci., Univ. of Trento, Povo, Trento, Trento (TN), Italy), Lucrezia N. Giacomaz, Riccardo Trevisan (Dept. of Information Eng. and Comput. Sci., Univ. of Trento, Trento, Trento, Italy), Andrea Smargiassi (Fondazione Policlinico Universitario A. Gemelli IRCCS, Gemelli, Italy), Riccardo Inchingolo (Fondazione Policlinico Universitario A. Gemelli IRCCS, Rome, Rome, Italy), Tiziano Perrone (Emergency Dept., Humanitas Gavazzeni, Pavia, Italy), and Libertario Demi (Dept. of Information Eng. and Comput. Sci., Univ. of Trento, Trento, Italia, Italy)

With the outbreak of the COVID-19 pandemic, remote diagnosis, patient monitoring, collection, and transmission of health data from electronic devices are rapidly taking its share in the health sector. These devices are, however, limited on resources like energy, memory, and processing power. Consequently, it is highly relevant to investigate how to minimize the size of data, keeping intact the information content. The objective of this study is to, thus, observe the impact of pixel resolution on the automated scoring by DL algorithms for LUS videos. First, 448 videos from 20 patients were normalized to a common pixel resolution, i.e., the largest found over the dataset (841 pixels/cm^2). Next, the pixel resolution was further reduced by factor 2 by resampling the normalized data to 210 pixels/cm^2. Original, normalized and re-sampled videos were evaluated using the DL algorithm [Roy *et al*., IEEE Trans. Med. Imaging 39, 2676–2687 (2020)]. At frame level, for normalized and resampled videos, the level of agreement of DL results with the original videos was 93.2% and 86.6%, respectively. Similar performance was found at video level with the agreement to 95.75% and 85.93%, respectively. The study showed that with a significant reduction in the pixel resolution of LUS data, low variation in the DL performance is observed.

3:30

2pBAb6. Neuro-symbolic interpretable AI for automatic COVID-19 patient-stratification based on standardised lung ultrasound data. Leonardo Lucio Custode (Dept. of Information Eng. and Comput. Sci., Univ. of Trento, Via sommarive 9, Trento, TN 38123, Italy, leonardo.custode@unitn.it), Federico Mento (Dept. of Information Eng. and Comput. Sci., Univ. of Trento, Povo, Trento, Trento (TN), Italy), Sajjad Afrakhteh (Dept. of Information Eng. and Comput. Sci., Univ. of Trento, Trento, TN, Italy), Francesco Tursi (UOS Pneumologia di Codogno, ASST Lodi, Lodi, Italy), Andrea Smargiassi (Pulmonary Medicine Unit, Dept. of Med. and Surgical Sci., Fondazione Policlinico Universitario Agostino Gemelli IRCCS, Rome, Italy), Riccardo Inchingolo (Pulmonary Medicine Unit, Dept. of Med. and Surgical Sci., Fondazione Policlinico Universitario Agostino Gemelli IRCCS, Gemelli, Italy), Tiziano Perrone (Emergency Dept., Humanitas Gavazzeni, Bergamo, Italy), Libertario Demi (Dept. of Information Eng. and Comput. Sci., Univ. of Trento, Trento, Italia, Italy), and Giovanni Iacca (Dept. of Information Eng. and Comput. Sci., Univ. of Trento, Trento, Italy)

In the current pandemic, being able to efficiently stratify patients depending on their probability to develop a severe form of COVID-19 can improve the outcome of treatments and optimize the use of the available resources. To this end, recent studies proposed to use deep-networks to perform automatic stratification of COVID-19 patients based on lung ultrasound (LUS) data. In this work, we present a novel neuro-symbolic approach able to provide video-level predictions by aggregating results from frame-level analysis made by deep-networks. Specifically, a decision tree was trained, which provides direct access to the decision process and a high-level explainability. This approach was tested on 1808 LUS videos acquired from 100 patients diagnosed as COVID-19 positive by a RT-PCR swab test. Each video was scored by LUS experts according to a 4-level scoring system specifically developed for COVID-19. This information was utilised for both the training and testing of the algorithms. A five-folds cross-validation process was utilised to assess the performance of the

presented approach and compare it with results achieved by deep-learning models alone. Results show that this novel approach achieves better performance (82% of mean prognostic agreement) than a threshold-based ensemble of deep-learning models (78% of mean prognostic agreement).

3:45

2pBAb7. Ultrasound at the pulmonary air-tissue interface: Mechanism of haemorrhage induction and implications for safe use of diagnostic and therapeutic lung ultrasound applications. Frank Wolfram (Clinic of Thoracic and Vascular Surgery, SRH Wald-Clinic Gera, Strasse des Friedens 122, Gera 07548, Germany, frankwolfram@gmx.de), Holger Gutsche, and Thomas G. Lesser (Clinic of Thoracic and Vascular Surgery, SRH Wald-Clinic Gera, Gera, Germany)

Despite the high-diagnostic value of sonographic lung imaging, providing substantial patient benefit, clinician should be aware that ultrasound exposure on lung is not absolutely without risk of harm. As shown in animal models, lung ultrasound (LUS) even in the diagnostic regime can induce pulmonary capillary haemorrhage (PCH) at the alveolar epithelium-gas interface. Acoustic peak negative pressure in the range of 1.0–1.5 MPa correlates well with PCH threshold. However, in retrospective human studies, adverse events could not be demonstrated following LUS examination, which is evidence of symptomatic PCH after therapeutic ultrasound ablation in proximity of lung exists. Therefore, this subject remains controversial. This work summarizes the underlying physical causes and consequences of ultrasound induced PCH. The complexity of PCH thresholds depending on imaging modes, pathological lung conditions including ventilation parameters will be presented. In addition, current recommendations for the safe use of LUS based on Safety Indices given by AIUM and EFSUMB guidelines and the impact of LUS specific settings, available on modern scanners, will be discussed.

4:00

2pBAb8. Mechanistic analysis of B-line formation in lung ultrasound. Gilles P. Thomas (Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, gillespierre.thomas@gmail.com), Oleg A. Sapozhnikov (Univ. of Washington, Seattle, WA), Adam Maxwell (Urology, Univ. of Washington, Seattle, WA), Jeff Thiel, Bryan W. Cunitz (Univ. of Washington, Seattle, WA), Michael R. Bailey (Ctr. for Industrial and Med. Ultrasound, Appl. Phys. Lab, Univ. of Washington, Seattle, WA), Kyle Steinbock, Layla Anderson, Ross Kessler, Adeyinka Adedipe (Univ. of Washington, Seattle, WA), and Tatiana D. Khokhlova (Dept. of Medicine, Univ. of Washington, Seattle, WA)

The number and distribution of lung ultrasound (LUS) imaging artifacts—B-lines—is correlated with the presence of lung interstitial syndrome such as viral infection and pulmonary edema. The detection and quantification of B-lines is machine and operator dependent, and the mechanisms for B-line formation are not fully understood. The goals of this work were to collect RF data during LUS exams in patients with pulmonary edema and to compare to the signals and corresponding images simulated numerically to elucidate the B-line formation process. Verasonics ultrasound engine (VUE) with a phased array probe (4.5 MHz) was used to perform standard 10-zone LUS in ten patients with confirmed pulmonary edema. The RF data corresponding to each B-mode image and a series of 35 plane wave acquisitions were collected for off-line analyses. Finite element modeling of LUS pulse propagation was performed in COMSOL with fluid-filled inclusions of variable sizes and shapes in air-like medium simulating edematous lung areas. The B-mode image resulting from the simulated RF signals was reconstructed using VUE beamforming. The B-lines similar to those observed experimentally formed in simulated images only under specific conditions of sizes and shapes of water-filled areas relatively to LUS imaging wavelength. [Work supported by NIH R01EB023910.]

4:15–4:30 Break

4:30–5:00
Panel Discussion

Session 2pED

Education in Acoustics, Engineering Acoustics, and Architectural Acoustics: Connecting Industry and Education

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

Jim DeGrandis, Cochair
Acoustics First Corporation, 2247 Tomlyn Street, Richmond, VA 23230

Chair's Introduction—1:00

Invited Papers

1:05

2pED1. Building industrial-education partnerships in acoustics. Andrew Barnard (Acoust., Penn State, Penn State University, State College, PA 16801, barnard@psu.edu)

Strong industry relations can provide many positive impacts for an educational program. The rewards of such an educational-industrial partnership only occur after the hard work of trust-building between the educational provider and the industry partners. A strong partnership requires value generated for both the industrial partner and the educational institution. In this talk, examples will be given of successful and unsuccessful industrial partnerships with academia. The focus will be on educational partnerships, as opposed to industrial research funding. Examples of undergraduate class projects, guest lecturing, professional seminars and webinars, and others will be discussed, in addition to the value streams realized by long time industrial partners.

1:25

2pED2. A Capstone Acoustical Engineering Design course with industry-sponsored projects. Christopher M. Jasinski (Mech., Aerosp., and Acoust. Eng., Univ. of Hartford, 200 Bloomfield Ave., West Hartford, CT 06117, jasinski@hartford.edu) and Robert Celmer (Mech., Aerosp., and Acoust. Eng., Univ. of Hartford, West Hartford, CT)

One of the transitions that engineering students must make as they enter the work force is the progression from theoretical concepts to applied/real world applications. At the University of Hartford, engineering design courses serve as culminating experiences devised to bridge this passage. One such course, Acoustics Capstone Design, challenges the student to apply the past three years' conceptual base of two acoustics and two vibration courses to a problem-solving opportunity replete with actual scenarios encountered in industry. Each year local firms approach our Engineering Applications Center for assistance with a variety of sound or vibration problems. After an initial training period, students make use of the laboratory's FFT/real time analyzers, anechoic/reverberation chambers, sound intensity and modal analysis software, acoustic modeling software, and vibration shaker/transducers. Using a consultant-client model, students work collaboratively in teams of two defining the problem, developing a method of approach, making appropriate measurements, devising alternate solutions, and ultimately delivering a written and oral presentation at the end of the semester. The arrangement regularly results in employment offers for the graduating seniors. The paper discusses specific projects and some experiences students have had with their first industrial assignment, as well as creative means of equipment acquisition.

Contributed Papers

1:45

2pED3. Enhancing graduate, undergraduate, and vocational school education with research and development internships and capstone projects. David A. Brown (Elec. Eng., Univ. of Massachusetts, Dartmouth, 151 Martine St., Fall River, MA 02723, dbAcousticsDB@gmail.com)

The Center for Innovation and Entrepreneurship (CIE) and Advanced Technology and Manufacturing Center (ATMC) of UMass Dartmouth provides a facility for collaboration with industry. Local businesses provide internships, research opportunities, and specialized training in acoustics that enhance student growth and practical learning. The College of Engineering benefits from the industrial sponsorship for undergraduate capstone projects as well as graduate research. Students benefit from practical training and earning money for tuition and potential post-graduation employment. This tradition has grown over decades with the maturation of facilities for advancement in engineering and underwater acoustics. This presentation will present some student focused success stories within the area of engineering acoustics and transduction.

2:00

2pED4. Underwater acoustic spiral wave navigation system. Isabella Di Bona (Elec. and Comput. Eng., Univ. of Massachusetts Dartmouth, 151 Martine St., Ste. 123, Fall River, MA 02723, idibona@umassd.edu), Christopher Gravelle, Zakaria Faddi (Elec. and Comput. Eng., Univ. of Massachusetts Dartmouth, Dartmouth, MA), David A. Brown (Elec. and Comput. Eng., Univ. of Massachusetts Dartmouth, Fall River, MA), and Corey Bachand (Electroacoustics, BTech Acoust. LLC, Fall River, MA)

A spiral wave navigation beacon is described that creates an outgoing signal with a wavefront that is a diverging spiral cylindrical wave followed by a reference signal of circular wavefront as previously described by Dzikowicz and Hefner in JASA. In this demonstration, the signals are created by driving a cylindrical transducer's orthogonal sine and cosine modes of vibration in phase quadrature as previously described by Brown and Bachand in JASA. The two dipolar radiation patterns combine to create a helically diverging spiral wavefront that may be used for navigation and communication. The electronic system design and transducer characterization will be presented and are the subject a senior-capstone design project in the College of Engineering at UMass Dartmouth for four of the authors (I.D.B., C.G., K.C., Z.F.). [The project provided a multidisciplinary opportunity in acoustical engineering, instrumentation, calibration, and signal processing and was sponsored by industry (BTech).]

2:15

2pED5. Ensuring the kids are alright: Ways to help students network with industry professionals in the age of virtual meetings and career fair disillusionment. Ryan L. Harne (The Penn State Univ., 2482 Raven Hollow Rd., State College, PA 16801, ryanharne@psu.edu)

The pandemic has derailed the traditional networking vehicles to connect aspiring, ambitious students with industry professionals. As a result, the flow of young talent to best-fitting industry outlets is being misdirected. Moreover, many students are not inclined to accept virtual meetings as a replacement for face-to-face engagement while professionals often prefer virtual meetings as means to maximize productivity. The latter disparity has led to campus "career fairs" turning into stay-at-home experiences that students notoriously lament. Without bridges to network students with the right industry professionals to optimize hiring and onboarding processes, our society faces increasing early career turnover, loss of productivity, and disenchantment of young talent with the technical outlets that could await them. This talk will first clarify the crisis facing industries as a result of pandemic-motivated physical distancing practices. We will discuss lessons learned from recent attempts to foster genuine connections between students and the industry professionals they could work alongside in the future. The talk will conclude with a call to action on part of both students and professionals to collectively rebuild the networking system before a temporary disruption turns into a generational failure.

Invited Paper

2:30

2pED6. Working with industry to create new educational pathways for students to diversify the acoustics workforce. Kimberly A. Riegel (Phys., Queensborough Community College, Bayside, NY 11361, kriegel@qcc.cuny.edu)

Many undergraduates are not aware of the opportunities that a career in acoustics provides. To meet the students where they are and provide more representation in the field, we need to create new pathways. One big hurdle for first time, low-income students is that they are working while attending classes. The STEM pipeline is changing, and students are taking different pathways as they navigate their academic careers in addition to their other responsibilities. More than 60% of students attend community college at some point in their careers, and more than 70% of students attend a public university with many of those colleges being minority serving institutions. I am proposing to create an acoustical technician track that would allow students to learn practical skills that would allow them to enter the workforce earlier. These degrees would need to be developed alongside industry to ensure that the students would be particularly prepared for the local needs of the industry. This will also financially allow students to continue their education if they choose.

2:50–3:15
Panel Discussion

Session 2pNS

Noise, Architectural Acoustics, and ASA Committee on Standards: Standards, Codes, and Criteria Applications in the Real World II

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

Derrick P. Knight, Cochair
Trane Technologies, 2313 20th Street South, La Crosse, WI 54601

Chair's Introduction—2:00

Invited Papers

2:05

2pNS1. Where does an airport end? Civic considerations of acoustics as practiced. Davyd H. Betchkal (Natural Sounds and Night Skies Div., National Park Service, PO Box 9, Denali Park, AK 99755, Davyd_Betchkal@nps.gov) and Scott McFarland (Natural Sounds and Night Skies Div., National Park Service, Fort Collins, CO)

In the United States, Federal agencies work on behalf of the public within domains defined relative to their purpose (i.e., mission). The mission of most agencies—and certainly land management agencies—includes a spatial manifestation. Legal authorities shape the practice of acoustics for use in decision-making, but often without reciprocal influence from scientists in the field. As a consequence, the propagation of sound complicates many legal constructions of space. In this talk, we consider a spatial contradiction that results in an avoidable problem: misuse of Day-Night Level (L_{dn}) to estimate residual sound level in natural areas. We resolve the contradiction by offering a standard alternative, ANSI/ASA S12.100-2014 "Methods to define and measure the residual sound in protected natural and quiet residential areas."

2:25

2pNS2. On the 50th anniversary of ASTM E33, a history of ASTM International standards on building and environmental acoustics. Noral D. Stewart (Stewart Acoust. Consultants, Raleigh, NC), Eric P. Wolfram (Riverbank Acoust. Labs., 1512 South Batavia Ave., Geneva, IL 60134, eric@riverbankacoustics.com), Richard J. Peppin (RION Co., Rockville, MD), and Robert Putnam (None, Winter Springs, FL)

This year marks the 50^{th} anniversary of ASTM International Committee E33 on Building and Environmental Acoustics which combined all ASTM activity in these areas into one committee, though work on ASTM standards for measurement of sound absorption and transmission had started prior to 1950. This presentation will discuss the history of the committee and ASTM International standards in acoustics including the leadership that has made it possible.

2:45

2pNS3. Revision of a common product noise measurement standard in response to user concerns. Robert D. Hellweg (Hellweg Acoust., 13 Pine Tree Rd., Wellesley, MA 02482, Hellweg@HellwegAcoustics.com) and Stephen J. Lind (LindAcoustics LLC, Onalaska, WI)

Many manufacturers of products and equipment use ISO 3744 to measure and report their sound power levels. (ISO 3744 is nationally adopted as an American standard—ANSI/ASA S12.54.) The products include almost all sources that are primarily stationary such as household appliances, computers, heavy machinery, and construction equipment. The standard is referenced in European regulations and hundreds of machine-specific standards. Many users of ISO 3744 have complained that it was too complicated, complex, and difficult to use. In response to these user concerns, the ISO working group responsible (ISO TC43/SC1/WG28) is revising ISO 3744 to make it simpler to use and less complex. Among the many changes, the draft standard includes the following: removing rarely used conditions and reducing the complexity of background noise calculations. The revision also removes certain aspects and places them into two new standards also under development: (1) qualification of measurement rooms ISO 26102-1 and (2) measurement uncertainty ISO 5114-1. Draft standards are being balloted in member countries of ISO TC43/SC1 (Noise). This paper will describe the provisions in the draft revision of each of these three noise standards and their current status.

3:05

2pNS4. Streamlining ASTM acoustical standards. Elliott Dick (North Orbit Acoust. Labs., 917 Rice St., St. Paul, MN 55117, Elliott@northorbit.com), Samantha Rawlings (Veneklasen Assoc., Santa Monica, CA), Benjamin M. Shafer (PABCO Gypsum, Tacoma, WA), and Matthew Golden (Pliteq, Woodbridge, ON, Canada)

ASTM E33, the Committee on Building and Environmental Acoustics, has decided to undertake a process to streamline and harmonize existing E33 standards. As the committee's standards have developed over the decades, inconsistencies, omissions, and contradictions have wound their way into the standards due to cross-references among standards and revisions. Members of this committee volunteered to spearhead an effort to restructure the standards with the goal of making the standards easier to understand, simpler to use, and to facilitate future revision efforts. This paper summarizes the beginnings of this effort and steps being taken at this time.

Contributed Papers

3:25

2pNS5. Experimental investigation of the variance of ground impedance measurments in coastal environments. Faith A. Cobb (Eng., East Carolina Univ., East Carolina University, Slay Hall 248, Greenville, NC 27858, cobbf18@students.ecu.edu), Andrea Vecchiotti, Diego Turo, Joseph Vignola (Mech. Eng., The Catholic Univ. of America, Washington, DC), and Teresa J. Ryan (Eng., East Carolina Univ., Greenville, NC)

The purpose of this work is to understand the results obtained with the ANSI/ASA S1.18 standard for outdoor acoustic surface impedance measurements when applied to surfaces with characteristics at or outside the recommended use case. This study is part of a larger effort working to develop a numerical model for long-range atmospheric acoustic transmission loss over coastal areas. Previous studies were done to characterize the relationship between moisture content and effective flow resistivity of sandy shores by conducting acoustic surface impedance measurements following the procedure specified in the ANSI/ASA S1.18. While the standard specifies surface variation of less than 5 cm and at least four measurements per location, this work seeks to understand the consequences of surface variation in natural terrains on the obtained results. These consequences were evaluated by measuring the effective flow resistivity of multiple outdoor surfaces such as uniform grass, raked sandy shores, and undisturbed sandy shores. At every location, 16 measurements were performed in grid pattern. Subsets of the 16 measurements are used to perform an analysis of the variance of the measured flow resistivities collected at locations with different surface characteristics.

3:40

2pNS6. The need for appliance sound ratings. Corey Taylor (Owens Corning, 2790 Columbus Rd., B75, Granville, OH 43023, Corey.taylor@owenscorning.com) and Kevin Herreman (Owens Corning, Granville, OH)

Home life has changed over the past 50 years with appliances, providing automated cycles to aid in storage, cleaning, and cooking. Appliances have proven to do tasks conveniently with more efficiency than people. Home designs have changed to adapt to the change in lifestyles. Locating the appliances close areas where they are needed is a highlight of home designs. Clothes washers, long relegated to the basement in older homes have found their way into the main living areas. The price of that convenience is the increase in background sound levels in typically quiet areas of the home. Consider that the sound emitted by a clothes washer in the spin cycle is of similar levels as a car passing a person standing on the side of the road or traffic noise. This level of sound can be disruptive in the home, especially if the appliance is in the main living space or near the bedrooms. In the early 2000s, a major appliance retailer implemented a sound level rating for dishwashers that provided data for customers to compare products. This paper will discuss the positive effects of that standard on dishwasher sound and discuss the need for additional appliance sound level ratings.

3:55

2pNS7. Improving upon standard approaches for mapping road traffic noise. Mylan R. Cook (Dept. of Phys. and Astronomy, Brigham Young Univ., N201 ESC, Provo, UT 84602, mylan.cook@gmail.com), Kent L. Gee, Mark K. Transtrum (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), Shane V. Lympany, and Matthew F. Calton (Blue Ridge Res. and Consulting, LLC, Asheville, NC)

The Federal Highway Administration's Traffic Noise Model is the standard model used to predict traffic noise in the United States. The Department of Transportation's National Transportation Noise Map (NTNM), based on average annual daily traffic counts, uses the traffic noise model to map out an average A-weighted equivalent sound level based on road traffic. Because the NTNM does not account for temporal variation, measured acoustic levels often differ from the average predicted levels. A recent internally developed method uses state highway agencies' reported traffic counts to create a model to predict hourly traffic across the United States based on geospatial features. This predictive approach can be combined with acoustical propagation algorithms in the traffic noise model to create temporally variable traffic noise maps. Because predicted traffic counts match reported traffic counts more closely than averaged counts, predicted sound levels are more characteristic of measured sound levels than the NTNM. The expected sound level errors using the predictive approach are shown to be significantly smaller than the expected NTNM errors. [Work supported by U.S. Army SBIR.]

Session 2pPAa

Physical Acoustics, Noise, Computational Acoustics, and Biomedical Acoustics: Sonic Boom Focusing Predictions

Alexandra Loubeau, Cochair
NASA Langley Research Center, MS 463, Hampton, VA 23681

Sriram Rallabhandi, Cochair
Aeronautics Systems Analysis Branch, NASA Langley, Room 190-25, Mailstop 442, NASA Langley Research Center, Hampton, VA 23681

Chair's Introduction—1:00

Invited Paper

1:05

2pPAa1. Focus boom simulations using sBOOM and lossy nonlinear Tricomi equation solvers. Sriram Rallabhandi (Aeronautics Systems Anal. Branch, NASA Langley, Rm. 190-25, Mailstop 442, NASA Langley Res. Ctr., Hampton, VA 23681, sriram.rallabhandi@nasa.gov)

The low-boom prediction committee, comprised of several research agencies and companies across the world, is organizing a special session dedicated to the prediction and comparison of focused sonic booms. This presentation will document and discuss focus boom simulations corresponding to the benchmark cases prescribed for this special session using a combination of sBOOM and lossy nonlinear Tricomi equation (LNTE) solvers. sBOOM is an augmented Burgers equation based sonic boom propagation solver that has been developed at NASA over the past decade. Focus boom simulations involve regions where simulations based solely on traditional ray theory approaches, such as those used in sBOOM, are inadequate. The results from sBOOM are, therefore, coupled with an LNTE solver that considers diffraction effects, which become dominant in the vicinity of imaginary surfaces (caustics) in the atmosphere on which singularities arise when using ray theory approaches. The LNTE solver used in this study has been developed and successfully demonstrated as part of the NASA's superboom caustic analysis and measurement project (SCAMP). Sonic boom focused signatures and other relevant metrics of the coupled simulations will be documented and discussed along with interesting observations.

Contributed Paper

1:30

2pPAa2. Contribution to the definition of a simple focusing test case and ONERA results for the two test cases of the sonic boom focusing predictions special session. Gérald Carrier (Wind Tunnels Directorate, ONERA Fluid Mech. and Energetics Branch, ONERA Mecanique des fluides et energetique, Palaiseau, Île-de-France, FR, academic/eng, 8 rue des Vertugadin, Meudon 92190, France, carrier@onera.fr) and François Coulouvrat (Institut Jean Le Rond d'Alembert, Sorbonne Université & CNRS, Paris, France)

For usual cruise flight conditions, the prediction of the sonic boom on ground caused by an aircraft flying supersonically can be predicted numerically by ray tracing codes according to the geometrical acoustics hypotheses. Such codes account for the non-linear and dissipative (thermoviscous and molecular relaxation) effects that occur during the propagation from flight level down to the ground by solving dissipative Burgers' equations along the rays, accounting for the evolution of the ray tube area. Such an approach is not applicable to predict the pressure waveform in the neighbourhood of a caustics that can be generated by the cusp of the focussing pressure wave generated under specific unsteady flight conditions of the aircraft. For such focussing conditions, the pressure waveform can be modelled by the Tricomi equations. The BANGV code has been used to predict the pressure waveforms in focussing conditions of the sonic boom focusing predictions special session of the 182nd ASA meeting. The results obtained for the two test-cases will be presented in this paper to be compared to other participants' results and contribute to the validation of numerical simulation methods of focussing boom.

Invited Paper

1:45

2pPAa3. Analysis of focus boom test cases using the lossy nonlinear Tricomi equation. Joseph A. Salamone (Eng., Boom Supersonic, 12876 East Adam Circle, Centennial, CO 80112, joe.salamone@boom.aero)

Focused sonic booms will result from an aircraft accelerating from subsonic to supersonic airspeeds. Sonic boom focusing leads to amplification and phase change of the sonic boom signature and can be modeled by a form of the nonlinear Tricomi equation. This presentation will use the lossy nonlinear Tricomi equation (LNTE) to examine sonic boom focusing for two sample cases. The first case depicts an aircraft with constant rectilinear acceleration at a constant altitude. The second case is slightly more complicated with an aircraft maneuvering at a small climb angle, while accelerating and simultaneously pitching downward. Time-pressure histories at various locations in the illuminated zone above the caustic and in the evanescent zone below the caustic will be provided for each test case.

Contributed Paper

2:10

2pPAa4. Sonic boom focus modeling of benchmark cases using the PCBoom suite. Robert Downs (US DOT Volpe National Transportation Systems Ctr., 55 Broadway, V-324, Cambridge, MA 02135, robert.downs@dot.gov) and Sophie Kaye (US DOT Volpe National Transportation Systems Ctr., Cambridge, MA)

Focusing of sonic booms is a phenomenon that can arise from particular atmospheric conditions or aircraft trajectories including acceleration, turns, or other variations from steady level flight. Because such maneuvers are unavoidable, development of regulations for civil supersonic flight will depend on the capability to understand and predict the scenarios in which focus booms can lead to ground overpressures several times higher than typical carpet booms. Models for predicting sonic boom focal zones and focus boom signatures have been developed by several organizations, and a recent AIAA Sonic Boom Prediction Workshop provided an opportunity to compare results among such models. Rallabhandi *et al*. developed benchmark cases to further extend verification of focus boom models. The PCBoom suite of tools, including the lossy nonlinear Tricomi equation (LNTE) module, was used to predict caustic geometry and pressure signatures near the caustic arising from a constant acceleration level trajectory and also from an accelerating climb trajectory. A recently developed method to streamline the focus boom modeling process within PCBoom was exercised as part of this benchmark case evaluation. [Work supported by the National Aeronautics and Space Administration under project VXAHA321.]

2:25–2:55
Panel Discussion

Session 2pPAb

Physical Acoustics, Biomedical Acoustics, and Structural Acoustics and Vibration: Vortex Beams and Radiation Torques II

Likun Zhang, Cochair
National Center for Physical Acoustics and Department of Physics and Astronomy, University of Mississippi, University, MS 38655

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

Chair's Introduction—1:00

Invited Papers

1:05

2pPAb1. Vortex beam interactions for the measurement of nonlinear acoustic behaviour. Grace Richard, Sandy Cochran (James Watt School of Eng., Univ. of Glasgow, Glasgow, United Kingdom), Gabriel Spalding (Illinois Wesleyan Univ., Bloomington, IL), and Martin Lavery (James Watt School of Eng., Univ. of Glasgow, University of Glasgow, Glasgow G12 8QQ, United Kingdom, martin.lavery@glasgow.ac.uk)

Spatially shaped optical fields have been widely implemented for remote sensing, enhanced imaging and communications. Many of these principles can be readily translated into acoustics, where acoustic fields that carry orbital angular momentum (OAM) have gained considerable interest in recent years. These spatially shaped fields can provide unique acoustic interactions with materials and gasses as they propagate that can be leveraged to create novel sensor technologies. We will present a technique to amplify the visibility of nonlinear acoustic effects in gasses, via the controlled interference of acoustic pulses that carry OAM. This amplification allows accurate determination of the nonlinear parameter with acoustic pressures that are 66dB lower than a direct measurement of pulse chirping. In controlled conditions, non-linear parameters can be directly measured; however, variations in environmental parameters can lead to ambiguity, which can be addressed through the use of a trained Support Vector Machine machine learning approach. This will allow measurements in variation in the nonlinear parameter 0.01 with over 99.5% accuracy and 0.001 with over 75% accuracy based on single measurements, where repeated measurements can further reduce the power requirements and increase accuracy. This approach can be used for monitoring changes in air quality and if extended to other frequencies could be used for monitoring fatiguing in metals.

1:30

2pPAb2. Doing high power airborne vortex beams with ultrasonic vibrations. Joao Ealo (School of Mech. Eng., Universidad del Valle, Ciudad Universitaria Meléndez, Bldg. E49 - Of.2014, Cali, Valle del Cauca 760032, Colombia, joao.ealo@correounivalle.edu.co), Jhon Pazos-Ospina, Jhoan Acevedo-Espinosa (School of Mech. Eng., Universidad del Valle, Cali, Valle del Cauca, Colombia), Juan Rojas-Valencia (School of Mech. Eng., Universidad del Valle, Cali, Colombia), and Carlos Muñoz-Delgado (School of Mech. Eng., Universidad del Valle, Cali, Valle del Cauca, Colombia)

Vortex beams (VBs) are able to transfer angular momentum to matter and exhibit self-reconstruction capacity and a great potential in engineering applications such as particle manipulation, communications, imaging, among others. Several VB generation methods have been put forward, e.g., physically helical sources, phase-driven multi emitters, passive metastructured devices, and recently, electroactive spiral gratings. Even though airborne ultrasound technologies offer interesting benefits, such as the free access to the manipulated specimen/particle, most of the reported methods are intended for water applications. Here, a sonotrode-based approach is introduced, in which ultrasonic vibrations are induced to produce a high quality, high power VB in air. We designed, fabricated, and characterized two different prototypes with radiant structures that operate in length-extension and transverse vibration modes at 27.6 kHz and 40.7 kHz, respectively. Numerical simulation models were implemented at the design stage, which allowed us a proper selection of the materials and geometry of the assembly. The prototypes allow for intensities over 0.5 W/cm2 in open-loop operation. A discussion on the electro-acoustic performance of the devices is included. New design configurations are possible for higher order VB generation with amplified acoustic output. This work opens up new manipulation possibilities for engineering applications in air and also in water.

1:55

2pPAb3. Synthesizing acoustic drill beams by the superposition of detuned vortices. Noé Jiménez (Instituto de Instrumentación para Imagen Molecular, Universitat Politècnica de València - CSIC, Camino de Vera s/n, Valencia 46022, Spain, nojigon@upv.es), Enrique González-Mateo, Francisco Camarena (Instituto de Instrumentación para Imagen Molecular, Universitat Politècnica de València - CSIC, València, Spain), and Kestutis Staliunas (UPC, Dep. de Física, ICREA, Terrassa, Spain)

We present acoustic drill beams. These singular beams show a dynamic intensity distribution matching the shape of a helix. The intensity distribution rotates along the axis of the beam with a controlled direction and angular frequency, therefore resembling the shape of a mechanical drill bit. Acoustic drills emerge elegantly as the spatio-temporal interference of two confocal and detuned vortex beams. The beam parameters are fully tuneable. The detuned frequency, detuning wave number, and detuned topological charge of the composing beams control the number of arms of the drill, the winding period, the rotation velocity, and its direction. We show that elongated drill beams are obtained using two high-order Bessel beams, enabling analytical solutions for optimal overlapping. In addition, drill beams can also be synthesized by focused ultrasound vortices. Analytic, numeric, and experimental results are shown in the ultrasound regime using a low-cost device based on two confocal 1-MHz piezoelectric transducers and 3D-printed acoustic holograms. This new wave structure opens novel avenues for wave-matter interaction such as contactless particle manipulation, matter processing, or biomedical applications.

2:20–2:35 Break

2:35

2pPAb4. Generation and steering of airborne acoustic vortices with broadband electro-active spiral gratings. Joao Ealo (School of Mech. Eng., Universidad del Valle, Ciudad Universitaria Meléndez, Bldg. E49 - Of.2014, Cali, Valle del Cauca 760032, Colombia, joao.ealo@correounivalle.edu.co), Ruben Muelas-Hurtado (School of Mech. Eng., Universidad del Valle, Cali, Valle del Cauca, Colombia), and Karen Volke-Sepúlveda (Instituto de Física, Universidad Nacional Autónoma de México, Ciudad de Mexico, Mexico, Mexico)

Acoustic beam shaping technology has flourished in the last decade, motivated by its important applications. For example, in the context of contactless manipulation, the use of structured ultrasonic fields has led to significant advances, such as the development of the acoustic tweezers based on the use of ultrasonic vortex beams. In this work, we present a powerful and efficient technique to structure acoustic fields based on the use of planar electro-active diffraction gratings, which can be tailored in practically arbitrary shapes. For this purpose, a lower electrode with the desired shape is printed on a circuit board and covered with an active ferroelectret film, whose metallized top surface acts as a continuous upper electrode. Importantly, these devices can be operated within a broad spectral range of ultrasonic frequencies. Two kinds of structured fields are demonstrated: a focused acoustic vortex and the simultaneous generation of multiple acoustic Bessel beams of different topological charges, well separated among each other along the propagation axis. In both cases, the main parameters of the field can be finely and continuously tuned by setting the operation frequency, which allows an axial steering of the vortices. Experimental results exhibit very good agreement with theoretical analysis and numerical simulations.

3:00

2pPAb5. Active holographic tweezers based on spiraling interdigitated transducers. Michael Baudoin (IEMN, Université de Lille, Ave. Poincaré, Villeneuve d'Ascq 59652, France, michael.baudoin@univ-lille.fr), Roudy Al Sahely (IEMN, Université de Lille, Villeneuve d'Ascq, France), Zhixiong Gong (IEMN, Université de Lille, Lille, France), Olivier Bou Matar (IEMN, Ecole Centrale de Lille, Villeneuve d'Ascq, France), Jean-Claude Gerbedoen (IEMN, CNRS, Lille, France), Nikolay Smagin (IEMN, UPHF, Valenciennes, France), and Ravinder Chutani (IEMN, Université de Lille, Villeneuve d'Ascq, France)

Recently, our team demonstrated the selective manipulations of microparticles [Baudoin *et al*., Sci. Adv. 5, eaav1967 (2019)] and cells [Baudoin *et al*., Nat. Commun. 11, 4244 (2020)] in a standard microscopy environment with active holographic tweezers based on spiraling interdigitated transducers. In this talk, we will present (i) the ultra-high frequency (250 MHz) selective manipulation of 4 μm beads with nanoNewton forces and (ii) first results on the the 3D manipulation of particles and in particular their axial displacement without any motion of the transducer, by tuning the driving frequency [Gong *et al*., Phys. Rev. Appl. 16, 024034 (2021)]. We will also introduce a new mixed finite element/angular spectrum code, which enables to understand the underlying physic and opens perspectives to improve the tweezers design.

3:25

2pPAb6. Single focused-beam acoustical tweezers. Zhixiong Gong (CNRS UMR8520 IEMN, IEMN, Cité Scientifique Ave. Henri Poincaré CS 60069, Lille 59652, France, zhixiong.gong@iemn.fr) and Michael Baudoin (IEMN, Université de Lille, Villeneuve d'Ascq, France)

Single beam acoustical tweezers have seen rapid development for their potential applications *in vitro* and *in vivo* in different fields such as acoustofluidics, microrobotics, precision medicine, and so on. The first single beam acoustical tweezers used a focused beam in analogy with optical tweezers [Lee *et al*., Appl. Phys. Lett., 95, 073701 (2009)]. However, they only succeeded to trap light particles with negative acoustic contrast factor in two dimensions. In this talk, we revisit theoretically the capabilities of acoustical tweezers based on focused beams and demonstrate that they can trap (i) some specific light elastic particles and droplets in three dimensions in and beyond the Rayleigh regime, (ii) dense particles (with positive acoustic contrast factor) in two dimensions near the resonance frequencies, and (iii) lipid cells in three dimensions. Some preliminary experimental results will be presented.

3:50–4:10
Panel Discussion

Session 2pPPa

Psychological and Physiological Acoustics: Animal Physiology

Laurel H. Carney, Chair
Biomedical Engineering, University of Rochester, 601 Elmwood Ave., Box 603, Rochester, NY 14642

Chair's Introduction—1:00

Contributed Papers

1:05

2pPPa1. Physiological sensitivity to across-channel temporal patterns in response to Schroeder harmonic complexes, based on sequence rather than coincidence detection. Hsin-Wei Lu (Neurosciences, KU Leuven, Leuven, Belgium), Philip H. Smith (Neurosci., UW-Madison, Madison, WI), and Philip X. Joris (Neurosciences, KU Leuven, Herestraat 49, bus 1021, Leuven B-3000, Belgium, Philip.Joris@kuleuven.be)

The flat-envelope harmonic complexes devised by Schroeder (1970) contain a linear frequency sweep whose direction and speed depend on phase curvature C. Physiological studies have focused on responses at a given characteristic frequency (CF), which show only a subtle dependence on the sign and magnitude of C. The question arises whether CNS neurons are sensitive to the temporal pattern across nerve fibers differing in CF. Octopus cells in the cochlear nucleus receive convergent excitatory input from nerve fibers spanning a wide range of CFs and are thought to be monaural coincidence detectors. We studied their spike output and membrane responses to Schroeder complexes in anesthetized gerbils. Octopus cells were indeed tuned to a wide frequency range but showed discrete frequency "hotspots" of differing strength. Most cells showed marked sensitivity to C, with some neurons preferring upward and others downward frequency sweeps, which was broadly stable across SPL and fundamental frequency. The sensitivity to C arises not from coincidence detection, but from a sensitivity to the temporal sequence of activation of inputs tuned to different frequencies. We conclude that marked sensitivity to Schroeder phase exists at the first CNS processing stage and is based on temporal sequence detection across frequency channels.

1:20

2pPPa2. Testing the generation of harmonic templates from coincidence patterns in response to noise: Physiological and modeling considerations. Yi-Hsuan LI (KU Leuven, Campus Gasthuisberg, O&N 2 Herestraat 49 - bus 102, Leuven 3000, Belgium, yihsuan.li@kuleuven.be) and Philip X. Joris (Neurosciences, KU Leuven, Leuven, Belgium)

Harmonic templates are a critical component of spectral pitch theories. Shamma and Klein (2000) hypothesize that such templates can arise from neural coincidence patterns in response to stochastic stimuli without pitch. The key feature of their computational model is a higher number of coincidences between neurons with harmonically related characteristic frequencies (CFs). We tested the hypothesis with physiological data and a computational model. We recorded neural responses in chinchilla to various periodic and non-periodic stimuli, both from the auditory nerve and the trapezoid body. We then examined counts of spike coincidences across fibers. The key prediction of a higher number of coincidences between pairs of fibers with harmonically related CFs than for other pairs was observed for responses to narrowband signals (tones at CF), but not to broadband noise. We then tested a model similar to that of Shamma and Klein. Only with cochlear filters sharper than plausible even in humans, followed by additional spectral sharpening through lateral inhibition and extreme temporal sharpening, do we obtain harmonic templates similar to their model. Although our results do not speak to the existence of harmonic templates *per se*, we conclude that the specific mechanism of this hypothesis has little physiological plausibility.

1:35

2pPPa3. Physiological studies of timbre encoding in the auditory midbrain. Johanna B. Fritzinger (Neurosci., Univ. of Rochester, 601 Elmwood Ave. Box 603, Rochester, NY 14642, johanna_fritzinger@urmc.rochester.edu) and Laurel H. Carney (Biomedical Eng., Univ. of Rochester, Rochester, NY)

Perceived timbre, especially brightness, is influenced by the spectral peaks of sounds. We hypothesize that spectral peaks are reliably encoded in the inferior colliculus (IC) due to capture and off-characteristic frequency (CF) inhibition. Capture occurs when a component near CF saturates the inner-hair-cell response, reducing the amplitude of neural fluctuations, the low-frequency changes in auditory-nerve firing rate that arise from the interaction of components within complex sounds. IC neurons are sensitive to amplitude modulation and respond to neural fluctuations with enhanced or suppressed rates. Extracellular single-unit recordings were made in the IC of awake rabbits in response to synthetic harmonic stimuli with triangular-shaped spectral magnitudes and zero-phase. Stimuli were presented at 40–80 dB SPL to test for robust encoding over a range of suprathreshold levels. Responses were compared to the output of an auditory-nerve model that exhibits capture followed by a modulation-tuned midbrain model. Results were generally consistent with model predictions, especially for cells suppressed by modulations. Dependent on the modulation transfer function type, responses show dips or peaks in average rate over many levels when the peak harmonic frequency matches CF. This work fills a gap in neural timbre research by investigating spectral peak encoding in the auditory midbrain.

1:50

2pPPa4. Mechanisms of masking by Schroeder-phase harmonic tone complexes in the budgerigar. Kenneth Henry (Otolaryngol., Univ. of Rochester, 601 Elmwood Ave. Box 629, Rochester, NY 14642, kenneth_henry@urmc.rochester.edu), Yingxuan Wang, Kristina Abrams (Univ. of Rochester, Rochester, NY), and Laurel H. Carney (Biomedical Eng., Univ. of Rochester, Rochester, NY)

Schroeder-phase harmonic tone complexes have a flattened temporal envelope and either rising or falling instantaneous frequency glides within F0 periods, depending on the polarity of the phase relationship between harmonics. Human detection thresholds for a pure-tone target added to a Schroeder masker can be 10–15 dB lower for the negative Schroeder (upward gliding) compared to positive (downward gliding), possibly due to human cochlear mechanics though this hypothesis remains controversial. In contrast, several avian species show minimal behavioral masking differences between Schroeder polarities for F0s of 50 and 100 Hz. To gain further

insight into the mechanisms of Schroeder masking, we performed operant-conditioning experiments in budgerigars, a parakeet species, including F0s up to 400 Hz that are potentially better matched to avian cochlear mechanics. Awake neurophysiological recordings at the midbrain processing level characterized neural encoding of behavioral test stimuli. Behavioral Schroeder masking thresholds decreased with increasing F0 and showed minimal difference between Schroeder polarities even at high F0s. Neural recordings showed prominent temporal and rate-based encoding of F0 periodicity and, in some neurons, marked response asymmetries between Schroeder polarities. These results suggest that neural response asymmetries between Schroeder polarities need not produce large behavioral masking differences for pure-tone targets.

2:05

2pPPa5. Comparing frequency-chirp sensitivity in the inferior colliculus to basic response properties using novel stimuli. Paul Mitchell (Biomedical Eng., Univ. of Rochester, 601 Elmwood Ave. Box 603, Rochester, NY 14642, pmitche4@ur.rochester.edu) and Laurel H. Carney (Biomedical Eng., Univ. of Rochester, Rochester, NY)

Most neurons in the central nucleus of the inferior colliculus (ICC) are sensitive to the direction of phase-curvature of Schroeder-phase (SCHR) stimuli. This phase-curvature controls the direction and velocity of frequency chirps within the fundamental SCHR period. Differences in responses to different SCHR stimuli may originate from an underlying sensitivity to frequency chirp direction or velocity. However, known ICC sensitivities, such as those to frequency or envelope periodicity and duty cycle, may also contribute to the observed effect. To parse these confounding SCHR stimulus features, we designed novel frequency-chirping stimuli to isolate the influences of periodicity, duty cycle, and frequency-chirp direction or velocity on ICC neuron responses. Extracellular, single-unit recordings of chirp responses were made in awake rabbit ICC. Simultaneously, basic response properties were characterized using methods such as frequency response maps, modulation transfer functions, and spectro-temporal receptive fields. By comparing basic response properties across chirp-sensitive neurons (comprising 80% of ICC neurons), we can outline the shared characteristics among them. Our goal is to determine whether frequency-chirp sensitivity in ICC neurons is independent of other established feature sensitivities. Frequency-chirp-sensitive neurons are interesting, because they may play a unique role in encoding sounds with complex phase, including speech. [Work supported by NIDCD-R01-001641.]

2:20

2pPPa6. Towards auditory stream segregation in *Tursiops truncatus*. Matt Schalles (Neurosci. Inst., Carnegie Mellon Univ., Neurosci. Inst., 4400 Fifth Ave., Pittsburgh, PA 15213, matt.schalles@gmail.com), Jason Mulsow, Dorian Houser (National Marine Mammal Foundation, San Diego, CA), James J. Finneran (SSC Pacific Code 71510, US Navy Marine Mammal Program, San Diego, CA), Peter Tyack (Biology, Woods Hole Oceanographic Inst., Falmouth, MA), and Barbara Shinn-Cunningham (Neurosci. Inst., Carnegie Mellon Univ., Pittsburgh, PA)

In humans, understanding a voice amidst competing sounds depends on parsing the sound mixture into “streams” representing each source’s content. Streaming can be influenced by top-down attentional focus, while acoustic features can affect streaming percepts through bottom-up, automatic processing of pitch, timbre, and location. Dolphins regularly navigate a cacophony of echoes generated during echolocation, conspecific echolocation clicks, and other, passively received sounds (eg. communication signals, environmental noise), and thus, may rely on streaming. Using auditory evoked potentials (AEPs), we asked whether dolphins exhibit evidence of bottom-up, frequency-based stream segregation. Initial results using a classic A-B-A sequence of repeated tone triplets (either low-high-low or high-low-high tones) suggest that the triplets break apart perceptually into low and high streams; specifically, the AEP magnitude evoked by the middle tone increases with increasing frequency separation. Differences in dolphin hearing sensitivity across the frequencies tested seem to account for some of the frequency manipulation effect. Additionally, earlier studies suggest that the dolphin auditory temporal integration window decreases as frequency increases, which may interact with streaming processes. This lays a foundation for future tests of top-down effects on dolphin auditory stream processing such as attention and expectation.

Session 2pPPb

Psychological and Physiological Acoustics: Clinical Populations and Devices I (Poster Session)

Steven P. Gianakas, Cochair
Speech-Language-Hearing Sciences, University of Minnesota, 164 Pillsbury Dr SE, Minneapolis, MN 55455

Kristina DeRoy Milvae, Cochair
University of Maryland College Park, 7251 Preinkert Drive, College Park, MD 20742

All posters will be on display from 2:00 p.m. to 5:00 p.m. Authors of odd-numbered papers will be at their posters from 2:00 p.m. to 3:30 p.m. and authors of even-numbered papers will be at their posters from 3:30 p.m. to 5:00 p.m.

Contributed Papers

2pPPb1. Children with amblyaudia show less flexibility in auditory cortical entrainment to periodic non-speech sounds. Sara Momtaz, Deborah Moncrieff (School of Public Health, Univ. of Memphis, Memphis, TN), Meredith Ray (School of Public Health, Univ. of Memphis, Robison Hall 3825 DeSoto Ave., Memphis, TN 38152, maray@memphis.edu), and Gavin Bidelman (School of Public Health, Univ. of Memphis, Memphis, TN)

We investigated auditory temporal processing in children with amblyaudia (AMB), a subtype of auditory processing disorder, via cortical neural entrainment. Evoked responses were recorded to click-trains at slow versus fast (8.5 versus 14.9/sec) rates in n = 14 children with AMB and n = 11 age-matched controls. Source and time-frequency analyses decomposed EEGs into oscillations (reflecting neural entrainment) stemming from the bilateral auditory cortex. Phase-locking strength in AMB depended critically on the speed of auditory stimuli. In contrast to age-matched peers, AMB responses were largely insensitive to rate manipulations. This rate resistance was seen regardless of the ear of presentation and in both cortical hemispheres. Children with AMB show a stark inflexibility in auditory cortical entrainment to rapid sounds. In addition to reduced capacity to integrate information between the ears, we identify a new functional characterization of AMB in the form of more rigid tagging of external auditory stimuli. Our neurophysiological findings may account for certain temporal processing deficits commonly observed in AMB and related auditory processing disorders (APDs) behaviorally. More broadly, our findings may inform communication strategies and future rehabilitation programs; increasing the rate of stimuli above a normal (slow) speech rate is likely to make stimulus processing more challenging for individuals with AMB/APD.

2pPPb2. Perception of temporal cues in word segments under bimodal divided attention in simulated cochlear implant listening. Viv Song, Hannah Brown, Michelle Low (Hearing and Speech, Univ. of Kansas Medical Ctr., Kansas City, KS), and Zilong Xie (Hearing and Speech, Univ. of Kansas Medical Ctr., M.S. 3047, 3901 Rainbow Blvd., Kansas City, KS 66160, zxie2@kumc.edu)

Auditory temporal processing is important for cochlear-implant (CI) users to understand speech. Speech processing often unfolds along with sensory inputs from other modalities. This study examined whether dividing attention across modalities affects temporal processing of word segments in simulated CI listening. Adults with normal hearing performed categorization tasks on unprocessed and 8-channel vocoded versions of a Dish-Ditch contrast with varying silent intervals preceding the word-final fricative. The categorization performance was measured concurrently with a visual working memory task on meaningless images. This divided attention (AV-div) condition was compared with an audiovisual selective attention (AV-sel) condition in which participants focused on the categorization task and ignored the visual stimuli, and with an auditory-only (AO) condition in which participants performed the categorization task without the visual stimuli. Preliminary results (N = 5) revealed that participants tended to report more "Dish" responses for words with longer silence intervals under the two AV conditions than the AO condition, especially for the AV-div condition. The AV-related effects on the categorization performance appear to be reduced for vocoded than unprocessed words. These results suggest that concurrent visual information affects auditory temporal processing, which might lead to speech understanding difficulties of CI users in multisensory contexts.

2pPPb3. Perception of temporal cues in word segments as a function of spectral resolution and stimulus context. Hannah Brown, Viv Song, Michelle Low (Hearing and Speech, Univ. of Kansas Medical Ctr., Kansas City, KS), and Zilong Xie (Hearing and Speech, Univ. of Kansas Medical Ctr., M.S. 3047, 3901 Rainbow Blvd., Kansas City, KS 66160, zxie2@kumc.edu)

Auditory temporal processing is affected by the spectral resolution of the target stimuli (e.g., processed via a cochlear implant) and the surrounding stimulus context. This study examined how spectral resolution interacts with stimulus context to influence the perception of temporal cues in word segments. Normal-hearing adults performed categorization tasks on unprocessed and 8-channel vocoded versions of a Buy-Pie word contrast with varying voice-onset time (VOT). Besides being presented in isolation, the target words were preceded by a sentence or sentence-envelope-modulated or unmodulated speech-shaped noises. The carrier stimuli were presented ipsilaterally or contralaterally to the target words. Preliminarily results (N = 7) showed that participants tended to report more "Buy" responses in conditions with carrier stimuli than in isolation, particularly for words with longer VOTs. The context effect appears to be largely unchanged between unprocessed and vocoded words. The context effect was significantly reduced for the contralateral presentation of the carrier stimuli than the ipsilateral presentation. These results suggest that stimulus context may affect auditory temporal processing independent of spectral resolution, and the context effect appears to be primarily governed by peripheral processing. Such context effect might present some challenges for cochlear-implants users to perceive connected speech in real-life situations.

2pPPb4. Abstract withdrawn.

2pPPb5. Measuring sensitivity to envelope interaural time differences by adapting modulation depth. Virginia Best (Dept. Speech, Lang. and Hearing Sci., Boston Univ., 635 Commonwealth Ave. Boston, MA 02215, ginbest@bu.edu) and Christopher Conroy (Dept. Speech, Lang. and Hearing Sci., Boston Univ., Boston, MA)

Listeners are sensitive to interaural time differences carried in the envelope of high-frequency sounds (ITD_{ENV}), but the salience of this cue depends on several envelope properties. Making use of the fact that sensitivity to ITD_{ENV} varies systematically with the depth of modulation of sinusoidally amplitude-modulated tones, we devised a task in which modulation depth is varied adaptively to measure ITD_{ENV} sensitivity. In our task, the stimulus is a 4 kHz tone modulated at rates of 32, 64, or 128 Hz, with a fixed ITD of ± 500 μs. Modulation depth is adaptively varied (in dB) and listeners make left-right judgements. The data are used to obtain a psychometric function and threshold value. For listeners with normal hearing, we find that thresholds vary with modulation rate and with sensation level as expected. This task provides us with a convenient means for investigating ITD_{ENV} sensitivity in listeners with sensorineural hearing loss. Given that this population shows enhanced sensitivity to amplitude modulation under certain conditions, generally attributed to a loss of cochlear compression, we will test the hypothesis that this translates into superior ITD_{ENV} sensitivity under similar conditions.

2pPPb6. Feasibility of a hearing-aid speech quality index designed for noisy measurement conditions. James M. Kates (Speech Lang. Hearing Sci., Univ. of Colorado, 409 UCB, Boulder, CO 80309, james.kates@colorado.edu), Emily Lundberg (Speech Lang. Hearing Sci., Univ. of Colorado, Boulder, CO), Ramesh Kumar Muralimanohar (Dept. of Speech, Lang., and Hearing Sci., Univ. of Colorado Boulder, Beaverton, OR), and Kathryn Arehart (Speech Lang. Hearing Sci., Univ. of Colorado, Boulder, CO)

A goal in fitting hearing aids is to find settings that improve listener judgments of the amplified speech quality. Objective metrics that predict speech quality could assist in this fitting procedure and, thus, become useful tools in the clinic. The Hearing-Aid Speech Quality Index version 2 (HASQI v2) is effective in predicting speech quality ratings for speech processed through a hearing aid. However, HASQI v2 requires a signal-to-noise ratio (SNR) of approximately 40 dB to provide reliable measurements of low levels of nonlinear distortion, and this favorable an SNR is not always found in a clinical examination room or medical office where hearing-aid measurements are performed. Two modifications to HASQI v2 are considered in this presentation to improve the measurement accuracy in noisy situations. The first is applying dynamic-range compression to the test signal to enhance low-intensity speech sounds that could be masked by background noise, and the second is using a multilayer neural network to match noisy measurements to the corresponding measurements made in a quiet laboratory. The benefits of these modifications to HASQI are evaluated using a hearing-aid simulation, and verification is provided using commercial hearing-aid measurements made in an audiology clinic.

2pPPb7. Initial validations of an iterative hearing-aid self-fitting procedure with Monte-Carlo simulations. Bertan Kursun (Speech & Hearing Sci., Univ. of Washington, 4547 8th Ave. NE 409, Seattle, WA 98105, bertanks@uw.edu), Chemay Shola (Bioengineering, Univ. of Washington, Vancouver, WA), and Yi Shen (Speech and Hearing Sci., Univ. of Washington, Seattle, WA)

Self-directed gain adjustments may present a satisfactory solution for fitting a hearing aid (HA) without the requirement of supervision from a clinician. The current study develops a self-fitting procedure and provides initial evaluations of the procedure using Monte-Carlo simulations. In the procedure, the user interacts with the HA using a mobile device with a touch screen, while a continuous speech as a stimulus and a babble background noise is played in the room. The user explores the locations on the touch screen while hearing its effects on the hearing-aid processed audio in real time, until a preferred setting is identified. This adjustment process is repeated over trials, with the mathematical map between the touch-screen location (2D-coordinates) and the amplification profile of the HA being updated from trial to trial. This algorithm is evaluated using simulated listeners with known preferred amplification profiles serving as the ground truth. Results showed that estimates of the user preferred amplification profile, close to the ground truth, can be obtained by taking the mean of the user identified gain settings at the end of each trial.

2pPPb8. Perception of prosodic cues to correct mistakes, in listeners with normal hearing and with cochlear implants. Harley J. Wheeler (Speech-Language-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, wheel488@umn.edu), Tereza Krogseng, and Matthew B. Winn (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN)

Prosody is used to mark important information in speech yet is not an integral part of speech recognition testing among people with hearing difficulty. While a listener may correctly perceive words spoken, they may not perceive meaningful emphasis on a certain word, which could be critically important in a variety of everyday situations. This study introduces a new paradigm for assessing perception of prosodic cues in listeners with normal hearing and with cochlear implants, who have notorious difficulty perceiving pitch. Stimuli consisted of spoken sentences where one word (in various sentence positions) was emphasized in a manner that indicated that a prior statement was incorrect in a specific way. Participants used a visual analog scale to mark the timing and degree of emphasis aligned with the target words. Here, the perceptual data are linked with acoustic measures of voice pitch contour, intensity, duration, and vocal quality to characterize how contrastive stress cues are recovered by listeners with and without hearing impairment.

2pPPb9. Optimizing hearing aids for music perception in noise using subjective and objective measures. Shubhaganga D (Dept. of Audiol., All India Inst. of Speech and Hearing, Dept. of Audiol., all India Inst. of speech and hearing, Mysuru 570006, India, dshubhaganga94@gmail.com) and Nisha K V (Dept. of Audiol., All India Inst. of Speech and Hearing, Mysuru, India)

The perception of music is significantly compromised in the individuals using hearing aids (HA), which becomes further degraded in the presence of noise. Although HAs are heavily optimized for speech intelligibility in noise, they often are not well optimized for music perception, especially in the presence of noise. This study investigated the effect of signal-to-noise ratios (SNR) on music processed through HAs in two programs (1- Speech & 2- Music) using subjective and objective measures. The HA (fitted to NAL-NL2) processed music stimuli in the two programs at 5 SNRs (0, + 5, + 10, + 15 & + 20) were rated perceptually by 20 normal-hearing participants on a visual analogue scale of "Adjective descriptors of Music" adapted from "Iowa Musical Background Questionnaire," while the quality of music was objectively analyzed using a music quality index (HAAQI) using MATLAB code. Results showed that the music perception was better for higher SNRs (+ 15 & + 20) than lower SNRs (0, + 5 &+10) in both subjective and objective measures for both the programs. The effect of SNRs was more pronounced in the music program. These results emphasize the need for subjective and objective validation of HA programs for music listening in clinical setups.

2pPPb10. The effects of cochlear-implant processing and current spread on the weighting of spectrotemporal speech information in noise. Bobby E. Gibbs (Hearing and Speech Sci., Univ. of Maryland, College Park, 7251 Preinkert Dr., College Park, MD 20742, gibbsbe@umd.edu) and Matthew J. Goupell (Hearing and Speech Sci., Univ. of Maryland-College Park, College Park, MD)

CI listeners show a different weighting of acoustic information in quiet compared to NH listeners. For example, they rely less on cues such as formant transitions. However, less is known about how CI listeners weight spectrotemporal speech information in noise. It is hypothesized that the spectrotemporal regions of noise that are most critical to intelligibility for CI listeners are distinct from those for NH listeners. Specifically, CI processing and current spread are hypothesized to shift importance towards coarse temporal regions. NH listeners should weight regions associated with formant transitions and rapid transients more heavily. To test these hypotheses, an experiment was designed to reveal spectrotemporal importance functions for speech-in-noise intelligibility in NH listeners presented unprocessed and CI-simulated stimuli. The stimuli will include unprocessed vowel-consonant-vowel words presented in "bubble" noise, which randomly attenuates different spectrotemporal regions. Listeners (N = 8 expected) will also be presented with vocoded stimuli with attenuation rates simulating broad or shallow current spread. Spectrotemporal importance functions will be assessed through point-biserial correlation by comparing how different bubble noise regions impact intelligibility. These data could help inform improvements in CI processing and noise-reduction strategies, which often assume relatively uniform weighting of acoustic information.

2pPPb11. The changes to interaural acoustics imparted by placing circumaural headphones over bilateral cochlear-implant sound processors. Paul G. Mayo (Hearing and Speech Sci., Univ. of Maryland-College Park, 0100 LeFrak Hall, 7251 Preinkert Dr., College Park, MD 20742, paulmayo@umd.edu) and Matthew J. Goupell (Hearing and Speech Sci., Univ. of Maryland-College Park, College Park, MD)

Interaural level differences (ILDs) are the primary binaural cue used by bilateral cochlear-implant (BiCI) listeners for horizontal-plane sound localization. As such, factors affecting their fidelity are important to understand. One approach for delivering binaural stimuli to BiCI listeners is to present virtually spatialized signals via circumaural headphones placed over the listener's sound processors. An assumption of this approach is that the binaural cues presented are relatively unaltered by the transmission process and headphone placement; yet there is a lack of evidence supporting this assumption. Therefore, this study measured the effect of small changes to headphone placement on the ILDs received by BiCI sound processors. The intent was to determine both the extent and frequency range in which ILDs are affected using recordings of sine-sweeps received by the device microphones. Results show that slight changes in headphone placement affect coupling to the sound processor microphones and outer ear, primarily influencing ILDs between 1 and 5 kHz as much as 8.6 dB at the midline. They also show that occluding the listener's ear canals with earplugs or moldable putty significantly reduces ILD variability from inconsistent headphone placement. In summary, these data suggest ways to improve this presentation method for BiCI research studies.

2pPPb12. The effects of aging and hearing loss on the across-frequency processing of interaural time differences. Anhelina Bilokon (Hearing and Speech Sci., Univ. of Maryland-College Park, 7251 Preinkert Dr., College Park, MD 20742, abilokon@umd.edu) and Matthew J. Goupell (Hearing and Speech Sci., Univ. of Maryland-College Park, College Park, MD)

Aging and hearing loss cause speech understanding deficits, particularly in the presence of competing sounds, as well as temporal processing deficits, including binaural encoding of interaural time differences (ITDs). However, there is little understanding of the independent effects of aging and hearing loss on across-frequency ITD processing, which demonstrates a clear dependence on frequency weighting or "dominant" region around 600–700 Hz. We hypothesize that aging and hearing loss will reduce extent of laterality of ITDs and will shift the frequency weighting lower. Young normal-hearing, older normal-hearing, and older hearing-impaired listeners will be tested on an intracranial lateralization task. The stimuli will consist of narrowband noises. They will have upper frequency boundaries = 500, 600, 700, or 800 Hz, bandwidths = 50, 100, 200, or 400 Hz, and ITDs = 0, ±0.5, ±1, ±1.5, and ±2 ms. These findings will help us better understand the role of the dominant region in across-frequency ITD processing, and how age- and hearing-loss-related changes in ITD processing may contribute to spatial-hearing deficits.

2pPPb13. Evaluating central auditory processing in Veterans using population-appropriate norms. Lauren Stadel (Audiol., Veterans Affairs Pittsburgh Healthcare System, 4100 Allequippa St., Pittsburgh, PA 15240, LAP142@pitt.edu), Leslie Zhen (Commun. Sci. and Disord., Univ. of Pittsburgh, Pittsburgh, PA), Frederick J. Gallun (Oregon Hearing Res. Ctr., Oregon Health and Sci. Univ., Portland, OR), David Jedlicka, Elizabeth Haley (Audiol., Veterans Affairs Pittsburgh Healthcare System, Pittsburgh, PA), Lindsey Jorgensen, Michelle Novak (Sioux Falls Veterans Affairs Healthcare System, Sioux Falls, SD), Serena Dann (Veterans Affairs Portland Healthcare System, Portland, OR), Lee Baugh (Sioux Falls Veterans Affairs Healthcare System, Sioux Falls, SD), Kelene Fercho (FAA Civil Aerosp. Medical Inst., Sioux Falls, SD), Malcolm McNeil (Veterans Affairs Pittsburgh Healthcare System, Pittsburgh, PA), and Sheila Pratt (Audiol., Veterans Affairs Pittsburgh Healthcare System, Pittsburgh, PA)

Increasing numbers of military Veterans have reported substantive self-perceived hearing handicap despite normal/near normal hearing on standard audiometric testing. These hearing difficulties are especially true for Veterans with histories of blast exposure and presumed mild traumatic brain injury, with many of these Veterans also being comorbid for post-traumatic stress disorder (PTSD). Clinical tests of Central Auditory Processing Disorder (CAPD) commonly are used to investigate their hearing concerns, but performance on CAPD tests typically is compared to published norms established from non-Veteran populations. Since Veterans and non-Veterans differ significantly in many relevant domains, using non-Veteran norms is potentially problematic for the interpretation of CAPD test results. Veterans ($n = 217$; with and without histories of blast exposure and PTSD) completed a large CAPD test battery. An $N = 1$ approach was used to compare the performance of the blast-exposed Veterans and those with PTSD to published norms and data from the Veterans without histories of blast exposure and PTSD (control group). The number of abnormal cases identified based on the Veteran control group was lower than that found with the published norms for all tests. Therefore, establishing and using population-appropriate norms are critical for accurately diagnosing CAPD in Veterans.

2pPPb14. Noise exposure, binaural envelope processing, and spatial release from masking in student musicians with normal hearing. Chhayakanta Patro (Speech Lang. Pathol. and Audiol., Towson Univ., 326 Stevenson Ln., B8, Towson, MD 21204, cpatro@towson.edu) and Nirmal Kumar Srinivasan (Audiol., Speech-Lang. Pathol., and Deaf Studies, Towson Univ., Towson, MD)

Musicians are at risk of cochlear synaptopathy, because they are frequently exposed to hazardous sound levels that exceed the daily dose of noise exposure. In this study, we evaluated the effects of noise exposure on physiological, and perceptual correlates of cochlear synaptopathy in student musicians and non-musicians with normal audiometric thresholds. Auditory brainstem responses (ABRs) were recorded at various click rates to investigate the physiological findings consistent with synaptopathy. Sensitivity to interaural envelope time difference, and spatial release from speech-on-speech masking, were measured to characterize the perceptual deficits consistent with those expected from cochlear synaptopathy. Preliminary analyses suggested that rate-dependent ABR wave I amplitude reductions and wave V latency shifts were greater in the musicians, compared to their non-musician counterparts. However, performance on the two perceptual tasks did not differ across the two groups. These results suggest that there may be sub-clinical effects of excessive noise exposure in student musicians, but the effects are too subtle or too diffuse to precipitate perceptual deficits.

2pPPb15. Characterizing an auditory symptom subtype of concussion. Tess K. Koerner (VA RR&D NCRAR, 3710 SW US Veterans Hosp. Rd., Portland, OR 97239, Tess.Koerner@va.gov), Melissa A. Papesh, Sarah M. Theodoroff (VA RR&D NCRAR, Portland, OR), Nicole Dean (Oregon Hearing Res. Ctr., Oregon Health and Sci. Univ., Portland, OR), Sean D. Kampel (VA RR&D NCRAR, Portland, OR), Jennifer Wilhelm (Neurology, Oregon Health & Sci. Univ., Portland, OR), Ryan Rockwood, James Chesnutt (Family Medicine, Oregon Health & Sci. Univ., Portland, OR), and Frederick J. Gallun (Oregon Hearing Res. Ctr., Oregon Health and Sci. Univ., Portland, OR)

Anecdotal clinical reports and recent research findings suggest that auditory symptoms are common following a mild traumatic brain injury (mTBI). Typically, only tinnitus is captured on a case history. Self-reported noise sensitivity/hyperacusis and difficulty with speech are not commonly included in the assessment of mTBI symptoms. The present study completed a Principal Component Analysis on responses to three commonly administered hearing health questionnaires from 95 normal-hearing Veterans and non-Veterans with and without history of mTBI. Results allowed for the identification of self-report items reflective of auditory symptoms in individuals with mTBI. These results were combined with expert opinion to develop eight auditory symptom rating-scale items that were added to a new mTBI symptom questionnaire, the Concussion Symptom Subtype Inventory (CSSI). Data collection using the CSSI is ongoing at the Oregon Health and Science University Concussion Clinic. The development of the auditory items and current data from the CSSI will be discussed. This work will provide important information about auditory symptoms in this patient population and will help drive future research that aims to better understand and measure the effects of mTBI on auditory processing. Future work will aim to assess the validity of the CSSI for clinically assessing mTBI symptoms.

2pPPb16. Formant-frequency discrimination in listeners with hearing loss: Behavioral results and neural models. Laurel H. Carney (Biomed. Eng. & Neurosci., Univ. of Rochester, 601 Elmwood Ave. Box 603, Rochester, NY 14642, Laurel.Carney@Rochester.edu), Douglas M. Schwarz, David A. Cameron (Biomed. Eng. & Neurosci., Univ. of Rochester, Rochester, NY), U-Cheng Leong (Otolaryngol., Univ. of Rochester, Rochester, NY), and Joyce M. McDonough (Linguist, Univ. of Rochester, Rochester, NY)

In responses of the healthy ear to harmonic sounds, the amplitude of f0-related fluctuations of auditory-nerve (AN) responses varies along the tonotopic axis. Neural fluctuation amplitudes encode spectral peaks, including formant frequencies of vowels, because responses of inner-hair-cells (IHCs) tuned near spectral peaks are captured (or dominated) by a single harmonic, resulting in lower fluctuation amplitudes than IHCs tuned between spectral peaks. This code is robust across a wide range of sound levels and in background noise, and is converted into a rate-place representation in the auditory midbrain, where neurons are sensitive to low-frequency fluctuations. This neural-fluctuation code is vulnerable to sensorineural hearing loss (SNHL), because it depends upon saturation of IHCs and the interaction of cochlear amplification with IHC transduction. We tested formant-frequency discrimination in listeners with normal hearing or mild to moderate SNHL. The difficulty of the task was modulated by varying formant bandwidth. The relationship between spectral peaks and harmonic frequencies was also varied. Formant frequencies were 600 and 2000 Hz, in the range of first and second formants of many vowels. Results were compared to predictions made using model AN and midbrain neurons, with audiograms for the individual listeners included in the peripheral model. [Work supported by NIDCD-R01-001641]

2pPPb17. A guinea pig model of hidden hearing loss: Prelude to the development of a human model. Joann McGee (VA Loma Linda Healthcare System, VA Loma Linda Healthcare System, Loma Linda, CA 92357, mcgeej@umn.edu), Xiaohui Lin, Ashley Vazquez, Hongzhe Li, Marjorie R. Leek, Jonathan H. Venezia, Nicole Whittle, and Edward J. Walsh (VA Loma Linda Healthcare System, Loma Linda, CA)

Cochlear synaptopathy, otherwise known as hidden hearing loss, has been at least partially characterized in a number of mammalian species. Although well-established generally, the disorder is incompletely understood, particularly with regard to the extent of interspecies pathology associated with the condition. Furthermore, the extent to which recovery of lost function is achieved among species thus far studied is also incompletely understood. In this context, the existence of evidence suggesting that humans experience this form of synapse pathology calls for the development of a noninvasive protocol designed to further address the question. If confirmed, where on the disorder spectrum humans are positioned takes on a heightened sense of importance. To that end, a protocol that reliably identifies the disorder in a guinea pig model is under development and will, if successful, serve as the foundation for the development of an equivalent human protocol employing the same noninvasive electrophysiological strategy. In this report, preliminary findings from an ongoing study centered on a battery of electrophysiology and immunohistochemical studies supporting model development in guinea pigs will be reviewed with the goal of identifying response outcomes with diagnostic potential. [Work supported by the Department of Defense Award No. W81XWH-19-1-0862.]

Session 2pSA

Structural Acoustics and Vibration, Engineering Acoustics, and Physical Acoustics: Acoustic Metamaterials II

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Chair's Introduction—1:15

Invited Papers

1:20

2pSA1. Dynamics of quasiperiodic and quasicrystalline elastic metamaterials. Massimo Ruzzene (Dept. of Mech. Eng., Univ. of Colorado Boulder, P. M. Rady Dept. of Mech. Eng., 1111 Eng. Dr UCB 427, Boulder, CO 80309, massimo.ruzzene@colorado.edu), Matheus Rosa, and Yuning Guo (Dept. of Mech. Eng., Univ. of Colorado Boulder, Boulder, CO)

This talk describes recent progress on quasiperiodic and quasicrystalline metamaterial research done by the group. While the vast majority of metamaterials are based on periodic designs based on a repeating unit cell, quasiperiodic and quasicrystalline metamaterials break the periodicity paradigm and open new interesting possibilities. The first category is based on modulations of the properties of inclusions of otherwise periodic lattices, such as point masses, springs, resonators, or stiffeners, that are given by a deterministic non-periodic pattern. In the second case, quasicrystalline materials are based on rotational symmetries which are forbidden in periodic materials, such as 5,7,8, and 10-fold rotational symmetries. The investigation of the dynamics of such materials reveals intriguing properties such as topological bandgaps and localized vibration modes that can be manipulated by a few key parameters, and wave directionalities enabled by higher order rotational symmetries that expand the behavior known to be possible in periodic metamaterials. The talk will summarize the progress done in the area, including a large number of numerical and experimental investigations.

1:40

2pSA2. Controlling cylindrical waves using effective phononic crystals. Kathryn Matlack (Univ. of Illinois at Urbana-Champaign, 1206 W Green St., Urbana, IL 61801, kmatlack@illinois.edu) and Ignacio Arretche (Univ. of Illinois at Urbana-Champaign, Urbana, IL)

Phononic crystals (PCs) are periodic media that can exhibit novel wave propagation behaviors. The analysis of PCs relies on the Bloch theorem, and the application of which implies that certain canonical behaviors can be supported, e.g., band gaps. The Bloch theorem assumes plane wave propagation; however, this is not always the case, e.g., for cylindrical or spherical wave fronts. Here, we redefine material properties in PCs in the presence of cylindrically propagating waves, such that the resulting wave equations contain periodic coefficients. In this sense, an equivalent system can be defined on which Bloch theorem can be applied to predict its wave response. This talk will discuss our recent work on *effective phononic crystals*, which are PCs that are not geometrically periodic but whose material properties result in periodic coefficients in the wave equation. We demonstrate these concepts using radially-propagating torsional waves and use finite element method models to solve for material properties that result in Bragg scattering-based band gaps, locally resonant band gaps, and topologically protected interface modes. Finally, we conduct experimental validations on 3D printed samples fabricated with multi-material polyjet printing. This work has applications to mitigating damaging torsional vibrations in rotating machinery such as turbines, compressors, and engines.

2:00

2pSA3. Spatiotemporally modulated reflective acoustic metasurfaces. Janghoon Kang (Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX) and Michael R. Haberman (Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, haberman@utexas.edu)

Nonreciprocal acoustic wave propagation has been the topic of intense study for nearly a decade with the objective of understanding physical requirements to increase control over propagating acoustic waves [Nassar *et al*., Nat. Rev. Mater. **5**(9), 667–685 (2020)]. One approach to achieve non-reciprocity is deterministic spatiotemporal modulation of the material properties of the medium in which waves propagate. Achieving this modulation requires external sources of energy throughout the domain of interest, which presents a significant scientific and technical challenge. Furthermore, the non-reciprocal nature of the acoustic response is not guaranteed for finite domains and finding parameter sets of modulation amplitude, frequency, and wavenumber that lead to significant non-reciprocal behavior is non-trivial [Goldsberry *et al*., Phys. Rev. B **102**(1), 014312 (2020)]. Input impedance modulation of finite acoustic metasurfaces (AMS) presents a more tractable technical challenge. This work considers semi-analytical and numerical modeling of modulated reflective AMS. We present parametric studies of the static input impedance profiles and modulation functions as a linear, time-varying boundary condition of an acoustic half-space. We consider static admittance profiles that are uniform, discontinuous, and continuously varying and how this may influence the effectiveness of spatiotemporal modulation in achieving performance objectives such as acoustic diffusivity.

Contributed Papers

2:20

2pSA4. Elastic bilayer phononic crystal. Yuanchen Deng (Graduate Program in Acoust., The Penn State Univ., 105 Gala Dr., 201, State College, PA 16801, ypd5099@psu.edu), Mourad Oudich, and Yun Jing (Graduate Program in Acoust., The Penn State Univ., State College, PA)

Inspired by twisted bilayer graphene, we present an elastic bilayer consisting of a plate with two honeycomb lattices of mechanical resonators on both sides of the plate. Surface acoustic waves (SAW), which can interact depending on the thickness of the plate, are supported by each side of the plate. Beyond twisting a lattice with respect to the other, the coupling strengths between the two sides can be controlled via the plate thickness, which strongly affects the dispersion of elastic waves in the bilayer structure. The dispersion relationship of the SAW is theoretically characterized by calculating the band structure for the cases of AA and AB stacking configurations, as well as for special cases of twisting angles that produce sublattices with even and odd symmetries. Furthermore, we explore the topological characteristics of the bands and uncover a possible mechanism of creating topological elastic Valley states. The proposed bilayer plate could constitute a promising platform for manipulating mechanical waves and exploring quantum analog phenomena which could open routes toward innovative mechanical and optomechanical devices at the microscale, for instance.

2:35

2pSA5. Metamaterials you can talk to: Speech recognition with elastic neural networks. Tena Dubcek (ETH Zurich, Zurich, Zurich, Switzerland), Daniel Moreno-Garcia, Luis Guillermo Villanueva (EPFL, Lausanne, Switzerland), Dirk-Jan van Manen, Johan Robertsson (ETH Zurich, Zurich, Switzerland), and Marc Serra Garcia (AMOLF, Carolina Macgillavrylaan 2268, Amsterdam, Noord Holland 1098XK, Netherlands, m.serragarcia@amolf.nl)

To detect spoken commands, smart devices (for example, a speaker with Alexa or Siri) continuously convert acoustic waves to electronic signals, translate them into the digital domain, and analyze them in a signal processor. Each of these steps constantly consumes energy, imposing the need for tethered operation or large batteries. We propose to solve this problem using elastic neural networks, metamaterials consisting of arrays of coupled (potentially nonlinear) resonators. The frequencies and couplings of the resonators are optimized to maximise the speech classification accuracy (energy transmitted when excited with one word but not another). Even in purely linear metastructures, we observe binary classification accuracies exceeding 90% for a large number of pairs of words. This is demonstrated on a dataset from a large and diverse group of speakers. To attain these results, we have developed refined modelling techniques involving localised oscillations and machine learning. A unique feature of metamaterial-based speech processing is that speech classification is entirely passive, requiring no external energy. This is possible due to the very low energy dissipation of elastic waves.

2:50–3:05 Break

3:05

2pSA6. Machine learning non-reciprocity of a passive linear waveguide with a local nonlinear, asymmetric gate. Chongan Wang (Univ. of Illinois at Urbana Champaign, 1206 W. Green St., MEL 1412, Urbana, IL 61801, chongan2@illinois.edu), Alireza Mojahed (MIT, Cambridge, MA), Sameh Tawfick, and Alexander F. Vakakis (Univ. of Illinois at Urbana Champaign, Urbana, IL)

In this work, we propose a new, simple, and highly effective passive phononic waveguide with controllable global non-reciprocity by means of local nonlinear and asymmetric elements. The nonlinearity and asymmetry in the waveguide are realized by means of a local nonlinear gate with a cubic stiffness nonlinearity that couples two detuned, dissipative gate oscillators. A wave transmits across the nonlinear gate by applying a harmonic excitation. The transmitted wave is either monochromatic or strongly modulated (SMR) and triggers the global non-reciprocity. The objectives of this work are to study the effect of local nonlinearity and asymmetry on the global non-reciprocal acoustics and to optimize the non-reciprocal performance of the waveguide. We employ the complexification averaging method (CX-A) in the acoustics to predict the bifurcations that govern the non-reciprocal acoustics. However, a limitation of the single-frequency CX-A method is its failure to predict the SMRs. Alternatively, we train machine learning simulators in order to optimize the two performance measures, i.e., energy transmissibility and non-reciprocity. The trained machine learning model drastically saves the simulation time and allows the optimization of performances. Our results show how powerful machine learning approaches can be employed for designing and optimizing practical waveguides with tailored non-reciprocity features.

3:20

2pSA7. Incorporation of acoustic black hole stiffeners into composite airframes for reduction of noise radiation. Anna Moorhouse (Acoust., Penn State Univ., 201 Old Main, University Park, PA 16802, aqm6654@psu.edu), Micah Shepherd (The Penn State Univ., State College, PA), and Benjamin S. Beck (Eng. Acoust., Penn State Appl. Res. Lab, State College, PA)

Stiffened composite panels are commonly used in aerospace structures, because they are lightweight, while maintaining a high load-bearing ability. However, their high stiffness-to-mass ratio makes them efficient noise radiators. In rotorcraft cabins made with composite panels, for example, the internal noise levels can be quite high such that pilot and passenger communication and comfort are disrupted. This has led to a need for innovative noise reduction strategies for composite rotorcraft panels. A specialized

stiffener, which incorporates the acoustic black hole (ABH) effect into the cross section, is proposed to improve the damping of stiffened composite panels. By incorporating the damping concept into the stiffeners, the panel's radiated noise can be reduced while maintaining the weight advantages and panel strength. To determine the advantages and trade-offs of this concept, numerical models have been developed and incorporated into an optimization scheme. Computational studies reveal promising results from the optimized ABH stiffeners as compared to a baseline panel with traditional stiffeners.

3:35

2pSA8. Acoustic analog of twisted bilayer graphene. Steven R. Craig (Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., NW, Atlanta 30318, Georgia, scraig32@gatech.edu), Zhenglu Li, Jiawei Ruan, Steven G. Louie (Phys., Univ. of California at Berkeley, Berkeley, CA), and Chengzhi Shi (GWW School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

The emergence of twistronics in bilayer graphene has inspired the creation of new phononic structures that translate quantum effects into macro systems. Here, we introduce an acoustic analog of twisted bilayer graphene that is built with 3D-printed star arrays confined in a two-dimensional acoustic waveguide. The lattices on the top and bottom of the waveguide are coupled by spoof surface acoustic waves. Like its quantum counterpart, the twisting angle of the structure influences wave propagation within the system and its resulting band structures. In analytical models, full-wave simulations, and experiments, we observe mode localization at magic twisting angle that hosts the flat bands. We also study other twisted bilayers with different twisting angles to compare with the magic-angle results. The observation of unusual twistronics effects in acoustic systems demonstrates the potential to identify new quantum materials with simplified acoustic models.

3:50

2pSA9. Optimization methods in design of locally resonant metamaterials for noise and vibration mitigation. Klara Juros (AGH Univ. of Sci. and Technol., Mickiewicza 30, Cracow 30-059, Poland, juros@agh.edu.pl), Aleksander Kras (Silencions Sp. z o. o., Wroclaw, Poland), and Tadeusz Kamisinski (AGH Univ. of Sci. and Technol., Cracow, Poland)

In recent years, acoustic metamaterials are broadly investigated especially for noise and vibration mitigation. For this purpose, the best-suited option are locally resonant metamaterials (LRS). By creation of a band gap effect in flexural wave propagation in structure, improve its Sound Transmission Loss (STL). The effectiveness of the structure in STL and vibration mitigation depends on several different parameters for example mass of the base structure to LRS mass ratio distances between resonators or its geometry. Many different shapes of LRS can lead to band gap for selected frequency range but with many different results when it comes to STL. Most of the solutions presented in recent years were based on simple beam resonators tuned to a selected frequency, and barely ever the geometrical optimization process of the resonator's shape was considered. This project investigates optimization methods of the LRS presented in the literature, considering their efficiency and geometrical shape. Several algorithms were selected and combined with a numerical simulation process to obtain a solution with optimized mass and STL. Measurements for selected optimized structures are compared with simulation results and discussed in detail.

4:05

2pSA10. Metamaterials to control both longitudinal and transverse waves simultaneously. Pravinkumar R. Ghodake (Mech. Eng., Indian Inst. of Technol., Bombay, Dept. of Mech. Eng., Indian Inst. of Technol. Bombay, Mumbai, Maharashtra 400076, India, mech7pkumar@gmail.com)

Mechanical metamaterials are used in various applications to control bulk wave propagation through elastic materials. Controlling both the longitudinal and transverse waves at the same time by designing a single phononic device is challenging through theoretical, computational, and experimental perspectives. Suppression of system generated second harmonics due to interaction of monochromatic longitudinal wave and during one-way two-wave mixing of two longitudinal waves by designing effective phononic material is demonstrated in previous work by the author [Ghodake, J. Acoust. Soc. Am. 150, A149 (2021)]. In this talk, the design of linear periodic metamaterials which can control both the longitudinal and transverse waves, their second harmonics during propagation of monochromatic waves, and one-way two-wave mixing between longitudinal and transverse waves, is discussed by solving inverse problems using finite element analysis, effective boundary conditions, and optimization techniques. Along with the widths of periodic elastic materials, the number of repeated periodic cells (N) in a phononic lattice is defined as a design parameter. Different strong and weak contains are implemented so that in every iteration either during or after the iteration N will remain integer. A gradient-free optimization algorithm is used during constrained optimization. Strong constraint solves the inverse problem in a few minutes.

Session 2pSC

Speech Communication: Instrumentation and Method for Speech Analysis (Poster Session)

Isabelle Marcoux, Chair
Linguistics, UQAM - Universite du Quebec a Montreal, 405 Rue Sainte-Catherine Est, Montreal, H2L 2C4, Canada

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

Contributed Papers

2pSC1. Mechanical models as ground truth for vowel resonance analysis. Christine H. Shadle (Haskins Labs., 300 George St. Ste. 900, New Haven, CT 06511, shadle@haskins.yale.edu), Wei-Rong Chen (Haskins Labs., New Haven, CT), Sean A. Fulop (Linguist, California State Univ. Fresno, Fresno, CA), and D. H. Whalen (Haskins Labs., New York, NY)

It has long been established that LPC analysis results in formant estimates that are not accurate representations of the resonances; they are biased towards the nearest harmonic, and this bias worsens as F0 rises to 200 Hz or more. Manual measurement of formants with the reassigned spectrogram (RS) has been shown to be more accurate, but the ground truth is needed in order to test automatic measurement methods. Here, "vocal tract" models were 3D-printed to allow resonances to be excited replicably. Using the principle of acoustic reciprocity, source signals were played over an external loudspeaker, and the output filtered by the model was recorded via a microphone located in the model's "glottis." A white noise source signal was used to determine the resonances. Sawtooth and impulse train source signals (F0s 83–400 Hz) were filtered by the physical model and then analyzed with both LPC and RS. LPC formants were biased towards the nearest harmonic, resulting in errors from 10 Hz at low F0 to 250 at 300 Hz. RS errors ranged from 0 to 20 Hz and were not correlated with F0. These results indicate physical models are useful for creating known resonances for validating accuracy in our measurements.

2pSC2. Acoustical testing of face coverings using an artificial acoustic head with speech audio. Laura Ruhala (Mech. Eng., Kennesaw State Univ., 840 Polytechnic State University, Rm. Q319, MD 9075, Marietta, GA 30060, lruhala@kennesaw.edu), Richard J. Ruhala, and Danny Hernandez-Borjas (Mech. Eng., Kennesaw State Univ., Marietta, GA)

The COVID-19 pandemic has caused many communities to require or at least recommend the wearing of face coverings, masks, and/or shields inside indoor public spaces to reduce the transmission rate of the virus. Several types of face coverings are studied, including standardized N95 and KN95, assorted cloth, and clear plastic face shields. They are evaluated on an acoustical head and torso simulator (HATS) setup in a classroom with two different locations for monitoring via a sound level meter. The HATS is used as a controlled and repeatable artificial voice or sound source, which reproduces the ITU-T artificial speech signals as well as playback of real speech signals for American English, male and female voices. The signals are recorded with a sound level meter at a distance of 2 and 6 m between source and receiver in a classroom environment. These signals evaluated in time and frequency domains to better understand the acoustical difference between the no-mask and various masks conditions. These are compared with the same classroom environment but using white noise test signals.

2pSC3. An evaluation of methods: Measuring American /aɪ/ raising. Alyssa Strickler (Linguist, Univ. of Colorado, Boulder, CO 80309, alyssa.strickler@colorado.edu)

/aɪ/ raising (also called American raising, or the raising of /aɪ/ to /ʌɪ/ preceding voiceless consonants) occurs in many US dialects including in Fort Wayne, Indiana [Berkson *et al.* (2017)]. American raising has often been measured using the maximum of F1 in the diphthong as the nucleus of /a/ [e.g., Fruehwald (2016); Hualde *et al.* (2019)] or a timepoint at approximately 33% of the vowel as the midpoint of the first vowel of the diphthong [Berkson *et al.* (2017)]. Other descriptions of American raising include the measurements of the offglide as well, often approximated using the F2 maximum (e.g., Moreton and Thomas, 2007). There are other ways to measure diphthongs, such as General Additive Models that capture the dynamic nature of a diphthong rather than reducing it to 1–2 points (Thomas and Mielke, to appear). The current study compares the results of using the different methods described to analyze subjects with varying degrees of diphthong raising with data collected in 2021 from college-aged speakers in Fort Wayne. With the goal of assessing the level of detail required to capture the /aɪ/ raising process in its incipient form, the results compare more complex analyses to simpler analyses on the same data.

2pSC4. Evaluating the accuracy of forced alignment across Mandarin varieties. Suyuan Liu (Linguist, Univ. of BC, 2613 West Mall, Vancouver, BC, Canada, suyuan.liu@ubc.ca) and Márton Sóskuthy (Linguist, Univ. of BC, Vancouver, BC, Canada)

Forced alignment is widely used in phonetics to align transcripts with acoustic signals. These tools are trained on specific language varieties; it is unclear if they generalize to others. Previous research on English by MacKenzie and Turton (2020) finds good agreement between automated and human alignments for varieties that differ from the training variety. Such evaluation has only been carried out for English. We evaluate the level of human-aligner agreement on four Mandarin varieties (Canto, Shanghai, Beijing, and Tianjin). For each variety, two recordings from the HUB5 Corpus (LDC 1998) were aligned manually and by the Montreal Forced Aligner [McAuliffe *et al.* (2017)] using acoustic models trained on Beijing, Wuhan, and Hekou Mandarin [Schultz (2002)]. We find strong agreement between human and machine-aligned phone boundaries, with 17 ms as the median onset displacement. A mixed model identifies little variation across varieties or according to speech rate, but significant interindividual variation. Notably, despite the generally close agreement between the machine and human alignments, for two of the speakers, more than 10% of the alignments are displaced by over 100 ms. In sum, the Mandarin forced-aligner yields reliable alignments for out-of-training varieties, but manual checking of the results is still crucial.

1:00

2pSC5. A spatially-aware dialogue system for immersive classrooms. Albert Chang (School of Architecture, Rensselaer Polytechnic Inst., Troy, NY), Mei Si (Cognit. Sci., Rensselaer Polytechnic Inst., Troy, NY), Samuel Chabot, Jonathan Mathews (School of Architecture, Rensselaer Polytechnic Inst., Troy, NY), Tomek Strzalkowski (Cognit. Sci., Rensselaer Polytechnic Inst., Troy, NY), and Jonas Braasch (School of Architecture, Rensselaer Polytechnic Inst., School of Architecture, 110 8th St., Troy, NY 12180, braasj@rpi.edu)

Dialogue systems have become a popular research medium as recent advances in task-oriented and open-domain systems combined with deep learning technologies have increased the potential for practical applications across many disciplines. One such vein of applications involves multi-modal dialogue systems deployed in interactive spaces that seek to provide an immersive experience for participants. This project proposes a combination of spatial awareness with a multi-modal, immersive dialogue system as a potential interactive medium to provide an additional layer of immersion. The system employs an array of audio/visual sensors that tracks participants within the interactive space. It responds contextually depending on the application and information domain, for example, by displaying and sonifying conversational agents at accurate spatial locations. The current application of this system involves Mandarin language learning in which the system will act as both a learning medium and conversation augmentation system to provide students with an immersive environment to learn a language and provide real-time feedback during the learning process. This project aims to provide insight into interactive spaces for education and general conversation applications, and demonstrate the capabilities of combining spatial awareness with a multi-modal dialogue system. [Work supported by NSF IIS-1909229, IBM GATOR, and CISL.]

2pSC6. The onset of voice onset time: Measuring the emergent timing of laryngeal and oral gestures in early speech development. Gordon Ramsay (Dept. of Pediatrics, Emory Univ., Marcus Autism Ctr., 1920 Briarcliff Rd NE, Atlanta, GA 30329, Georgia, gordon.ramsay@emory.edu)

Voicing contrasts in stop consonants are predominantly signaled by manipulation of voice onset time (VOT), the relative timing of oral and laryngeal gestures responsible for the release of the oral constriction and the onset of phonation. VOT is usually measured from the acoustic signal and defined by the time between the stop consonant burst and the following glottal period. Although this works well in adult speech, the same measure can be problematic when applied to child vocalizations. Infant productions of closant-vocant sequences perceived as consonant-vowel syllables by adults often do not exhibit full closure and are voiced throughout. Traditional measures of VOT cannot then be employed, even though evidence from the acoustic signal can still be used to deduce the presence and timing of precursors of the oral and laryngeal gestures from which adult voicing contrasts later emerge. This study illustrates these issues using data derived from home audio recordings made monthly from birth to three years as part of a large-scale, longitudinal study of infant vocal development. New methods for measuring VOT are proposed, using the relative amplitude, frequency and phase of the harmonics in the acoustic signal to infer the timing of underlying articulatory gestures.

2pSC7. Prosodyad: A Praat plugin to assess prosody in dyadic interactions. José J. Atria (none, London, United Kingdom), Jane Brusilovsky (Modern Human Anatomy, Univ. of Colorado Anschutz, 5245 S Jebel Way, Centennial, CO 80015, jane.brusilovsky@cuanschutz.edu), Kevin Cohen (Computational Bioscience, Univ. of Colorado Anschutz, Aurora, CO), and Peter Pressman (Neurology, Univ. of Colorado Anschutz, Aurora, CO)

Affective prosodic features, such as syllabification and fundamental frequency, are important aspects of human communication. Praat software has enabled detailed analysis of prosody but is usually used to analyze small snippets of task-based speech. To study prosody in more ecologically valid ten-minute conversations, we designed Prosodyad, a Praat script designed to extract several emotionally salient prosodic features from conversations between two people. Prosodyad is designed to overcome several technical obstacles inherent to multiple speakers by using human labeling on Praat "textgrids" to analyze signals only from noise-free, non-overlapping speech. Prosodyad works over lists of matched textgrids and audio files, extracting salient features such as F_0 floor, minimum, maximum, and standard deviation, mean and standard deviation of F_1 and F_2, HF500, and intensity mean, min, max, and SD, and a syllable count for each unbroken interval. Floor and ceiling for pitch, windowing options for the analysis, as well as max formant as well as thresholds for ASD can be set by the user. We illustrate potential research and clinical applications and give examples of analyses relevant to ten-minute conversations among people with dementia. The software is available on Github at https://github.com/jjatria/plugin$_{\text{prosodyad}}$.

2pSC8. Investigating the acoustic fidelity of remote recording methods. Emily P. Ahn, William Bowers, Ella Deaton, Gina-Anne Levow, Sara Ng, Marina Oganyan, Robert Squizzero (Univ. of Washington, Seattle, WA), and Richard A. Wright (Univ. of Washington, University of Washington, Dept. of Linguist, Box 352425, Seattle, WA 98195-2425, rawright@uw.edu)

Our study tests the acoustic fidelity of remote recordings, using a large variety of stimuli and recording environments. With standard recording environments not available due to COVID-19, more studies investigate remote recordings for acoustic analyses [e.g., Guan and Li (2021); Freeman and De Decker (2021)]. High fidelity remote recordings also support crucial uses like reaching isolated populations and more speakers. A 188-word list was constructed from each English consonant followed by each vowel. Words recorded by a male and female speaker in a sound attenuated booth were input for test recordings. Stimuli were recorded on six devices across five operating systems, four tele-conferencing platforms, and three browsers, using internal and external microphones. Acoustic analysis investigates the impact of these recording configurations on features including pitch, relative intensity, vowel formant measures, spectral moments, spectral tilt, spectral rolloff, and Mel Frequency Cepstral Coefficients. Preliminary analyses found durational differences between original and test-recorded stimuli, posing challenges for automatic segmentation and alignment. Aperiodic noise was also introduced. We hypothesize further distortions in other measures. The findings from this study will allow us to identify acoustic measures which are robust across varied remote recording conditions and to highlight configurations least likely to introduce problematic artifacts.

2pSC9. A digital pattern playback system implemented in Python. Hahn Koo (San Jose State Univ., One Washington Square, San Jose, CA 95192-0093, hahn.koo@sjsu.edu)

Pattern playback systems were instrumental in speech perception research [e.g., Cooper *et al*. (1951)] and can be valuable for pedagogical purposes [e.g., Arai *et al*. (2006)]. They would be utilized further if one could integrate them with other speech processing software written in a common programming language. In response, I present an open-source digital pattern playback system implemented in the Python programming language. The software allows the user to provide an image of a magnitude spectrogram as input by either selecting an image file (e.g., PNG, JPG) or drawing one directly on a blank canvas using a pointing device (e.g., computer mouse, stylus, fingertip). It first translates pixel values of the image to an array of magnitude spectral coefficients and then applies the inverse short-time Fourier transform assuming zero phase to convert the array into a waveform. Users can readily manipulate basic parameters of conversion (e.g., sampling rate, frame length) and augment the process by utilizing various signal processing methods available in Python libraries such as SciPy and librosa. The source code is available for download and will be maintained on the author's GitHub repository and personal website.

2pSC10. Vocal aesthetics—Theoretical and methodological considerations. Piotr Sorokowski (Univ. of Wroclaw, Dawida 1, Wroclaw 50-529, Poland, sorokowskipiotr@yahoo.co.uk), Jerzy Luty, Agata Groyecka-Bernard, and Katarzyna Pisanski (Univ. of Wroclaw, Wrocław, Poland)

Research on the influence of human voice on perceived attractiveness and other characteristics has a long tradition in psychology, aesthetics, and neuro-cognitive science. Here, we would like to present the perspectives,

methodologies and goals of voice studies within these disciplines. Then, we will focus on the most common voice studies in psychology (i.e., evaluation of different characteristics based on the voice samples). Hundreds of studies have measured properties of the human voice to understand what vocal features tell about the speakers and how they influence their social perception. However, there is surprisingly little consensus about their methodology, specifically in terms of the verbal content and duration of voice recordings (i.e., speech stimulus types). While in some studies researchers record only vowel sounds or single words, others use sentences, longer, scripted statements or fragments of spontaneous conversations. The duration and verbal content of voice recordings could affect key nonverbal properties of speech, and therefore, in our project, we assessed the effect of speech stimulus type on voice perception in playback experiments. In our presentation, will describe the results of our previous studies and based on these we will also present our recommendations on future psychological experiments. [Work supported by NCN 2016/23/B/HS6/00771]

2pSC11. Benefits of app-based speech in noise screener for children. Darchayla Lewis (Communicative Sci. and Disord., Hampton Univ., 200 William R. Harvey Way, Hampton, VA 23668, darcyl99@yahoo.com), Jessica Sullivan (Communicative Sci. and Disord., Hampton Univ., Hampton, VA), Julia Irwin (Psych., Southern Connecticut State Univ., New Haven, CT), and Peggy Nelson (Ctr. for Applied/Translational Sensory Sci., Univ. of Minnesota, Minneapolis, MN)

Current hearing screening methods focus on assessing a child's ability to hear tones in quiet. However, this isn't reflective of real-world classroom situations. Our proposed screening test uses speech in quiet and noise to evaluate a child's auditory comprehension and possible auditory processing abilities. In this study, we investigate whether a speech in noise screener, Hearing Assessment in Response to Noise Screener (HeARS), is sensitive to detecting possible listening difficulties. We measured responses to a four-choice task with adapting signal to noise ratios based on individual performance. Following an auditory stimulus, the child is presented with four choices that are visual images of the spoken words. The child selects the correct image they hear, and based on their performance the signal to noise ratios increase or decrease in difficulty. Scores are reported as percent correct. We hypothesized that results from the screener would capture differences between speech understanding in noise and quiet conditions. In addition, a speech in noise screener may be more sensitive to deficits in auditory comprehension than traditional screening methods. Preliminary data suggest that the accuracy of understanding words in background noise increases with age. Results have implications for identifying children with auditory comprehension and processing deficits sooner, especially in underserved populations.

2pSC12. Comparison of one-dimensional and three-dimensional glottal flow models in left-right asymmetric vocal fold conditions. Tsukasa Yoshinaga (Toyohashi Univ. of Technol., 1-1 Tempaku, Hibarigaoka, Toyohashi 441-8580, Japan, yoshinaga@me.tut.ac.jp), Zhaoyan Zhang (UCLA School of Medicine, Los Angeles, CA), and Akiyoshi Iida (Toyohashi Univ. of Technol., Toyohashi, Japan)

While the glottal flow is often simplified as one-dimensional (1D) in computational models of phonation to reduce computational costs, the 1D flow model has not been validated in left-right asymmetric vocal fold conditions, as often occur in both normal and pathological voice production. It is unclear to what extent the 1D model approximates the effect of three-dimensional (3D) flow phenomena and fluid-structure interaction. In this study, we performed 1D and 3D flow simulations coupled with the two-mass vocal fold model and compared vocal fold vibration patterns at different degrees of left-right stiffness asymmetry. The flow and acoustic fields in 3D were predicted by solving the compressible Navier-Stokes equations using the volume penalization method and considering the moving vocal fold wall as an immersed boundary. The results showed that vocal fold vibration amplitudes and left-right phase differences in the 3D flow were predicted by the 1D flow model under conditions of small left-right asymmetry, while vocal fold vibration amplitudes were underestimated at conditions of large left-right asymmetry. This indicates that 1D flow models may be sufficient in modeling phonation under left-right asymmetric conditions, although the performance can be further improved by more accurately predicting air pressure on vocal fold surface.

2pSC13. Tagged-MRI to audio synthesis with a pairwise heterogeneous deep translator. Xiaofeng Liu (Gordon Ctr. for Medical Imaging, MGH and Harvard Med. School, 5 Fruit St., Thier 304, Boston, MA 02114, xliu61@mgh.harvard.edu), Fangxu Xing (Radiology, Harvard Med. School, Boston, MA), Maureen Stone (Univ. of Maryland Dental School, Baltimore, MD), Jerry L. Prince (Johns Hopkins Univ., Baltimore, MD), Jangwon Kim (Amazon, Los Angles, CA), Georges El Fakhri, and Jonghye Woo (Gordon Ctr. for Medical Imaging, MGH and Harvard Med. School, Boston, MA)

Identifying the underlying relationship between visual movements in tagged-MRI and intelligible speech is a vital problem to better understand speech production in health and disease. Due to their heterogeneous representations, however, direct mapping between the two modalities is challenging. We develop a deep learning framework that can synthesize a sequence of tagged MRI data to its corresponding mel-spectrogram, and then convert back into the audio waveform. Our network adopts a parallel encoder-decoder structure to take as input a pair of tagged MRI sequences. The 3D CNN-based encoders learn to extract the feature of spatiotemporally varying motions. The decoder then learns to generate the corresponding spectrograms conditioned on the latent space feature. For the pair of the same utterance, we further make the latent space feature as close as possible with the Kullback-Leibler divergence. To demonstrate the performance of our framework, we used a leave-one-out evaluation strategy on a total of 63 tagged MRI sequences from two utterances, including 43 "ageese" and 20 "asouk." Our framework enabled the generation of clear audio given a sequence of tagged MRI unseen in training, which could potentially aid in better understanding speech production and improving treatment strategies for patients with speech-related disorders.

2pSC14. Learnability of ultrasound tongue imaging devices in speech-language pathology. Isabelle Marcoux (Linguist, UQAM - Universite du Quebec a Montreal, 405 Rue Sainte-Catherine Est, Montreal, QC H2L 2C4, Canada, marcoux.isabelle.8@courrier.uqam.ca), Lucie Ménard (Linguist, UQAM - Universite du Quebec a Montreal, Montréal, QC, Canada), and Catherine Laporte (Ecole de Technologie Superieure, Montreal, QC, Canada)

Ultrasound tongue imaging has shown potential for speech-language pathologists (SLPs) to evaluate and treat persistent articulatory disorders. However, SLPs typically begin with low to no familiarity with ultrasound. Thus, this study investigated an important aspect of ultrasound device usability: learnability for SLPs. The project was funded by an NSERC Engage grant in partnership with Clarius Mobile Health. Twelve SLPs learned to use two ultrasound devices: a wireless device, provided by our partner Clarius, and a traditional device, to record clips of their or the experimenter's tongue. They then completed a questionnaire (French translation of the System Usability Scale (Brooke, 1996)). Two expert judges evaluated the clips recorded by the SLPs for the choice of settings and the positioning of the probe. Results of the SUS show a better usability for the wireless device than the traditional device. SLPs appreciated the user-friendly tablet interface, possibly because they are already used to interacting with tablets. Clips analyses show a better choice of settings by the SLPs with the wireless device. The positioning of the probe, however, was better with the traditional device, possibly due to its smaller probe. In conclusion, US seems to have a good potential of usability in speech-language pathology, provided that SLPs receive training for US image interpretation. A traditional US device may require a longer learning period than a wireless model with tablet interface.

2pSC15. An experimental model for the study of tracheoesophageal phonation. André Miazaki da Costa Tourinho (Laboratório de Vibrações e Acústica, Universidade Federal de Santa Catarina, Laboratório de Vibrações e Acústica, Universidade Federal de Santa Catarina, Campus Trindade, Florianópolis, Santa Catarina 88040-900, Brazil, andre.miazaki@posgrad.ufsc.br) and Andrey R. da Silva (Laboratório de Vibrações e Acústica, Universidade Federal de Santa Catarina, Florianópolis, Santa Catarina, Brazil)

Tracheoesophageal speech is the most widely used method of speech rehabilitation for those who have undergone a total laryngectomy. Despite its high success rate, the interplay between different factors may inhibit voice production. Among these factors, the amount of muscle contraction in the pharyngoesophageal segment is the most significant one. The present work is aimed at studying the effect of muscle contraction on tracheoesophageal voice production. An experimental model has been developed, in which a silicone tube, acting as the pharyngoesophageal segment, is connected at both ends to rigid tubes representing the pharynx and the esophagus. The effect of the musculature of the pharyngoesophageal segment has been included in the model by two different components. The first one, modeling the tendency of the muscle layer to close the pharyngoesophageal segment, has been taken into account by placing the tube inside a pressurized chamber. The second, modeling the stretching caused by longitudinal muscle fibers, has been included by means of a sliding device in one of the terminations of the silicone tube. The necessary parameters for self-sustained oscillations to occur were measured and compared with predictions made by a mathematical model [Tourinho *et al.*, *J. Acoust. Soc. Am.* **149**, 1979–1988 (2021)].

2pSC16. Communication behavior observation. Stefan Klockgether (R&D, Sonova AG, Laubisrütistrasse 28, Stäfa, Zürich 8712, Switzerland, stefan.klockgether@sonova.com), Conradin Kleinstein, Michael Schneeberger (Eastern Switzerland Univ. of Appl. Sci., Rapperswil, St. Gallen, Switzerland), and R. Peter Derleth (R&D, Sonova AG, Stäfa, Zürich, Switzerland)

Communication is one of the most important aspects of human society. 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 unconsciously or as a mixture of both. To investigate communication behavior, real two-person conversations were observed in the Sonova Real Life Lab. Within the lab, participants were allowed to move around freely in an acoustic scene. The head position and orientation of both participants were monitored with a motion capturing system and the voices of both participants were recorded with wireless headset microphones. The voice intensity and spectral content were analyzed from the speech recordings to assess vocal effort and detect Lombard speech. The audio setup of the Real Life Lab was used to playback an acoustic background scene, which could be directly manipulated in level and spectral content. This contribution shows data from pilot experiments in which the participants had to communicate in acoustically controlled background scenes. The data show the characteristics of and interactions between different strategies humans use to successfully communicate in challenging acoustic environments. Learnings for future experiments are discussed.

Session 2pSP

Signal Processing in Acoustics: General Topics in Signal Processing I

Ryan L. Harne, Cochair
The Pennsylvania State University, 2482 Raven Hollow Rd., State College, PA 16801

Geoffrey F. Edelmann, Cochair
U. S. Naval Research Laboratory, 4555 Overlook Ave. SW, Code 7145, Washington, D.C. 20375

Chair's Introduction—1:15

Contributed Papers

1:20

2pSP1. A practical spatially oversampled array concept enabling reconfigurability. Stephen Trickey (Optical Sci. Div., U.S. Naval Res. Lab., 4555 Overlook Ave. SW, Code 5674, Washington, DC 20375, stephen.trickey@nrl.navy.mil), Kirk Daley (Optical Sci. Div., U.S. Naval Res. Lab., Washington, DC), Alvaro Bautista (Jacobs, Washington, DC), and Clay Kirkendall (Optical Sci. Div., U.S. Naval Res. Lab., Washington, DC)

Typically, acoustic arrays are cut for a desired performance characteristic where acoustic performance is balanced with physical and cost constraints. Once constructed, the sensor placement in the array is fixed in a permanent configuration. Adaptive processing techniques can be used to optimize the acoustic characteristics of the array but are bounded by the fixed sensor placement. Reconfigurable arrays, where the sensor locations and apertures can be dynamically adjusted, can offer improved acoustic performance and signal processing power savings for a given array aperture. This work presents a new concept for fiber optic hydrophone arrays that uses a highly spatially oversampled multiplexing strategy enabling a large degree of flexibility in sensing configuration. The ease of multiplexing optical hydrophones allows for implementing the oversampled strategy with no additional processing hardware required. A review of fiber optic acoustic sensing and the optical techniques required to implement high performance spatially oversampled arrays will be presented. [DISTRIBUTION STATEMENT A. Approved for public release. Distribution is unlimited.]

1:35

2pSP2. 3D printed deployable origami patterns at intermediate folding configurations for wave guiding applications. Haley Tholen, Lance Hyatt (The Penn State Univ., University Park, PA), Christopher Bentley (The Penn State Univ., 336 Reber Bldg., University Park, PA 16802, csb5532@psu.edu), and Ryan L. Harne (The Penn State Univ., State College, PA)

Origami-based engineering employs the use of complex reconfigurable structures for a variety of applications in science and engineering. Most origami-inspired structures are designed to be fully deployed, yet wave guiding arrays require deployment to stable intermediate folding configurations to focus waves. Furthermore, current research for origami-inspired wave guiding does not study kinetic and kinematic behavior during compaction and deployment. In this study, the force required to compact the Miura-ori unit cell with compliant joints is investigated for intermediate folding configurations. By starting at intermediate folding angles, the maximum force to reach a fully compact shape can be reduced by over an order of magnitude compared to starting at a flat state.

1:50

2pSP3. Analysis of turbulence and scattering effects on detecting elevated sources with a microphone array. Geoffrey H. Goldman (U.S. Army Res. Lab., 2800 Powder Mill Rd., Adelphi, MD 20783-1197, geoffrey.h.goldman.civ@mail.mil)

Sound waves propagating in urban environments are affected by turbulence, scattering from numerous structures, and atmospheric refraction. These effects can be incorporated into beamforming algorithms but at the cost of increased processing time and algorithm complexity. An analysis of incorporating refraction effects into the beamformers for a simplified scenario is performed on detection algorithms based on four generalized beamforming algorithms. Simulated results for beamforming algorithms implemented with no model mismatch error are compared to results for classical beamforming algorithms implemented with model mismatch errors generated by atmospheric refraction. Their relative performance is evaluated using the area under the receiver operating characteristic curve statistic calculated with Monte Carlo simulations. The results processed with data that include effects from turbulence and rough surface scattering show that there is no significant improvement in detection performance obtained by including refraction effects into the beamforming algorithms over a wide range of frequencies. The results indicate that complex beamforming algorithms are not required for many urban acoustic applications.

2:05

2pSP4. Partial field visualization using an advanced circular aperture, continuous-scan acoustic array. John McShane (ATA Eng., Inc., 13290 Evening Creek Dr. S, San Diego, CA 92128, john.mcshane@ata-e.com), Abe Lee, Parthiv Shah, and Peter Kerrian (ATA Eng., Inc., San Diego, CA)

ATA Engineering, Inc. designed, fabricated, and tested a novel circular aperture acoustic array for continuous-scan beamforming and acoustical holography applications. The array comprises an outer perimeter of stationary reference sensors coupled to a multi-arm inner rotating array that samples the entire aperture plane. The unique architecture allows the user to perform both fixed-receiver and continuous-scan beamforming with a single array with the latter method also enabling high-spatial-resolution partial field visualization. The presentation will show results from far- and near-field partial field visualizations over the rotating array circular aperture taken at different orientations. The presentation will also describe the use of the array to measure and model isolated sources, and experimentally verify their boundary element method-predicted sound fields in the presence of a scattering body.

2:20–2:35 Break

2:35

2pSP5. Denoising and deconvolving sperm whale data in the northern Gulf of Mexico. Kendal Leftwich (Phys., Univ. of New Orleans, 1021 Sci. Bldg. University of New Orleans, New Orleans, LA 70148, kmleftwi@uno.edu) and Juliette W. Ioup (Phys., Univ. of New Orleans, New Orleans, LA)

The Fourier wavelet based regularized deconvolution (ForWaRD) algorithm is a combination of Fourier deconvolution and wavelet denoising originally proposed by Neelamani *et al*. (2004) and modified by Herrera *et al*. (2006). The research described here uses the ForWaRD algorithm with recent broadband underwater acoustic data from sperm whales acquired in the northern Gulf of Mexico. Several modifications were required for the algorithm to yield the best results. Applying wavelet denoising and Fourier deconvolution allows smoothing of the data and separating a signal from the impulse response. Results indicate that the modification of using lease squares deconvolution instead of Weiner deconvolution improves the percent error in data reconstruction with our underwater acoustic data. Other modifications made to the type of wavelet thresholding as well as the wavelet choice for wavelet decomposition greatly improve the error in reconstruction of the signal. Results will be shown as well as an outline of the modifications.

2:50

2pSP6. Range estimation within a shallow-water waveguide via active and reactive acoustic intensity. Benjamin Cray (NUWC, 1176 Howell St., Newport, RI 02841-1708, benjamin.cray@navy.mil)

The proposed ranging algorithm examines low-order modal variations in vertical reactive (Q_z) and horizontal active (I_r) acoustic intensity and provides a *gated estimate* of the range from a receiver to a source. The receiver (measuring horizontal and vertical acoustic intensity) estimates range bins, for example, a source, which is confined within a range of 1 km to 2 km. Accurate initial range estimates provide key information for follow-on processors, such as tracking and depth classification algorithms. The technique relies on predictive broadband frequency characteristics between the superposition of low order trapped modes; modes which are restricted to the first few orders, for example, modes one through five. The binned range estimates can deliver, for example, initial values for higher level iterative and interpolative Detection, Classification and Localization (DCL) algorithms. Similarly, given an accurate opening estimate of a target's range, neural networks techniques are enhanced. Existing at-sea shallow-water (<300 m), acoustic intensity data are compared to theoretical predictions to assess the feasibility and performance of the proposed algorithm.

3:05

2pSP7. Deconvolution filter for parameter estimation of composite stress wave propagating in the wooden cylindrical structure. Yishi Lee (Metropolitan State Univ. of Denver, 1449 7th St., Denver, CO 80204, Denver, CO 80204, ylee24@msudenver.edu)

Through-transmission using the embedded waveguide technique in the cross-section plane of cylindrical structure produces radial and Rayleigh modes. These two modes propagate in the heart and shell regions of the plane, allowing a comprehensive characterization of the material. The significant spatial difference generates a distinct temporal separation between the two modes for large cylindrical structures, allowing direct mode characterization. For cylindrical structures with smaller diameters, the pronounced multi-path interference produces an erroneous energy response that hinders the accurate mode characterization. In this so-called multiple-input-single-output (MISO) system, the arrival time and amplitude estimations are an underdetermined problem. This work presents a novel decomposition filter design consisting of three steps. (1) Gabor's wavelet transforms the stress wave signal in the time-frequency domain; (2) The approximation of the inverse point-spread-function (PSF) response via a series of frequency polynomials generates a set of optimal coefficients; (3) Derive the inverse deconvolution kernel via the dual derivative operator from the obtained optimal coefficients in step 2. This presentation will demonstrate the algorithmic formulation with numerical and empirical validations. Its broader impact will enhance the parametric estimation of the diagnostic stress wave for material characterization.

3:20

2pSP8. In-water and in-air vehicle velocity estimation via harmonic and Doppler analysis. Jonah Singer (Mahomet Seymour High School, 302 W State St., Mahomet, IL 61853, jonahryansinger@gmail.com) and Eden Oelze (Mahomet Seymour High School, Mahomet, IL)

Remotely-controlled (RC) vehicles, such as RC cars, boats, planes, and drones, use high energy-density lithium polymer batteries that enable powerful brushless DC motors to propel them at remarkable velocities. In prior work, measurements of the acoustic emissions from such motors on RC cars have been processed to estimate vehicle velocity, based on a spectral analysis of the emissions, together with a parametric model for the acoustic emissions, relating them to motor speed and vehicle velocity. This work builds on prior models for the acoustic emissions of the DC motors to estimate the motor speeds for in-water and in-air craft, including RC boats and drones. Spectrograms of the acoustic recordings of the vehicles at moving at constant velocity provide sufficient harmonic structure to effectively measure the Doppler shift at closest point of proximity, enabling vehicle velocity estimates. These, in turn, enable calibration of the harmonic structure for motor speed estimation. Preliminary results demonstrate the correlation between the speed profile of the vehicle, acoustic harmonic structure, and Doppler shift.

Session 2pUW

Underwater Acoustics and Acoustical Oceanography: Collaborative Measurements in Underwater Acoustics: A Memorial Session for John R. Preston

David L. Bradley, Cochair
University of New Hampshire, Durham, NHY 03824

Dale D. Ellis, Cochair
Physics, Mount Allison University, 18 Hugh Allen Drive, Dartmouth, B2W 2K8, Canada

Paul C. Hines, Cochair
ECE, Dalhousie, 1360 Barrington Street, C367, PO Box 15000, Halifax, B3H 4R2, Canada

Chair's Introduction—1:00

Invited Papers

1:05

2pUW1. John Preston's contributions to ONR collaborative research efforts. Kyle M. Becker (Office of Naval Res., 875 N Randolph St., Arlington, VA 22203, kyle.becker1@navy.mil)

The Office of Naval Research (ONR) was established by public law in 1946 to plan, foster, and encourage scientific research in recognition of its paramount importance as related to the maintenance of future naval power. To fulfill its mission, the ONR model is to invest not only in ideas, but in people. It seeks to attract and retain high performing scientists and engineers to lead and collaborate on research that is recognized nationally and internationally for its quality and importance. Dr. John R. Preston was an exemplar of the ONR model. This presentation will feature the many and various contributions John has made to the scientific community through his ONR supported work. It will emphasize the scientific relationships and outcomes John fostered, both nationally and internationally, in academia and government by his championing development of the ONR Five Octave Research Array (FORA). Under John's stewardship, the FORA became a community asset that was used in a host of different experiments, deployed from a variety of different platforms, and operated in many different environments. John's unselfishness, dedication, and enthusiasm for working with an ever growing number of partners have made FORA a great success for both ONR and the community.

1:25

2pUW2. Pacific Echo: A deep ocean collaborative experiment. Ross Chapman (Univ. of Victoria, University of Victoria, 3800 Finnerty Rd., Victoria, BC V8P5C2, Canada, chapman@uvic.ca)

This paper summarizes experiments carried out using towed line arrays in the Pacific Echo sea trials. Although John Preston was not a participant, the experiments are examples consistent with his efforts in promoting collaborative research at sea with horizontal arrays. The overall goal of Pacific Echo was to study the impact of thin sediment ocean bottom environments on sound propagation in deep water. The objective of the experiments described here was to study the evolution of young oceanic crust. The hypothesis was that the sound speed of young basalt increased with age of the crust. Sound speed was inferred from low frequency measurements of the reflection coefficient versus grazing angle at sites of increasing distance from the deep ocean spreading center. The experiments introduced a novel design for measuring the reflection coefficients using two ships and small explosive charges. The signals at the array were spatially filtered to resolve the specular reflections as the ships opened range on set courses. The data provided estimates of both the compressional and shear wave speeds of the basalt. Results from the broadside reflection technique showed that sound speed increased with crustal age and were consistent with measurements obtained from conventional seismic reflection surveys.

Contributed Paper

1:45

2pUW3. Shallow water acoustic spatial coherence. Ashley Kamal (MIT Lincoln Lab., 244 Wood St., Lexington, MA 02421, ashley.kamal@ll.mit.edu) and Madeline Miller (MIT Lincoln Lab., Lexington, MA)

The understanding and ability to predict acoustic spatial coherence in dynamic ocean environments is limited, as it depends on many small scale physical processes. Development and validation of theoretical models requires the assessment of coherence in diverse ocean conditions. We compute spatial coherence measures for the SwellEx-96 experiment and evaluate the functional dependence of coherence on frequency in a shallow water environment. The acoustic signals are low-frequency multi-tones (50–400 Hz) transmitted from a ship-towed source and received on horizontal bottom-mounted arrays deployed across the 240-260m isobaths. Following the approach developed by Heaney (2011), coherence measures employed are the "power factor" and "eigenvalue ratio" of the eigenvalues of a cross-spectral density matrix. Advantageously, these methods account for both amplitude and phase, do not require a signal model, and were shown to have low sensitivity to array element position error. We assess whether spatial coherence calculated from the SwellEx-96 data is functionally related to frequency and determine the effect of range on the spatial coherence estimate. Results from this analysis are compared to existing coherence length data estimates and theoretical models.

Invited Papers

2:00

2pUW4. Use of towed array reverberation data for rapid environmental assessment. Dale D. Ellis (Phys., Mount Allison Univ., 18 Hugh Allen Dr., Dartmouth, NS B2W 2K8, Canada, daledellis@gmail.com)

Towed array reverberation beam time series can provide information about the underwater acoustic environment by providing a snapshot of the scattering in both range and bearing. By mapping time into range and beam angle into azimuth, John Preston pioneered the use of polar plots to survey an area [Preston *et al*., J. Acoust. Soc. Am., 87,119–134 (1991)]. When the polar plot is superimposed on the bathymetry of an area, a scattering map can be made. High scattering is generally associated with bottom features. With additional analysis and modeling, bottom loss and scattering strengths can be obtained [Preston and Ellis, J. Marine Systems 78, S359–S371 (2009)]. This talk emphasizes results from the rapid environmental assessment (REA) Rapid Response exercises 1996–1998, a multi-nation collaboration organized by NATO MILOC (military oceanography). While transmission loss experiments along a single radial could take hours, a single charge dropped near the towed array gave information on all radials in just minutes. Anomalies could be immediately identified, for investigation by more precise techniques. The REA exercises attracted interest and led to the Boundary Characterization and Clutter Joint Research Projects between CMRE (the NATO Centre for Maritime Research and Experimentation), Canada, and US. [Work supported by ONR.]

2:20–2:35 Break

2:35

2pUW5. Ocean acoustic boundary characterization multi-national experiments. Charles W. Holland (Elec. and Comput. Eng., Portland State Univ., Portland State Univ., Elec. and Comput. Eng., Portland, OR 97207, charles.holland@pdx.edu) and Peter Nielsen (None, Esbjerg, Denmark)

The condition of the sea surface and seabed boundaries of the ocean often significantly affect the characteristics of acoustic signal propagation, reverberation and noise particularly in shallow water. A series of collaborative experiments were conducted from 2000 to 2009 to better understand the impact of ocean boundary characteristics on ocean acoustic signals in the frequency range of 100 – 5000 Hz. Acoustic measurements included long-range (waveguide) propagation, reverberation, and clutter as well as measurements of local boundary reflection and scattering. These were supported by oceanographic, geophysical and geologic measurements. Experimental locations included the Tyrrhenian Sea, Straits of Sicily, the New Jersey shelf, and the Scotian shelf with collaborators from NATO-STO CMRE and national laboratories and institutions from Italy, Canada, France, and the US. This talk will provide an overview of some of the main results from these experiments. [Research sponsored by CMRE(NATO), ONR(US), NUWC(US), INGV(IT), SHOM(FR), DRDC(CA), Italian Navy(IT).]

2:55

2pUW6. Environmentally constrained modeling of mid-frequency transmission loss and reverberation measured during the Target and Reverberation Experiment 2013. Brian T. Hefner (Appl. Phys. Lab., Univ. of Washington, Appl. Phys. Lab., University of Washington, 1013 NE 40th St., Seattle, WA 98105, hefner@apl.washington.edu), Jie Yang, and Dajun Tang (Appl. Phys. Lab, Univ. of Washington, Seattle, WA)

The goal of the Target and Reverberation Experiment (TREX13) was to make contemporaneous measurements of mid-frequency (1.5–4 kHz) transmission loss and reverberation with extensive environmental measurements so detailed model/data comparison can be achieved and important environmental factors can be identified for different applications. APL-UW collaborated with ARL-Penn State, led by John R. Preston, to deploy the Five Octave Research Array (FORA) on a "clothesline" about 2.1 m above the seafloor. This fixed-source/receiver configuration helped eliminate uncertainties from the motion of a towed array and allow reverberation measurement along a narrow, 7-km-long section of seafloor. The experiment site had a fairly complex spatial distribution of both sediment type and sediment scattering properties and keeping the FORA in a fixed position was instrumental in understanding and modeling the reverberation. While soft sediments make up only 27% of the sediments by area at the site, it is necessary to account for this spatial dependence so that both transmission loss and reverberation can be modeled using a consistent set of environmental inputs. These models and their implications for our understanding of the environmental factors which most impact mid-frequency reverberation will be discussed. [Work supported by the U.S. Office of Naval Research.]

3:15

2pUW7. The Littoral Continuous Active Sonar Multi-National Joint Research Project 2014–2020. Kevin D. LePage (NATO STO Ctr. for Maritime Res. and Experimentation, NATO STO Ctr. for Maritime Res. and Experimentation, Viale San Bartolomeo 400, La Spezia 19126, Italy, kevin.lepage@cmre.nato.int), Alessandra Tesei (NATO STO Ctr. for Maritime Res. and Experimentation, La Spezia, Italy), Stefano Biagini (NATO STO Ctr. for Maritime Res. and Experimentation, La Spezia, SP, Italy), simon lourey (Sonar Technol. and Systems, Defence Sci. and Technol. Group, Edinburgh, South Australia, Australia), stefan murphy, Jeffrey Bates, Gary Inglis (Underwater Warfare, Defence Res. and Development Canada, Dartmouth, NS, Canada), Paul C. Hines (ECE, Dalhousie, Halifax, NS, Canada), Gary Wood, Catherine L. Smith (Platform Systems Div., Dstl, Fareham, Hampshire, United Kingdom), Vicenzo Manzari, Daniele Terracciano (Underwater Defence Section, Centro Sperimentazione e Supporto Navale, La Spezia, SP, Italy), Paul van Walree (Norwegian Defence Res. Establishment, Horten, Norway), Doug Grimmett (SPAWAR Systems Ctr. Pacific, San Diego, CA), and Doug Abraham (CausaSci, Ellicott City, MD)

One of John Preston's achievements during his career was the hosting of multinational sonar experimentation efforts whilst a scientist at SACLANTCEN. The Littoral Continuous Active Sonar Multi-National Joint Research Project is the most recent international collaborative experimentation activity focused on sonar hosted by NATO STO Centre for Maritime Research and Experimentation, SACLANTCEN's sucessor. Between 2014 and 2020 LCAS brought scientists and engineers from 7 NATO and Partner Nations together with the CMRE to evaluate the effectiveness of continuous active sonar in shallow littoral environments. In this talk, the objectives of the project are laid out, the scientific issues and experimental approach reviewed, details about the four sea trials conducted under LCAS are presented, and a summary of the major results of the project is provided.

3:35

2pUW8. The legacy of the five octave research array and its impact on future towed array development. Chad M. Smith (Appl. Res. Lab., The Penn State Univ., State College, PA 16804, chad.smith@psu.edu) and Jim Dorminy (Appl. Res. Lab., The Penn State Univ., State College, Pennsylvania, U.S. Minor Outlying Islands)

The five octave research array (FORA) was developed and implemented in support of the Office of Naval Research (ONR) Ocean Acoustics program's experimental efforts. Dr. John R. Preston spearheaded planning, development, and experimental operations of this well-known acquisition system. Planning and development of the system took place between 1997 and 2002, while engineering assessment and array enhancements continued through 2004. From initial delivery to the Penn State Applied Research Laboratory to the present, the FORA made possible over 20 very successful sea trials. These trials were primarily in the interest of oceanographic and underwater acoustics, as well as signal processing research. The data collected, with John at the helm, fostered hundreds (if not thousands) of scholarly works, while John's collaborative nature allowed an enormous number of researchers and students to benefit from the data and experience in at-sea experimentation. One of John's final contributions was to renew planning and discussions for a replacement research array to continue in place of the aging FORA, which was becoming less dependable and increasingly expensive to maintain. This talk will discuss the development and history of the FORA, its involvement in a selection of collaborative ONR trials, and the FORA replacement acquisition system.

TUESDAY AFTERNOON, 24 MAY 2022 DIRECTORS ROW E, 2:00 P.M. TO 3:15 P.M.

Meeting of Accredited Standards Committee (ASC) S3 Bioacoustics

W. J. Murphy, Chair ASC S3
National Institute for Occupational Safety and Health, 1090 Tusculum Ave., Cincinnati OH 45226

T. Ricketts, Vice Chair ASC S3
Vanderbilt University, 1215 21st Ave. South, Rm. 8310, Nashville TN 37232

Accredited Standards Committee S3 on Bioacoustics. Working group chairs will report on the status of standards under development. Consideration will be given to new standards that might be needed over the next few years. Open discussion of committee reports is encouraged.

People interested in attending the meeting of the TAGs for ISO/TC 43 Acoustics and IEC/TC 29 Electroacoustics, take note - those meetings will be held in conjunction with the Standards Plenary meeting at 9:15 a.m. on Tuesday, 24 May 2022.

Scope of S3: Standards, specifications, methods of measurement and test, and terminology in the fields of psychological and physiological acoustics, including aspects of general acoustics which pertain to biological safety, tolerance and comfort.

TUESDAY, 24 MAY 2022 DIRECTORS ROW E, 3:30 P.M. TO 4:45 P.M.

Meeting of Accredited Standards Committee (ASC) S3/SC 1, Animal Bioacoustics

D. S. Houser, Chair ASC S3/SC 1
National Marine Mammal Foundation, 2240 Shelter Island Dr., Suite 200, San Diego, CA 92106

M. Roch, Vice Chair ASC S3/SC 1
San Diego State University, 5500 Campanile Dr., San Diego, CA 92182

Accredited Standards Committee S3/SC 1 on Animal Bioacoustics. Working group chairs will report on the status of standards under development. Consideration will be given to new standards that might be needed over the next few years. Open discussion of committee reports is encouraged.

People interested in attending the meeting of the TAGs for ISO/TC 43/SC 1 Noise and ISO/TC 43/SC 3, Underwater acoustics, take note - those meetings will be held in conjunction with the Standards Plenary meeting at 9:15 a.m. on Tuesday, 24 May 2022.

Scope of S3/SC 1: Standards, specifications, methods of measurement and test, instrumentation and terminology in the field of psychological and physiological acoustics, including aspects of general acoustics, which pertain to biological safety, tolerance and comfort of non-human animals, including both risk to individual animals and to the long-term viability of populations. Animals to be covered may potentially include commercially grown food animals; animals harvested for food in the wild; pets; laboratory animals; exotic species in zoos, oceanaria or aquariums; or free-ranging wild animals.

Meeting of Accredited Standards Committee (ASC) S12 Noise

D. F. Winker, Chair ASC S12
ETS-Lindgren Acoustic Systems, 1301 Arrow Point Dr., Cedar Park, TX 78613

D. Knight, Vice Chair ASC S12
Applied Acoustics, Mechanics and Packaging, Ingersoll Rand Trane, 3600 Creek Rd, La Crosse, WI 54601

Accredited Standards Committee S12 on Noise. Working group chairs will report on the status of noise standards currently under development. Consideration will be given to new standards that might be needed over the next few years. Open discussion of committee reports is encouraged.

People interested in attending the meeting of the TAG for ISO/TC 43/SC 1, Noise, and ISO/TC 43/SC 3, Underwater acoustics, take note - that meeting will be held in conjunction with the Standards Plenary meeting at 9:15 a.m. on Tuesday, 24 May 2022.

Scope of S12: Standards, specifications and terminology in the field of acoustical noise pertaining to methods of measurement, evaluation and control, including biological safety, tolerance and comfort, and physical acoustics as related to environmental and occupational noise.

OPEN MEETINGS OF TECHNICAL COMMITTEES

Technical Committees of the Acoustical Society of America will hold open meeting on Tuesday, Wednesday, and Thursday evenings.

All meetings will begin at 7:30 p.m., except for Engineering Acoustics which will meet starting at 4:45 p.m. on Tuesday and Computational Acoustics which will meet starting at 4:30 p.m. on 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 discussion.

Committees meeting on Tuesday

Engineering Acoustics (4:45 p.m.)	Governors Square 10
Acoustical Oceanography	Governors Square 14
Animal Bioacoustics	Governors Square 17
Architectural Acoustics	Plaza Ballroom A
Physical Acoustics	Governors Square 11
Psychological and Physiological Acoustics	Plaza Ballroom D
Signal Processing in Acoustics	Governors Square 16
Structural Acoustics and Vibration	Governors Square 12

Committees meeting on Wednesday

Biomedical Acoustics	Governors Square 15

Committees meeting on Thursday

Computational Acoustics (4:30 p.m.)	Governors Square 12
Musical Acoustics	Directors Row H
Noise Plaza	Ballroom D
Speech Communication	Plaza Ballroom E
Underwater Acoustics	Governors Square 14

Session 3aAA

Architectural Acoustics, ASA Committee on Standards, Biomedical Acoustics, Physical Acoustics, and Noise: Advanced Measurement and Modeling of Sound Absorption and Scattering I

Mélanie Nolan, Cochair
School of Architecture, Rensselaer Polytechnic Institute, Greene Bldg., 110 8th Street, Troy, NY 12180

Ning Xiang, Cochair
School of Architecture, Rensselaer Polytechnic Institute, 110 Eighth Street, Troy, NY 12180

Peter D'Antonio, Cochair
RPG Acoustical Systems LLC, 99 South Street, Passaic, NJ 07055

Chair's Introduction—8:15

Invited Papers

8:20

3aAA1. Go's and no-go's towards more correct reverberation room measurements of absorption. Michael Vorlaender (IHTA, RWTH Aachen Univ., Kopernikusstr. 5, Aachen 52056, Germany, mvo@akustik.rwth-aachen.de) and Jamilla Balint (IHTA, RWTH Aachen Univ., Aachen, Germany)

Measurements of random-incidence absorption coefficients have been hotly contested since the ASA was founded in 1929. Yes, you read correctly, since 1929! Theories, explanations, myths, and strange interpretations appear, disappear, and reappear over the decades. What is wrong with this method of measurement? This paper discusses some of the methodological difficulties and reasons for discrepancies when comparing results between different reverberation chambers. It also presents possibilities arising from recent developments, such as the isotropy and angle dependence of the sound field incident on the absorber, the multi-exponential model for calculating decay times, and reference absorbers.

8:40

3aAA2. Absorption measurements—A never ending story?!. Christian Nocke (Akustikbuero Oldenburg, Sophienstr. 7, Oldenburg, Nds. 26121, Germany, nocke@akustikbuero-oldenburg.de)

A brief historical review on measurement techniques of sound absorption or the acoustic surface impedance will be presented, starting from procedures in the 1930s. Many *in situ* and laboratory methods have been described over the years. Most of these methods are based on the assumption of plane wave propagation or diffuse sound field. Measurement examples will be shown for some of the methods presented and compared to each other. A commonly used measurement procedure of so-called diffuse sound absorption coefficients is described in the international standard ISO 354. The present revision of the standard only focuses on the conditioning of the sound field and further applying Sabine equation. A diffuse sound field is atheoretical approach. Geometrical acoustics as the basis of sound field simulation has been applied to measurements in a virtual reverberation chamber on absorber baffle set-ups [see Probst (2015)] and showed good agreement between measured and calculated values. This approach is used to investigate the effect of sample size and sample position. Correspondingly, non-standard measurements are compared with simulation results.

9:00

3aAA3. Measuring angle-dependent sound absorption in ordinary rooms. Mélanie Nolan (School of Architecture, Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, nolanm5@rpi.edu)

The effective use of sound absorbing materials in room acoustical design requires the use of angle-dependent coefficients, as opposed to random incidence coefficients, to properly account for the dissipation of sound energy at the room's boundary. In this study, a method is proposed for measuring the angle-dependent surface impedance and absorption coefficient of a boundary material *in situ*, in an ordinary room. The proposed method relies on the reconstruction of the pressure and normal particle velocity at the boundary using a plane-wave decomposition of the sound field measured with an array of microphones. In sufficiently reverberant environments, where sound strikes the material from nearly all directions in space, the angle-dependent properties of the boundary can be obtained simultaneously for all angles of incidence, from a single source position. In addition, the proposed methodology enables us to identify the specific directions of incidence on the material and makes it possible to estimate its operational acoustic performance. The validity of the method is examined experimentally in a reverberation chamber and in a classroom, using a robotic arm to scan the sound field in the vicinity of an absorbing boundary. Consistent absorption data are measured for various source positions and room configurations.

9:20

3aAA4. Non-negative surface contributions for cavities based on sound energy density. Caglar Gurbuz (Chair of Vibroacoustics of Vehicles and Machines, Tech. Univ. Munich, Boltzmannstr. 15, Garching 85748, Germany, caglar.guerbuez@tum.de) and Steffen Marburg (Chair of Vibroacoustics of Vehicles and Machines, Tech. Univ. Munich, Garching, Germany)

Surface and panel contribution analysis provides a useful methodology to identify sources of radiated sound in cavities. In commercial software, contribution analyses are performed by considering the sound pressure at an internal point. For visualization, a bar chart is usually supplied for the vibrating surfaces. The current solution has two major drawbacks: At first, the traditional technique provides only results for surfaces which are in motion. At second, the sound pressure depends on the location of the field point, which can lead to a deteriorated performance in regions with low pressure values. It is our aim to present non-negative surface contributions for sound energy density in a cavity. Energy-based contributions provide further insight into the characteristics of cavities, as they provide a holistic evaluation of sound pressure and particle velocity. For this, the boundary element method is applied to solve the Helmholtz equation for interior problems. In close analogy to the non-negative intensity, the energy-based contributions are determined from a quadratic form in order to bypass cancellation effects. Results show that regions with high contributions to the energy density are effectively recovered. Some of these surfaces appear almost inactive if the contributions are analyzed only with respect to sound pressure. As such, the evidence from this study suggests that energy-based surface contributions provide an effective quantity to identify sound sources, particularly for regions with low pressure values.

9:40

3aAA5. Non-cuboid Iterative room optimizer. Peter D'Antonio (Res. & Development, REDI Acoust., 99 South St., Passaic, NJ 07055, pdantonio@rpgacoustic.com) and Rinaldi P. Petrolli (Res. & Development, REDI Acoust., Florianópolis, Santa Catarina, Brazil)

In past years, various iterative optimization programs emerged to separately determine the optimal room ratios, sources and listening positions of perfectly reflective cuboid rooms, using the image-source model. Despite its fast computation times, this approach does not account for scattering, phase change at the boundary and cannot be extended to non-cuboid rooms. This presentation describes the current status of a program called NIRO, that offers a solution to those issues, by using the Boundary Element Method (BEM) to compute the frequency response from 20–200 Hz, considering the effects of the boundary's complex admittance and all acoustical elements inside the room. With BEM as its engine, a room optimization genetic algorithm was developed to optimize source and receiver positions simultaneously with the room geometry, aiming to present the best possible acoustic environment given imposed restraints. To control the room's temporal decay, low-frequency acoustic treatments were added to the BEM model. By using transfer matrix models, the acoustical behavior of different multilayered treatments can be modeled and inserted into the BEM simulation to evaluate the change in the room's acoustic field and in the frequency response at the receiving positions. 3D waterfall plots illustrate the temporal decay following optimization. Examples will be presented.

10:00–10:15 Break

10:15

3aAA6. A multipole model-based boundary admittance estimation for wave-based room-acoustic simulations using Bayesian inference. Ziqi Chen (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, chenz33@rpi.edu), Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY), and Kirill V. Horoshenkov (Univ. of Sheffield, Sheffield, United Kingdom)

Room-acoustic simulations for sound wave propagation require significant knowledge of the boundary conditions for the room boundary surface. This work presents the application of model-based Bayesian inference on the surface admittance estimation problem. First, this work deals with selecting the multipole admittance model through the higher level of inference, Bayesian model selection. And the parameters of the chosen multipole admittance model are estimated through the first (lower) level of inference, Bayesian parameter estimation. This work focuses on the approximation of the surface admittance as a function of frequency given a set of acoustic admittance. This frequency-dependent admittance is deduced either from the experimental measurements of a porous material or numerical predictions in order to demonstrate wave-based boundary conditions to incorporate arbitrary admittance functions with frequency ranges under consideration. According to the analytical results and numerical verifications conducted in this work, Bayesian inference based on the multipole model is well-suited for the estimation of the frequency-dependent boundary conditions within a wave-based simulation framework.

10:35

3aAA7. Boundary element method virtual goniometer to predict the diffusion and scattering coefficients. Peter D'Antonio (RPG Acoust. Systems LLC, 99 South St., Passaic, NJ 07055, pdantonio@rpgacoustic.com), Luiz Augusto T. Ferraz Alvim (RPG Acoust. Systems LLC, São Paulo, São Paulo, Brazil), and Rinaldi P. Petrolli (Res. & Development, REDI Acoust., Florianópolis, Santa Catarina, Brazil)

The experimental measurement of the diffusion coefficient according to ISO 17497-2 is very time consuming and requires several sample periods to evaluate the effect of diffraction lobes, an anechoic or large reflection-free volume and far-field conditions. Wave-based BEM methods have predicted diffusion and correlation scattering coefficients very accurately [Hargreaves *et al.*, J. Acoust. Soc. Am. **108** (4), 1710–1720 (2000)]. This presentation will describe a new Python Virtual Goniometer program, called VIRGO, which predicts the free-field and surface-mounted diffusion and correlation scattering coefficients for any shaped surface that can be successfully meshed from a three-dimensional file. The program accurately predicts the periodic diffraction grating lobes of a reference one-dimensional hemicylinder and a two-dimensional hemisphere, both in the free-field and surface mounted on a boundary. The diffusion coefficient and three-dimensional polar responses of additional number-theoretic and optimized profiled and curvilinear shapes will also be compared with scale model boundary-plane goniometer measurements. The results will verify that it is possible to precisely predict the diffusion and correlation scattering coefficients, without having to fabricate or 3D print scale models or full-scale samples.

10:55

3aAA8. Acoustical properties of granular aerogel agglomerates. Amrutha Dasyam (Aerosp. Eng., Wichita State Univ., Wichita, KS), Yutong Xue (Midea Corporate Res. Ctr., West Lafayette, IN), Bhisham Sharma (Aerosp. Eng., Wichita State Univ., 1845 Fairmount St., Wichita, KS 67260, bhisham.sharma@wichita.edu), and J. S. Bolton (Ray W. Herrick Laboratories/School of Mech. Eng., Purdue Univ., West Lafayette, IN)

Granular aerogel agglomerates obtain their unique mechanical and functional properties from their underlying mesoporous structure and intergranular interactions. In this presentation, we focus on the acoustical properties of granular aerogel layers and present experimental results characterizing the absorption behavior of granular silica aerogels with average particle sizes ranging from 5 μm to 2 mm. Our results show that the absorption behavior of granular aerogels strongly depends on the average particle size. To better understand this particle size dependence, we model the granular agglomerates using Biot's limp and poro-elastic formulation. Finally, we use the developed model to design layered aerogel granular stacks with wideband sound absorption properties.

11:15

3aAA9. Modeling of micro-slit /perforated panels in multilayers for efficient design and validation of broadband sound absorbers. Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., 110 Eighth St., Troy, NY 12180, xiangn@rpi.edu), Michael Hoeft, and Cameron J. Fackler (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY)

This work focuses on design of multilayer micro-perforated/slit panels (MPP/MSP) to create high sound absorption over a wide bandwidth. Theoretically micro-slit panels are equivalent to micro-perforated panels and can similarly achieve high absorption coefficients. Yet, fabrication of microslit panels is much easier. Using transparent panels, they are visually more transparent than MPPs since the MSPs yield more unobstructed panel for a given perforation rate. However, MPP and MSP absorbers in single setting are frequency limited to a bandwidth often not sufficient in applications. Stacking multilayered MPP/MSP panels, a high absorption becomes achievable in a wider bandwidth. Using existing single-panel MPP/MSP models, a transfer matrix method is suitable for cascading multilayers to predict overall absorption performance. This prediction model is applied to a Bayesian framework in coping with design challenges arising from designing MPP/MSP parameters for multiple layers that fulfills a given design scheme. The Bayesian design reported here rapidly converges to a parsimonious number of layers with optimized parameters for each layer, and the overall absorption is able to fulfill the design scheme for minimally possible fabrication complexity. In addition, performing iterative Bayesian analysis during the panel fabrication and experimental validation guides corrections due to fabrication inaccuracies.

Session 3aAB

Animal Bioacoustics: Animal Bioacoustics: Behavior and Physiology

Michael Smotherman, Chair
Biology, Texas A&M University, 3258 TAMU, College Station, TX 77843-3258

Chair's Introduction—8:20

Contributed Papers

8:25

3aAB1. Perception at the extreme: Adaptive vocal-motor behavior of bats at high speed flight and in groups. Laura Kloepper (Dept. of Biological Sci., Univ. of New Hampshire, 230 Spaulding Hall, Durham, NH 03824, laura.kloepper@unh.edu), Alexandra Weesner, and Ian Bentley (Dept. of Chemistry and Phys., Saint Mary's College, Notre Dame, IN)

In order to successfully navigate through their environments without collisions at often high speeds, bats rely on two things: successful reconstruction of their environment from returning echoes and motor outputs in response to intermittent sensory information. This depends on the coordination between sensory input and locomotive output. Many bats regularly form dense aggregations in the wild and fly at speeds exceeding 100 km/h, likely posing significant sensorimotor challenges. We recorded the echolocation and flight behavior of wild frequency-modulated (FM) bats as they exited and returned to their cave roost to understand how bats coordinate flight and echolocation when faced with sensory challenges. We found that when exiting the roost in groups, many bats coordinate flight behavior and wingbeat movement with nearby conspecifics, but based on heading delays, this coordination is likely not driven by acoustic cues and instead may be guided by aerodynamic cues. During high-speed roost re-entry, bats modify echolocation signals consistent with increasing object discrimination as they approach the roost, yet adjustments in flight behavior do not coincide with the reception of echoes, suggesting that bats may be combining information from multiple pulses or senses to plan their flight path. [Work supported by ONR N000141612478 and NSF-IIBR-1916850.]

8:40

3aAB2. Abstract withdrawn.

8:55

3aAB3. Horseshoe bats use not changes in echo delay but Doppler shift to perceive approaching objects. Soshi Yoshida (Life and Medical Sci., Doshisha Univ., Ishinkan IN505N, Tataramiyakodani 1-3, Kyotanabe, Kyoto prefecture 610-0394, Japan, ctug1040@mail4.doshisha.ac.jp), Kazuma Hase (Psych., Neurosci. & Behav., McMaster Univ., Hamilton, ON, Canada), Olga Heim (Life and Medical Sci., Doshisha Univ., Kyotanabe, Japan), Kohta I. Kobayasi (Life and Medical Sci., Doshisha Univ., Kyotanabe, Kyoto prefecture, Japan), and Shizuko Hiryu (Life and Medical Sci., Doshisha Univ., Kyotanabe, Japan)

Echolocating bats use echo delay for target ranging and Doppler shift information for relative velocity recognition. However, how they perceive moving objects remains unclear. To investigate this question, we played back echolocation pulses in real-time as virtual echoes to Japanese horseshoe bats (*Rhinolophus ferrumequinum nippon*) on a perch in a flight room. Since echoes coming back from an approaching object are theoretically characterized by both changes in echo delay and the presence of Doppler shift, we reproduced an artificial approaching object by encoding these two acoustic parameters in the virtual echoes. As a result, only Doppler shift evoked bats flight reaction, showing that they use only Doppler shift and not change in echo delay to perceive approaching objects. Also, we played back only constant frequency (CF) component and confirmed that they use the CF component to detect Doppler shift. Furthermore, as a response to the Doppler shift in the perceived echo, bats increased the bandwidth of the terminal component of their pulse. Surprisingly, this response occurred in the very first pulse after Doppler shift, which indicates bats can adapt their echolocation pulse characteristics to changing situations within a pulse. [This work was supported by JSPS KAKENHI Grant Nos. 18H03786 and 16H06542.]

9:10

3aAB4. Topographical distribution of spectrotemporal receptive field properties in the bat primary auditory cortex. Kushal Bakshi (Inst. for Neurosci., Texas A&M Univ., 3258 TAMU, Biological Sci. Bldg. West, Rm. 107, College Station, TX 77843, kushalbakshi@tamu.edu), Silvio Macias (Biology, Texas A&M Univ., College Station, TX), Todd Troyer (Dept. of Neurosci., Developmental and Regenerative Biology, Univ. of Texas at San Antonio, San Antonio, TX), and Michael Smotherman (Biology, Texas A&M Univ., College Station, TX)

Spectrotemporal modulations are a prominent feature of natural sounds including animal vocalizations and human speech. Echolocating bats must process spectrotemporal cues such as echo delays and spectrum properties to navigate their environment; however, the neuronal networks in the primary auditory cortex (A1) that process features of natural sounds remain incompletely understood. Here, we investigated the topographical distribution of tuning properties throughout A1 of Mexican free-tailed bats. While the majority of the bats' A1 neurons are tuned to downward FM sweeps, the entire cortex is required to capture and categorize patterns of spectral details embedded within each biosonar echo. We captured the neural responses of the A1 to complex acoustic stimuli using linear, 16-channel translaminar microelectrode arrays. We analyzed 145 spectrotemporal receptive fields (STRFs) across A1, as well as the spectral and temporal modulation transfer functions (sMTF and tMTF, respectively) to determine neural tuning preferences in the spectral and temporal dimensions. We found evidence of neuronal sub-categories that were classified as having simple or complex (multi-peaked) STRFs. For simple STRFs, sMTFs and tMTFs were evaluated as a function of best frequency, and spatial location and revealed tonotopic trends similar to those reported for non-specialized animals.

9:25

3aAB5. Pre-pulse inhibition of the acoustic startle response as a behavioral assay for sound localization abilities of the Mongolian gerbil. Matthew D. Sergison (Physiol., Univ. of Colorado Anschutz Medical Campus, 12800 E 19th Ave. RC1 North 740G, Aurora, CO 80045, matthew.sergison@cuanschutz.edu), John Peacock, Monica A. Benson, Shani Poleg (Physiol., Univ. of Colorado Anschutz Medical Campus, Aurora, CO), Nathaniel Greene (Otolaryngol., Univ. of Colorado School of Medicine, Aurora, CO), Achim Klug (Physiol., Univ. of Colorado Anschutz Medical Campus, Aurora, CO), and Daniel Tollin (Physiol. and Otolarnygology, Univ. of Colorado Anschuz Medical Campus, Aurora, CO)

The Mongolian gerbil (*Meriones unguiculatus*) has been used to investigate mechanisms of binaural and spatial hearing. However, behavior assays used to test gerbil spatial hearing using operant conditioning are very time consuming. Pre-pulse inhibition (PPI) of the acoustic startle response offers a method that requires no training and is thus high throughput. Here, we examine whether PPI can be used to assess spatial hearing in gerbils. In eight gerbils, we presented a continuous broadband noise that swapped speaker locations as a pre-pulse prior to a startle stimulus and found PPI increased with wider angles of swaps. Swap angles at 30° (±15° re: midline) or higher showed significantly higher PPI compared to baseline for swaps across midline and in each hemifield. We also performed speaker swap with a low- (0.5 kHz) and high-pass (4 kHz) filter and found that PPI increased at wider angles for both conditions. PPI also increased at wider angles and lower intensity of a spatial broadband masker to a broadband chirp pre-pulse. We successfully demonstrate the gerbil's sound localization abilities and show that PPI paradigms are capable of quickly testing large numbers of gerbils and reveal performance similar to operant conditioning methods. [Work supported by R01-DC017924.]

9:40

3aAB6. Otitis media-induced cochlear immune response and opportunistic ototoxicity. Hongzhe Li (VA Loma Linda Healthcare System, 11201 Benton St., Res. Service (151), Loma Linda, CA 92357, Hongzhe.Li@va.gov), Yongchuan Chai, and Liana Sargsyan (VA Loma Linda Healthcare System, Loma Linda, CA)

Lipopolysaccharide (LPS), an essential component of the bacterial endotoxin, activates tissue macrophages and triggers the release of inflammatory cytokines. In animal models, intratympanic (*i.t.*) injection of LPS is known to simulate acute otitis media (AOM) and modifies the structure and function of the inner ear. However, whether LPS-induced AOM modulates the uptake of ototoxic aminoglycosides *in vivo* is unclear. Here, we established an AOM mouse model to investigate the gentamicin uptake in the inner ear and the change of cochlear inflammation for pertaining ototoxicity mechanisms. We found that LPS-induced AOM switched on the cochlear inflammatory response, including macrophage infiltration and upregulation of pro-inflammation cytokines, and significantly enhanced the cochlear uptake of gentamicin. Potential mechanisms of enhanced drug uptake may include increased strial permeability due to acute middle-ear inflammation, resulting in higher drug concentration in the stria vascularis and the endolymph, and subsequently higher drug uptake by hair cells. Other possible mechanisms, such as the activation of specific candidate channels and endocytosis followed by inflammation, need to be further interrogated. In sum, this study improves our understanding of drug trafficking in the pathological cochlea and provides a reference for drug treatment of AOM while protecting the inner ear function.

9:55–10:10 Break

10:10

3aAB7. Modeling the potential for vessel collision with southern resident killer whales. Dana A. Cusano (JASCO Appl. Sci. (Australia) Pty. Ltd., 14 Hook St., Capalaba, Queensland 4157, Australia, dana.cusano@jasco.com), Molly Reeve, Michelle Weirathmueller (JASCO Appl. Sci. (USA), Inc., Silver Spring, MD), Karlee Zammit (JASCO Appl. Sci. (Canada), Ltd., Victoria, BC, Canada), Steven Connell (JASCO Appl. Sci. (Australia) Pty., Ltd., Capalaba, Queensland, Australia), and David Zeddies (JASCO Appl. Sci. (USA), Inc., Silver Spring, MD)

Agent-based models can be used to simulate the behavior of animals within a stimulus field. For example, such models are used to estimate the sound exposure of simulated animals (animats) moving within a computed sound field. This approach has not yet been used to estimate the probability of vessel collision between marine mammals and ships. One reason is the difficulty of defining aversive behavior in response to disturbance. While many animals display aversion, models often use a simplified approach due to insufficient data. Building on the JASCO animal simulation model including noise exposure (JASMINE), a vessel collision framework was developed for southern resident killer whales (SRKWs) using modeled vessel sound fields and AIS data. Animats were programmed to increasingly avoid louder, closer, and additional vessels by changing their heading, speed, and behavioral state based on published data from SRKWs. Animats that remained within a calculated encounter radius despite aversive behavior were considered struck by the vessel. Parameters governing aversive behaviors were calibrated by comparing the proportion of animats struck in the simulation with the collision probability for real-world SRKWs. This model provides a starting point to model the risk of vessel collision for SRKW and other species.

10:25

3aAB8. Railway noise and long-distance calls of free-living maned wolves in Ecological Station of Itirapina, São Paulo, Brazil. Bruna Campos Paula (Mech. Eng., Univ. of São Paulo, 1317 Alves Guimaraes st., São Paulo, São Paulo 05410002, Brazil, brunacampospaula@gmail.com), Lilian Luchesi (Faculty of Philosophy, Sci. and Letters, Univ. of São Paulo, São Paulo, São Paulo, Brazil), and Patrícia Monticelli (Faculty of Philosophy, Sci. and Letters, Univ. of São Paulo, São Paulo, Ribeirão Preto, Brazil)

For the present study, we evaluated the long-distance vocalization of free-living maned wolves of the Ecological Station of Itirapina, São Paulo, Brazil. This population has been exposed to railway noise for at least 100 years, since the railroad activities, in 1916. For the present study, we aim to investigate long-distance calls of maned wolves in this area, verifying the presence and absence of calls during the railway noise. For six months, between February and December 2016, we collect data using passive acoustic monitoring systems. The evaluation data set comprised 600 h of file recordings. The measurement of sound activity used in this study was the occurrence of long-distance calls sequences and railway noise. To assess the variation of the occurrence in long-distance calls and railway noise, hourly calling rates were calculated. The railway noise was most frequent between 9 p.m. and 10 p.m. UTC. The time distribution of long-distance calls and railway noise varied between the 12 hours of record with more vocalizations from 12 a.m. to 3 a.m. UTC. Our results provided cues of how a vocally active mammal species may communicate in a noisy environment.

10:40

3aAB9. Natural cues for invertebrate and fish hearing: Particle motion measurements on coral reefs. Ian T. Jones (Biology, Woods Hole Oceanographic Inst., Woods Hole Oceanographic Inst., 266 Woods Hole Rd. MS #50, Woods Hole, MA 02543, ian.t.jones@unh.edu), Michael Gray (Inst. of Biomedical Eng., Univ. of Oxford, Oxford, United Kingdom), and T. Aran Mooney (Biology, Woods Hole Oceanographic Inst., Woods Hole, MA)

Coral reef soundscapes are increasingly studied for their ecological uses by invertebrates and fishes, their applications for monitoring habitat quality, and to investigate effects of anthropogenic noise pollution. Few examinations of aquatic soundscapes have reported particle motion levels and variability, despite their relevance to invertebrates and fishes. We quantified ambient particle acceleration from orthogonal hydrophone arrays over several months at four coral reef sites in the U.S. Virgin Islands, which varied in benthic habitat and fish communities. Temporal trends of particle

acceleration were similar to those of sound pressure, and the strength of diel trends in both metrics significantly correlated with percent coral cover. Low frequency (<100 Hz) and high magnitude empirical particle acceleration levels diverged from plane wave approximations. Particle acceleration levels were also determined for boat and example fish sounds. Comparisons with particle acceleration derived audiograms suggest greatest capacity of invertebrates and fishes to detect soundscape components below 100 Hz, and potentially poorer detectability of soundscape cues by some invertebrates compared to fishes. Based on our results, we discuss research questions in soundscape ecology for which reporting of particle motion is essential versus those for which sound pressure may suffice.

10:55

3aAB10. Comparison of three portable volumetric arrays to localize and identify fish sounds in the wild. Xavier Mouy (Passive Acoust. Res. Group, NOAA, 3377 SW 28th Terrace, Miami, FL 33133, xavier.mouy@outlook.com), Morgan Black, Kieran Cox, Jessica Qualley (Uvic, Victoria, BC, Canada), Stan Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada), and Francis Juanes (Uvic, Victoria, BC, Canada)

We describe three portable volumetric audio/video arrays capable of identifying species-specific fish sounds in the wild. Each array can record fish sounds, acoustically localize the fish in three-dimensions (using linearized or fully non-linear inversion), and record video to identify the species and observe their behavior. The design of each array accommodates specific logistical and financial constraints, covering a range of nearshore habitats and applications. The first platform is composed of six hydrophones, an acoustic recorder, and two video cameras secured to a $2 \times 2 \times 3$ m PVC frame. Hydrophone placement is defined using simulated annealing to maximize localization accuracy. The second platform uses a single video camera, four hydrophones, and an acoustic recorder on a one cubic meter PVC frame. It can be deployed on heterogeneous substrates but has lower localization capabilities. The third platform consists of four hydrophones connected to an acoustic recorder mounted on a tethered underwater drone with built-in video. It allows remote control and real-time positioning in response to observed fish presence but with reduced localization capabilities. The three platforms were deployed off British Columbia, Canada, and used to identify and characterize new sounds from quillback rockfish, copper rockfish, and lingcod.

WEDNESDAY MORNING, 25 MAY 2022 GOVERNORS SQUARE 14, 8:30 A.M. TO 12:00 NOON

Session 3aAO

Acoustical Oceanography Acoustic Sensing of Biological and Physical Processes in Littoral Environments

Kevin M. Lee, Cochair
Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758-4423

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

Chair's Introduction—8:30

Invited Papers

8:35

3aAO1. How prevalent is acoustic scattering from physical microstructure? Andone C. Lavery (AOPE, Woods Hole Oceanographic Inst., 98 Water St., Woods Hole, MA 02543, alavery@whoi.edu), Christopher Bassett (Appl. Phys. Lab., Univ. of Washington, Seattle, NJ), and Scott Loranger (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

Any oceanic environment with spatial gradients in sound speed and density can result in acoustic scattering hot spots. These acoustic hotspots can be rapidly evolving and vary in their spatial heterogeneity. Here, we present data collected with a variety of split-beam and multi-beam echosounders illustrating the broad array of environments and spatial scales associated with scattering from physical microstructure, including shear instabilities in estuarine environments, non-linear internal waves on the continental shelf, strong interface scattering due to double-diffusion, scattering from strong gradients, or interfaces, and turbulent microstructure at the new England Shelf Break Front. The theoretical acoustic scattering formulations for different types of physical microstructure are applied to these different environments, and recommendations are made for optimal frequency bands to sample the different types of physical microstructure and

the optimal measurements for inference of parameters that describe the physical microstructure. The impact of other scattering sources, such as suspended sediments, bubbles, and biological targets, on successful acoustic sampling of physical microstructure is also discussed. [This work was supported by the ONR.]

8:55

3aAO2. Echosounding in estuaries: Lessons learned from operations and data in four coastal systems. Christopher Bassett (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, cbassett@uw.edu) and Andone C. Lavery (AOPE, Woods Hole Oceanographic Inst., Woods Hole, MA)

The properties and dynamics of estuarine environments often exhibit strong spatial gradients and rapidly evolving structure that can be difficult to sample and resolve. High-frequency acoustic backscattering techniques spanning from 10 s to 100 s of kilohertz are well-suited for estuarine studies given that they can provide synoptic imaging, aid in discriminating between sources of acoustic backscattering, and in some cases quantify variables of interest. Here, we compare acoustic backscattering measurements obtained using vessel-based, towed, and underwater vehicle operations from the Columbia River, Connecticut River, James River, and Mobile Bay. These data are supported by other oceanographic measurements and circulation models. The sources of backscattering observed across these sites include bubbles, turbulent microstructure, stratification, suspended sediment, and biology. We focus on the quantification of and discrimination between areas dominated by unique processes, in addition to lessons learned regarding frequency selection and operations in estuarine environments.

9:15

3aAO3. Long-term monitoring of a seagrass meadow using wideband acoustic measurements. Megan Ballard (Appl. Res. Labs., Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78665, meganb@arlut.utexas.edu), Kevin M. Lee (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Kyle Capistrant-Fossa (Marine Sci. Inst., Univ. of Texas at Austin, Port Aransas, TX), Preston S. Wilson (Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Andrew R. McNeese (Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX), and Kenneth H. Dunton (Marine Sci. Inst., Univ. of Texas at Austin, Port Aransas, TX)

Seagrasses are sentinel species whose sensitivity to changing water conditions makes them an indicator for sea level rise and climate change. The biological processes and physical characteristics associated with seagrass are known to affect acoustic propagation due to gas bodies contained within the seagrass tissue as well as photosynthesis-driven bubble production that results in free gas bubbles in the water. In this work, acoustical methods are applied to monitor seagrass biomass and gas ebullition with an autonomous field-deployed system using broadband acoustic measurements. Supporting environmental measurements including water temperature and salinity, dissolved oxygen, and photosynthetically active radiation (PAR) were also collected and used to interpret the acoustic data. A ray-based propagation model that includes losses due to the dispersion, absorption, and scattering of sound is applied to relate the measured acoustic signals to the gas bodies in the seagrass tissue and free bubbles in the water. This talk will present preliminary results from the first six months of a year-long deployment of the acoustic system in a dense seagrass meadow dominated by *Thalassia testudinum* (turtle grass) in Corpus Christi Bay, Texas (Gulf of Mexico). [Work supported by NSF.]

Contributed Papers

9:35

3aAO4. Coral reef & temperate coastal soundscape features evident in directional and omnidirectional passive acoustic time series. Lauren A. Freeman (NUWC Newport, 1176 Howell St., Newport, RI 02841, lauren.a.freeman@navy.mil), Simon Freeman (Dept. of Energy, ARPA-E, Portsmouth, RI), Aaron M. Thode (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA), and Philip Caspers (NUWC Newport, Newport, RI)

Biologically complex coastal environments, such as coral reefs, demonstrate an equally rich ambient soundscape. Bioacoustic features of coastal soundscapes are closely tied with relative ecosystem health, functional groups present, and can be linked with specific behaviors. Biological contributions to ambient soundscapes have distinctive qualities as compared to sound associated with physical processes (i.e. wind and wave noise). While some biological components are readily identifiable, such as marine mammal or fish calls, the background noise associated with hundreds of thousands of biological clicks, snaps, and pops is not as well studied but contains a wealth of information about the ecosystem. A 64-element line array with 4.5 kHz design frequency was deployed for several field experiments off the coast of Kona, Hawaii in 2019 and 2020. Soundscape data from Hawaii were compared with comparable omnidirectional time series from Bermuda (2020) and coastal New England rocky reefs (2020–2021). Similarities in certain spectral features associated with biological sound sources were found between these unique ecosystems. The characteristic coral reef evening chorus, or significant increase in sound levels immediately prior to sunset, was consistent in Hawaii and Bermuda with comparable crepuscular changes in coastal New England.

9:50

3aAO5. Acoustic rainfall detection with linear discriminant functions of principal components. C. Mallary (UMass Dartmouth, 285 Old Westport Rd., Dartmouth, MA 02747, cmallary@umassd.edu), C. J. Berg (UMass Dartmouth, North Dartmouth, MA), John R. Buck, Amit Tandon, and Alan Andonian (UMass Dartmouth, Dartmouth, MA)

Ma and Nystuen (2005) pioneered passive acoustic measurement of rainfall rates. This project extends their work with signal processing algorithms exploiting the full frequency band of the acoustic signals. We also extend Schwock and Abadi's order-statistic power spectral density (PSD) estimation for outlier rejection to reject recreational anthropogenic noise sources and reject diurnal biological sources using two hydrophones spaced by 1 m. Ma and Nystuen reduced the data dimensionality by extracting a few discriminant frequencies. Our proposed detection algorithm implements principal component analysis (PCA) to reduce the estimated PSD to two principal components. Linear discriminant analysis (LDA) provides a simple detection statistic from the two dimensional principal components. We evaluated our algorithm on four months of acoustic and meteorological data collected from a dock in New Bedford, MA in shallow water (3 m deep). For 1% false alarms, the proposed PCA/LDA algorithm correctly detected 36% ($\pm$7%) of rain events exceeding 1 mm/hr, including 64% ($\pm$7%) of the rain by volume. Applying Ma and Nystuen's algorithm to the same data set for the same false alarm rate detected 23% ($\pm$11%) of events containing 52% ($\pm$26%) of the rainfall volume. [Work supported by ONR.]

10:05

3aAO6. Improved characterization of ambient sound levels in a coastal environment via use of an unmanned wave glider. Joseph Iafrate (Dept of Navy, 1176 Howell St., Newport, RI 02841, joseph.iafrate@navy.mil), Georges Dossot, and Stephanie Watwood (Dept. of Navy, Newport, RI)

In this project, we use an unmanned Liquid Robotics SV3 wave glider as a tool for basic research studying ambient acoustics in shallower, coastal waters off the coast of East-Central Florida. The primary objective is to characterize sound levels in varying conditions via collection of high-resolution environmental data. The SV3 is used to systematically characterize the acoustic environment in coastal waters, while simultaneously measuring oceanographic parameters such as wind speed, wind direction, and wave height. Quantifying the influence of these fundamental oceanographic influences on shallow water acoustics will provide improved characterization of surface noise and comparison with traditional ambient noise spectra. Ambient noise spectra will be presented for various sea states and wind scenarios, as measured at 25 foot depth using a towfish attached to the SV3. Some of these measurements are also compared to fixed station bottom recordings. Finally, this effort also includes characterization of unique sources of biological sounds recorded by the SV3, including not previously documented loud fish chorusing in this region.

10:20–10:35 Break

10:35

3aAO7. High resolution measurements of the epi- and mesopelagic ocean by a profiling vehicle equipped with environmental sensors and a broadband echosounder. Benjamin D. Grassian (Graduate School of Oceanogr., Univ. of Rhode Island, 215 S Ferry Rd., Narragansett, RI 02882, bgrassian@gmail.com), Chris Roman (Graduate School of Oceanogr., Univ. of Rhode Island, Narragansett, RI), Joseph Warren (School of Marine and Atmospheric Sci., Stony Brook Univ., Southampton, NY), and David Casagrande (Graduate School of Oceanogr., Univ. of Rhode Island, Narragansett, RI)

Collecting detailed surveys of the physical and biological heterogeneity in the epi and mesopelagic ocean is critical to describe the ecosystems within these vast and three-dimensional habitats. Common ocean sampling platforms (e.g., net systems, moored and shipboard sensors) are often unable to resolve marine biota at scales comparable to the variability existing in their physical environment. We have integrated a dual-frequency split-beam echosounder (Simrad EK80 with 70 and 200 kHz transducers) into the Wire Flyer profiling vehicle to achieve concurrent hydrographic and acoustic sections in environments between 0 and 1000 m. The Wire Flyer provides high-resolution repeat profiling (0–2.5 m/s vertical velocity, ~1 km horizontal repeats) within specified water column depth bands typically spanning 300–400 m. This system can provide acoustic backscatter data at depths unavailable to shipboard surveys and can be operated in tandem with shipboard echosounders to provide overlapping acoustic coverage with concurrent hydrographic sections near the surface. The side-looking transducer orientation samples orthogonal to the vehicle's profiling survey path provide a direct measurement of horizontal heterogeneity. The collected data have proven the system's capacity to resolve migrating layers, biological patches, and single targets in the horizontal, rising gas plumes, and scattering layer distributions tightly coupled to submesoscale environmental features.

Invited Papers

10:50

3aAO8. Relating fine-grained sediment consolidation to acoustic seabed surveying. Nina Stark (Civil & Environ. Eng., Virginia Tech, Dept. of Civil and Environ. Eng., 750 Drillfield Dr., 216 Patton Hall, Blacksburg, VA 24061, ninas@vt.edu), Reem Jaber, Liz Smith (Civil & Environ. Eng., Virginia Tech, Blacksburg, VA), Grace Massey (Virginia Inst. of Marine Sci., Gloucester Point, VA), and Joseph Calantoni (Naval Res. Lab, Stennis Space Ctr., MS)

The process of fine-grained seabed sediment consolidation is governed by a variety of factors including grain size distributions, organics, sediment dynamics, and anthropogenic activities (e.g., marine traffic and dredging). Poorly consolidated sediments are typically softer and more susceptible to mobility from shear, while over consolidated sediments are typically stiffer with a higher threshold for mobility from shear. Knowledge of the state of consolidation of seabed sediments is important to assess and maintain navigable depth, predict length scales of seafloor roughness, and assess seabed-object interaction (e.g., anchors, unexploded ordnance). For example, the porosity varies significantly for the same type of sediment for different states of consolidation. Variability is expected and has been qualitatively demonstrated in acoustic seabed surveying including high- and low-frequency techniques. In this study, seabed sediments were characterized using a portable free fall penetrometer along with geotechnical and sedimentological laboratory testing of fine-grained seabed sediments. Testing results quantified both sediment textural (e.g., bulk density, water content, void ratio, and porosity) and strength properties (e.g., undrained shear strength and consolidation state and behavior) and were related to the backscatter intensity observed with different high- and low-frequency off-the-shelf acoustic surveying devices.

11:10

3aAO9. Estimating geoacoustic properties of sediments with a wide grain-size distribution using sediment organic content and the viscous grain shearing model. Gabriel R. Venegas (Ctr. for Acoust. Res. and Education, Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824, g.venegas@unh.edu), Madeline R. Frey (Dauphin Island Sea Lab, Dauphin Island, AL), Kevin M. Lee, Megan Ballard (Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX), W. Cyrus Clemo, and Kelly M. Dorgan (Dauphin Island Sea Lab, Dauphin Island, AL)

Interstitial organic matter (OM) is a ubiquitous constituent of marine sediment, particularly prevalent in sediments with significant silt/clay fractions, and is often quantified across a wide range of disciplines. OM suspends silt and clay particles in the sediment matrix, adsorbs onto mineral surfaces, and resides primarily between mineral contacts, all of which are hypothesized to alter geoacoustic properties, yet OM has not been included in sediment acoustics models. Recent studies have shown that OM content (OC) correlates well with sediment geoacoustic properties, such as porosity and index of impedance, from sediments found in the Baltic Sea to seagrass-bearing sediments in Texas to mud banks in Brazil. In Mobile Bay, Alabama, OC correlated well not only with porosity and index of impedance, but also with grain-shearing coefficients described by the viscous grain shearing (VGS) model. Using empirical regressions derived from a subset of the Mobile Bay biogeoacoustic data and VGS, OC will be used to estimate geoacoustic properties of Mobile Bay sediments with a wide grain-size distribution. The efficacy and potential of using OC to predict sediment geoacoustic properties (and *vice versa*) in Mobile Bay and in other sediments with similar composition will be discussed. [Work supported by ONR.]

Contributed Papers

11:30

3aAO10. Normal incidence reflection coefficient measurements of layered sand-mud sediment and comparison to models. 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), Gabriel R. Venegas (Ctr. for Acoust. Res. and Education, Univ. of New Hampshire, Durham, NH), W. Cyrus Clemo, Kelly M. Dorgan, Madeline R. Frey (Dauphin Island Sea Lab, Dauphin Island, AL), and Preston S. Wilson (Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Depth-dependent variability in seabed acoustic and geophysical properties can be introduced by both physical and biogenic processes. An experiment was conducted near the mouth of Mobile Bay, Alabama to investigate spatial variability over a mud-sand gradient, where sediment properties showed lateral changes on the scales of tens of meters. Additionally, sediment layering on the scale of centimeters was evident in the top 20 cm of the seabed. Normal-incidence seabed reflection coefficient measurements (20–60 kHz) were acquired using a ship-mounted system to survey a 1.9 km^2 area of the seabed, and diver cores were collected for direct analysis of near-surface seabed properties. The cores were acoustically logged to obtain vertical profiles of sound speed and attenuation (10–1000 kHz) and then analyzed for vertical profiles of sediment physical properties (porosity, grain size, organic matter content), infauna community composition, and tensile strength. Physical and acoustic profiles obtained from the cores were then used as inputs to model the *n*-layer reflection coefficient. Comparison of the modeled and measured reflection coefficients will be discussed. [Work supported by ONR.]

11:45

3aAO11. Investigation of wavefield boundary interactions in a prototypical littoral zone. 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, Richard D. Costley (Geotechnical and Structures Lab., U.S. Army Engineer Res. & Development Ctr., Vicksburg, MS), Alanna Lester (Cold Regions Res. and Eng. Lab., US Army Engineer Res. and Development Ctr., Lyme, NH), Abigail Stehno (Costal Hydraulics Lab., US Army Engineer Res. and Development Ctr., Vicksburg, MS), Michael J. White (Construction Eng. Res. Lab., US Army ERDC, Champaign, IL), Christa Woodley, Aaron Urbanczyk (Environ. Lab., US Army Engineer Res. and Development Ctr., Vicksburg, MS), and Mihan McKenna (US Army Engineer Res. and Development Ctr., Vicksburg, MS)

Signals in a littoral environment may be generated and received in any of the three media present: land, air, and water. Interpretation of signals that have crossed media boundaries is challenging, as the preponderance of research has treated media boundaries as lossy reflectors rather than poor transmitters. Full wavefield characterization of broadband pressure signals coupled to the land/soil, air, or water (LAW) is critical to interpreting those signals that have passed through media boundaries. However, there is only a basic understanding of how waves transform as they cross subsurface inhomogeneities and boundaries. This presentation begins with an overview of the new multi-disciplinary effort currently underway at the US Army Engineer Research and Development Center to investigate the wavefield and boundary interactions in a littoral zone environment. The experiment is designed to measure wave transmission across mixed media boundaries in a controlled, constructed, prototype littoral zone. An overview of the design and construction of the test site is provided. The preliminary experimental results are presented and discussed. [Approved for public release: distribution is unlimited.]

Session 3aBA

Biomedical Acoustics and Physical Acoustics: Bubble-Cell Interaction I

Klazina Kooiman, Cochair
Erasmus MC, Wytemaweg 80, Room Ee2302, Rotterdam, 3015 CN, Netherlands

Eleanor P. Stride, Cochair
University of Oxford, Institute of Biomedical Engineering, Oxford OX3 7DQ, United Kingdom

Invited Papers

8:00

3aBA1. Bubble-vessel interactions: From physics to applications in brain disease theranostics. Hong Chen (Washington Univ. in St. Louis, 6338 Washington Ave. University City, MO 63130, chenhongxjtu@gmail.com)

Focused ultrasound combined with microbubbles (FUS + MB) provides a noninvasive and spatially targeted approach for inducing blood-brain barrier (BBB) disruption. Revealing the interactions among ultrasound, bubbles, and blood vessels is fundamental toward understanding this technique's physical mechanism. These interactions in the brain induce various bioeffects, including BBB disruption. FUS + MB-induced BBB disruption enables two-way trafficking between the brain and the bloodstream. While circulating agents can enter the brain through FUS-mediated BBB disruption for the treatment of brain diseases, brain disease-derived biomarkers (e.g., DNA, RNA, and protein) can also be released into the blood circulation to improve the sensitivity of blood-based liquid biopsy. The application of FUS + MB in brain drug delivery has been translated into the clinic with more than 20 clinical trials currently ongoing. The application of FUS + MB for blood-based liquid biopsy, called sonobiopsy, has been demonstrated feasible for the diagnosis of brain cancer. This talk will cover topics ranging from the fundamental physics of bubble-vessel interactions to their applications in the diagnosis and treatment of brain diseases.

8:30

3aBA2. Ultrasound-enhanced drug delivery to spinal cord tumors. Paige Smith, Min Choi, Danielle Charron, Ranjith Ramesh (Sunnybrook Res. Inst., Toronto, ON, Canada), and Meaghan O'Reilly (Sunnybrook Res. Inst., 2075 Bayview Ave., Rm C736a, Toronto, ON M4N3M5, Canada, moreilly@sri.utoronto.ca)

The blood-brain barrier (BBB) and blood-spinal cord barrier (BSCB) are major obstacles to the treatment of CNS diseases and disorders as they prevent the passage of most therapeutic agents from the blood pool to the CNS tissue. Transient, reversible opening of the BBB using ultrasound and microbubbles (MBs) has been extensively studied and has reached clinical investigations. Ultrasound and MBs can similarly modify the BSCB, although applications in the spinal cord have received less attention. We have previously demonstrated ultrasound and MB-mediated opening of the BSCB in rodents, as well as in large animals through the intact spine, using a spine-specific pulse scheme called short-burst phase keying (SBPK). In this talk, we will discuss our work evaluating treatment control approaches for controlling SBPK exposures and will present recent results from drug delivery studies in rodent models of primary and metastatic spinal cord tumors.

Contributed Papers

9:00

3aBA3. Ultrasound-targeted microbubble cavitation increases paracellular gaps in an *in vitro* blood brain barrier model. Grace E. Conway (Univ. of Pittsburgh, 3550 Terrace St., Pittsburgh, PA 15213, gec36@pitt.edu), Anurag N. Paranjape, Xucai Chen, and Flordeliza S. Villanueva (Univ. of Pittsburgh, Pittsburgh, PA)

Background: Ultrasound-targeted microbubble cavitation (UTMC) can transiently open the blood brain barrier (BBB). We sought to determine the timeline of paracellular gap formation after UTMC in an *in vitro* model of the BBB. Methods: We utilized a transwell model with murine brain endothelial cells (EC) and astrocytes on opposite sides of a support membrane. Ultrasound (1 MHz, 10 μs duration, 10 ms pulse interval) at 250 kPa was applied to lipid microbubbles in contact with ECs for 20 s. Z-stacks of transwells acquired by confocal microscopy were converted to maximum intensity projections to quantify number and size of paracellular gaps (NIS-Elements). Endothelial barrier function was assessed using transendothelial electrical resistance (TEER). One-way ANOVA with post-hoc *t*-testing was performed. Results: Compared to no UTMC, there was an increase in the number and size of paraceullar gaps two minutes after UTMC that lasted at least 60 min ($p < 0.05$). The increase in gaps was associated with a decrease in TEER ($p < 0.05$). The number of gaps started to decrease 15 min after UTMC, and total gap area started to decrease 30 min after UTMC ($p < 0.05$). Conclusions: In our model of the BBB, UTMC induces endothelial hyperpermeability, which appears mediated by dynamic paracellular gap formation.

9:15

3aBA4. Microbubble dynamics in brain microvessels at 1 MHz and 330 kHz driving frequencies. James H. Bezer (Bioengineering, Imperial College London, St. Peter's College, Oxford OX1 2DL, United Kingdom, james.bezer@spc.ox.ac.uk), Paul Prentice (Univ. of Glasgow, Glasgow, United Kingdom), William Lim Kee Chang (Bioengineering, Imperial College London, London, United Kingdom), Zheng Jiang (Dept. of Bioengineering, Imperial College London, London, United Kingdom), Sophie Morse, Christopher Rowlands (Bioengineering, Imperial College London, London, United Kingdom), Kirsten M. Christensen-Jeffries (Biomedical Eng. and Imaging Sci., King's College London, London, United Kingdom), Andrei Kozlov, and James Choi (Bioengineering, Imperial College London, London, United Kingdom)

Blood-brain barrier (BBB) disruption using ultrasound-driven microbubbles is a promising technique to deliver drugs to the brain. This study aimed to directly observe behaviours of individual microbubbles within the brain microvasculature when exposed to pulses typical in transcranial ultrasound therapy. Low centre frequencies of 1 MHz and 330 kHz were used, at mechanical indices of 0.2-1, and pulse lengths up to 10 ms. Acute brain slices were obtained from juvenile rats, transcardially perfused with SonoVue, heparin, and dye. In each slice, a suitable bubble in a microvessel (5–15 μm diameter) was observed at both microsecond and millisecond time scales during ultrasound exposure. Oscillating microbubbles cause microvessel walls to distend and invaginate at the ultrasound driving frequency and can cause micrometre-scale tissue displacements well beyond the endothelial wall. Microbubbles can also be forced out of small microvessels due to the primary radiation force; this occurred at both frequencies tested. The probability of extravasation scales approximately with mechanical index, being rare at low pressures, but much more common at MI $\geq$ 0.6. Microbubble extravasation due to the primary radiation force may, therefore, be a mechanism of BBB disruption or of tissue damage. These results may aid development of safer and more effective therapies.

9:30

3aBA5. Focused ultrasound effects on the glioblastoma immune microenvironment. Tao Sun (Radiology, Brigham and Women's Hospital, Harvard Med. School, Tufts Univ., 221 Longwood Ave., EBRC 514, Focused Ultrasound Lab., Boston, MA 02115, taosun@bwh.harvard.edu) and Nathan J. McDannold (Radiology, Brigham and Women's Hospital, Boston, MA)

Focused ultrasound (FUS) is an attractive method of treating neurological diseases, such as glioblastoma (GBM). An accumulating body of evidence has confirmed the efficacy and safety of using FUS with systematically circulating microbubbles, which operate as acoustic enhancers, to temporarily open the blood-brain barrier for drug delivery. However, there is very limited knowledge of how immune cell respond to mechanical stimuli such as FUS, particularly in the complex immune microenvironments of brain tumors. Here, we report FUS immunomodulation effects may also include the enhanced recruitment and activation of other mononuclear phagocytes in murine models of GBM in tests with or without monoclonal antibody-based passive immunotherapy. Flow cytometry, immunohistochemistry, and *in vitro* cell co-culture studies were performed for immunoprofiling and immune effector determination. In murine GBM models, our results demonstrated that FUS enhanced antigen presentation behaviors of tumor-associated macrophages without affecting the microglia. FUS also reprogrammed the macrophages locally towards the anti-cancer phenotype. Taken together, our results offer new evidence in FUS immunomodulation on the myeloid compartment of the brains in GBM mouse models.

9:45

3aBA6. A three-dimensional human brain spheroid model for studying sonoporation-induced drug penetration beyond the blood-brain barrier. Anurag N. Paranjape (UPMC Heart and Vascular Inst., Univ. of Pittsburgh, 3550 Terrace St., Pittsburgh, PA 15213, anuragnparanjape@gmail.com), Leonardo D'Aiuto, Wenxiao Zheng (Dept. of Psychiatry, Univ. of Pittsburgh School of Medicine Western Psychiatric Inst. and Clinic, Pittsburgh, PA), Xucai Chen, and Flordeliza S. Villanueva (UPMC Heart and Vascular Inst., Univ. of Pittsburgh, Pittsburgh, PA)

Ultrasound (US)-targeted microbubble (MB) cavitation (UTMC) facilitates delivery of cell-impermeant drugs across the blood brain barrier (BBB). The effects of UTMC on extravascular cells, and the extent of payload penetration, are unknown. Spheroids were generated using neurons and astrocytes derived from human iPSCs, primary human brain- microglia, endothelial cells (ECs) and pericytes, and placed in a water tank with lipid MBs. US (1 MHz frequency, 250 kPa peak negative pressure, 10 μs pulse duration, 10 ms pulse interval) was delivered for 10 s. Immunostaining showed ECs and pericytes at the spheroid periphery, with high expression of membranous ZO-1. BBB functionality was confirmed with histamine treatment, which increased permeability to Texas red dextran (TRD) (28.8% increase, $p = 0.0015$). TRD penetrated up to 100 μm beyond the BBB upon UTMC (12% increase compared to 0 kPa control). Inhibition of endothelial nitric oxide synthase with L-NAME reduced UTMC-induced TRD penetration (19% decrease, $p < 0.05$). Sonoporation, as shown by uptake of propidium iodide, occurred in cells beyond the BBB. In our novel 3D spheroid model with intact BBB, UTMC caused nitric-oxide dependent BBB breach. This is the first report of UTMC causing "remote" sonoporation of cells not in direct contact with MBs and bears further study.

10:00–10:15 Break

10:15

3aBA7. Assessment of clot degradation under the action of histotripsy and a thrombolytic drug. Samuel A. Hendley (Univ. of Chicago, Chicago, IL) and Kenneth B. Bader (Univ. of Chicago, 5835 South Cottage Grove Ave. Dept. of Radiology, MC 2026, Q301B, Chicago, IL 60637, baderk@uchicago.edu)

Deep vein thrombosis is a major global burden. For critical obstructions, administration of a thrombolytic drug via catheter infusion is the frontline therapy for restoration of flow. Histotripsy is a focused ultrasound therapy utilized for soft tissue ablation via the generation and action of bubble clouds. Thrombolytics have been shown to act synergistically with histotripsy to promote thrombus degradation via two hypothesized mechanisms: fractionation of erythrocytes within the thrombus (hemolysis) and enhanced thrombolytic delivery (fibrinolysis). The objective of this study was to quantify the relationship among histotripsy bubble activity, hemolysis, and fibrinolysis. Human whole blood clots were exposed to thrombolytic and histotripsy pulses. The strength of bubble cloud activity was quantified via passive cavitation images formed from acoustic emissions generated during the histotripsy pulse. Following histotripsy exposure, assays were conducted to quantify the generation of hemoglobin (metric of hemolysis) and D-dimer (metric of fibrinolysis). Using linear regression analysis, our findings indicated hemolysis and fibrinolysis contributed equally to overall treatment efficacy (clot mass loss). Moreover, the force generated by histotripsy bubble active promoted hemolysis or fibrinolysis in an equivalent fashion. These findings demonstrate histotripsy bubble clouds have more utility than ablation alone and reveal the benefits of this combination approach.

10:30

3aBA8. Microbubble-mediated adjunct treatment for cardiovascular bacterial infections. Kirby R. Lattwein (Biomedical Eng. Dept., Erasmus Univ. Medical Ctr., P.O. Box 2040, Rotterdam, Zuid Holland 3000 CA, Netherlands, k.lattwein@erasmusmc.nl), Margot E. Starrenburg, Joop J. Kouijzer, Mariël Leon-Grooters (Biomedical Eng. Dept., Erasmus Univ. Medical Ctr., Rotterdam, Netherlands), Moniek P. de Maat (Hematology Dept., Erasmus Univ. Medical Ctr., Rotterdam, Netherlands), Willem J. van Wamel (Medical Microbiology and Infectious Diseases Dept., Erasmus Univ. Medical Ctr., Rotterdam, Netherlands), and Klazina Kooiman (Biomedical Eng. Dept., Erasmus Univ. Medical Ctr., Rotterdam, Netherlands)

Cardiovascular bacterial infections of tissue and indwelling devices are associated with high mortality rates. Infecting bacteria encase themselves in fibrin-based biofilms that hinder the immune system and antibiotics. In this *in vitro* study, we investigated sonobactericide to treat these biofilms using ultrasound and microbubbles, with and without a thrombolytic agent (rt-PA). Cardiovascular-mimicking biofilms were grown using human plasma inoculated with *Staphylococcus aureus* from an inner heart infection. Treatment consisted of ultrasound (2 MHz, 400 kPa, 100 cycles, every second for 30 s and 1 s continuous wave) every min for 10 min, either alone or with low-dose rt-PA (158 ng/ml), microbubbles, or rt-PA and microbubbles, and was assessed by time-lapse confocal microscopy. Ultrasound-activated microbubbles bent fibrin strands ($\leq$5.2 μm), which returned to original positions ($\leq$25.6 s), and dislodged bacteria from fibrin encasement. Ultrasound alone did not physically alter biofilms. Oscillating microbubbles with rt-PA present led to fibrin strand bending, with and without recovery and eventual breaking, and faster fibrin degradation liberating bacterial aggregates (322 $\pm$ 161 s) than rt-PA alone (518 $\pm$ 86 s; excluding $\sim$45 min from rt-PA exposure until degradation began). These results demonstrate that microbubble-induced effects could aid in the treatment of cardiovascular biofilm infections.

10:45

3aBA9. The spatial distribution of the ligand on targeted microbubbles influences the binding efficacy. Simone A. G. Langeveld (Biomedical Eng., Erasmus MC, Rotterdam, Netherlands), Bram Meijlink (Biomedical Eng., Erasmus MC, Rotterdam, Zuid-Holland, Netherlands), Ines Beekers, Mark Olthof, Antonius F. van der Steen, Nico de Jong (Biomedical Eng., Erasmus MC, Rotterdam, Netherlands), and Klazina Kooiman (Biomedical Eng., Erasmus MC, Wytemaweg 80, Rm. Ee2302, Rotterdam 3015 CN, Netherlands, k.kooiman@erasmusmc.nl)

An important trait of targeted microbubbles is their binding to biomarkers for ultrasound molecular imaging and drug delivery. Using organic solvents in phospholipid-coated microbubble production results in a homogeneous ligand distribution, while dispersion of lipids directly into aqueous medium results in a heterogeneous ligand distribution. In this study, we compared the binding efficacy of $\alpha_\nu\beta_3$-targeted microbubbles with a homogeneous or heterogeneous ligand distribution *in vitro* and *in vivo* using confocal microscopy. For *in vitro* studies, human umbilical vein endothelial cells grown statically and under physiological flow were used. For *in vivo* studies, chicken embryos (day 5) were used. Microbubbles having a homogeneous ligand distribution bound 1.55$\times$ more than microbubbles having a heterogeneous ligand distribution *in vitro* statically, while *in vitro* under flow this was 1.49$\times$ more at 1.25 dyn/cm^2 and 1.56$\times$ more at 2.22 dyn/cm^2; *in vivo* this was 1.25$\times$ more. The dissociation rate *in vitro* was lower for bound microbubbles with a homogeneous than heterogeneous ligand distribution at low shear stresses (1–5 dyn/cm^2). In conclusion, when producing phospholipid-coated targeted microbubbles using organic solvents is preferable over directly dispersing phospholipids in aqueous medium for optimal binding. [Funding by the Phospholipid Research Center (No. KKO-2017-057/1-1) and NWO (VIDI Project No.17543) is gratefully acknowledged.]

11:00

3aBA10. Ultra-high speed quantification of cell strain during cell-microbubble interactions. Oliver Pattinson (Faculty of Eng. and Physical Sci., Univ. of Southampton, University Rd., Southampton, Hampshire SO17 1BJ, United Kingdom, op1g15@soton.ac.uk), Dario Carugo (Dept. of Pharmaceutics, School of Pharmacy, Univ. College London, Southampton, United Kingdom), Fabrice Pierron, and Nicholas Evans (Faculty of Eng. and Physical Sci., Univ. of Southampton, Southampton, United Kingdom)

Interactions between oscillating microbubbles and cells are of fundamental importance in understanding cell behaviour, including mechanotransduction, during therapeutic microbubble treatment. However, it is challenging to quantify cell deformation due to the short time domains at which microbubble-induced deformations occur. Developments in both ultra-high speed imaging and image processing may allow for quantification of cell strain at high temporal and spatial resolutions. Here, we tested the hypothesis that ultra-high speed imaging and digital image correlation could be used to measure and quantify microbubble-induced cell deformation. A hypervision HPV-X camera and a custom-designed, compact acoustic cell-culture device were used together to image interactions between DSPC-microbubbles and MG-63 cells at up to 5×10^6 fps, under ultrasound exposure at 1 MHz. Dynamic cell deformation was measured using digital image correlation with MatchID software. Microbubbles associated with MG63 cells in the acoustic device. Microbubble oscillation resulted in a peak deformation of 350 nm and strain of 5% on the cell during the bubble expansion phase, isolated locally to the point of interaction. These data show that cell deformation can be quantified dynamically during bubble-cell interactions, suggesting that mechanical properties, and potentially corresponding therapeutic effects, can be quantified at high-frequency strain rates.

11:15

3aBA11. Exploration of ultrasound-mediated microbubble-cell membrane interactions using novel protein-loaded microbubbles and their role in immunomodulation. Veerle A. Brans (Inst. of Biomedical Eng., Univ. of Oxford, Inst. of Biomedical Eng. (IBME), Old Rd. Campus Res. Bldg., Roosevelt Dr., Oxford OX37DQ, United Kingdom, veerle.brans@new.ox.ac.uk), Michael Gray (Inst. of Biomedical Eng., Univ. of Oxford, Oxford, United Kingdom), Erdinc Sezgin (SciLifeLab, Karolinska Institutet, Solna, Sweden), and Eleanor P. Stride (Inst. of Biomedical Eng., Univ. of Oxford, Oxford, United Kingdom)

The immune response is governed by the dynamic spatiotemporal regulation of signalling proteins at the interface between immune cells and their targets. The broadened knowledge of immunology and antitumour immune responses has led to the development of a novel cancer therapy avenue: immunotherapy. Elucidating the underlying mechanisms of immunotherapy is, however, crucial as in some patients, and it is accompanied by a characteristic toxicity profile and severe side effects such as autoimmune endocrinopathies. This requires quantitative investigation of cell-cell interactions, including the cell membrane receptors and their molecular behaviour upon cellular contact. This work applies a novel strategy employing protein-loaded microbubbles and confocal fluorescence microscopy to investigate cavitation-induced microbubble-cell membrane interactions. To this end, microbubbles functionalised with a model protein were successfully produced and characterised using nickelated lipids and His-tagged proteins as an efficient tagging strategy. These microbubbles form a novel platform for studying the impact of changing ultrasound parameters on the non-invasive delivery of these proteins to target cells, by means of transfer and/or fusion. This knowledge will be applied to explore efficient targeting strategies for custom-made functionalised bubbles to increase their visibility to immune cells.

11:30

3aBA12. Quantifying the role of microstreaming in sonoporation. Miles Aron (Univ. of Oxford, Oxford, 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 microstreaming has been identified as an important mechanism underpinning sonoporation. The reported evidence includes experimental observation of the permeabilization of non-adherent cells or vesicles

in microstreaming and theoretical and experimental estimates of the fluid shear stress required to induce pore formation in cell membranes and/or the activation of membrane components such as ion channels. The aim of this study was to assess the relative contribution of microstreaming to sonoporation observed *in vitro* with adherent cells, as commonly used in drug delivery studies. Streak velocimetry measurements were performed within a microfluidic cell of the microstreaming around a contrast agent microbubble driven at 0.5 and 1MHz and peak negative pressures up to 1MPa. The results suggest that, for the cell densities and microbubble concentrations typically reported in sonoporation studies, only a very small proportion of cells within a typical culture device would be exposed to sufficiently high shear stresses to be permeabilised. This finding is inconsistent with reported results for drug delivery efficiency and suggests that microstreaming may not be the dominant mechanism of sonoporation in this type of experiment.

WEDNESDAY MORNING, 25 MAY 2022 GOVERNORS SQUARE 10, 8:00 A.M. TO 10:15 A.M.

Session 3aEA

Engineering Acoustics, Structural Acoustics, and Vibration and Physical Acoustics: Smart Metamaterials and Metastructures I

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

Bogdan-Ioan Popa, Cochair
Univ. of Michigan, 2350 Hayward St., Ann Arbor, MI 48109

Chair's Introduction—8:00

Invited Papers

8:05

3aEA1. Active metamaterial for realizing odd micropolarity. Guoliang Huang (Univ. of Missouri, E2410 Lafferre Hall, Mech. & Aerosp. Eng., Columbia, MO 65211, huangg@missouri.edu)

Materials made from active, living, or robotic components can display emergent properties arising from local sensing and computation. Here, we realize a freestanding active metabeam with piezoelectric elements and electronic feed-forward control that gives rise to an odd micropolar elasticity absent in energy-conserving media. The non-reciprocal odd modulus enables bending and shearing cycles that convert electrical energy into mechanical work, and vice versa. The sign of this elastic modulus is linked to a non-Hermitian topological index that determines the localization of vibrational modes to sample boundaries. At finite frequency, we can also tune the phase angle of the active modulus to produce a direction-dependent bending modulus and control non-Hermitian vibrational properties. Our continuum approach, built on symmetries and conservation laws, could be exploited to design others systems such as synthetic biofilaments and membranes with feed-forward control loops.

8:25

3aEA2. Multichannel frequency-selective beaming in time-modulated electroacoustic phased arrays. Mostafa Nouh (Mech. and Aerosp. Eng., Univ. at Buffalo (SUNY), 240 Bell Hall, Attn: Mostafa Nouh, Buffalo, NY 14260, mnouh@buffalo.edu)

Acoustic phased arrays have been a cornerstone of non-destructive evaluation, sonar communications, and medical imaging for years. Conventional arrays work by imparting a static phase gradient across a set of acoustic transceivers to steer a self-created wavefront in a desired direction. In a reception mode, they abide by basic reciprocity principles and exhibit the strongest gain for waves incident from the same direction to which they transmit. This talk sheds light on a class of dynamic phased arrays which is capable of (a) generating multiple scattered harmonics of a single-frequency voltage input which simultaneously propagate in different directional lanes and (b) exhibiting non-identical TX/RX patterns. We will show that such lanes emerge in the form of artificially-synthesized directional channels each with a distinct frequency signature that can be predicted a priori. To achieve this, we devise a class of phase shifters, which augment the array elements with a dynamic phase modulation using an array of piezo-wafer discs bonded to an elastic medium. The scattered beams propagate simultaneously in the transmitted wave field but can be visualized using an FFT of the time-transient measurements via laser Doppler vibrometry. The experimental realization illustrates the array's ability to guide incident waves within tunable frequency channels that are commensurate with the modulation rate and along the intended directions.

8:45

3aEA3. Dynamics of temporally modulated materials: Adiabatic transformations and frequency conversion. Emanuele Riva (Dept. of Mech. Eng., Politecnico di Milano, Via La Masa, 1, Milano 20156, Italy, emanuele.riva@polimi.it)

The recent progress in the context of elastic metamaterials and modulated waveguides with digitally controllable properties has spurred the research in the context of time-varying and space-time varying mechanical systems. In other words, the search of new functionalities, such as nonreciprocity, frequency conversion, parametric amplification, and edge-to-edge pumping, to name a few, requires advanced space-time control of the material parameters, which justifies the emergence of active times in phononics. The work presented herein discusses temporal modulations in the context of stiffness-modulated elastic structures, with emphasis on frequency conversion and wave steering. It is shown that slow temporal modulations, compliant with the adiabatic theorem, can be functionally employed to change the frequency content and the propagation direction of incident wave packets. At the same time, adiabatic variations avoid the scattering of back-propagating waves, which are instead present in case of fast modulations and, in many cases, can be undesired from a practical perspective. Both transient and steady-state behaviors of stiffness-modulated waveguides are discussed with the goal of achieving controllable transmission of elastic signals between an emitter and a receiver.

9:05

3aEA4. Bandgap and mode shape tuning of piezoelectric metamaterial. Amr M. Baz (Mech. Eng., Univ. of Maryland, 2137 Eng. Bldg., College Park, MD 20855, baz@umd.edu)

The mode shape of piezoelectric metamaterials is tuned by manipulating spatially the electrical boundary conditions of the piezo-elements, in a desired and controlled manner, in order to tailor the wave propagation characteristics through these metamaterials. The proposed concept relies on the fact that open-circuit piezo-elements made of lead-zirconate-titanate (*PZT*4) are twice as stiff as the same piezo-elements when operating under short-circuit conditions. Appropriate switching of the boundary conditions of the different piezo-elements between open and short-circuit conditions results in any desirable spatially distribution of the stiffness over the entire metamaterial volume. With such capabilities, it would be possible to control the bandgap characteristics of the metamaterial. But, more importantly, it would also be feasible to alter the mode shape characteristics of the metamaterial in order to control the magnitude and direction of wave propagation. This in effect enables controlling or breaking the reciprocity characteristics of the metamaterial. A finite element model (*FEM*) is developed to model the bandgap and mode shape characteristics of one-dimensional piezo-metamaterial. The effect of various switching strategies on the location and spectral width of the bandgap characteristics is illustrated. Furthermore, the switching strategies are also shown to influence the mode shapes, energy flow, and reciprocity characteristics of the piezo-metamaterial.

9:25

3aEA5. Electromechanical metastructures for simultaneous wave attenuation and energy harvesting. Serife Tol (Mech. Eng., Univ. of Michigan, 2350 Hayward St., Ann Arbor, MI 48109, stol@umich.edu)

Periodic architectures designed with piezoelectric materials are favorable due to their potential to control waves without any need for structural modifications and also due to their multifunctional abilities, such as energy harvesting and vibration mitigation. This talk focuses on the latter and introduces a piezoelectric-based metastructure with broadband capability of low-frequency elastic wave energy conversion. Unlike the phononic crystal concepts consisting of piezoelectric patch arrays with heavy masses or resonance-based piezoelectric cantilever harvester arrays with tip mass attachments used for harvesting standing waves, our goal is to exploit the properties of locally-resonant metamaterials and phononic crystals within the same structure and harvest energy from travelling elastic waves. Specifically, we merge locally resonant and Bragg band gaps to achieve a multifunctional metastructure, which is capable for maximum energy conversion and wave attenuation in a broadband fashion. To this end, we develop a new wave-based fully coupled electroelastic transfer matrix method and study multifunctional harvesting and attenuation performance of the electromechanical metastructure. The theoretical frameworks and the applicability of the proposed metastructure are also validated using a full-scale experimental setup.

Contributed Papers

9:45

3aEA6. Nonreciprocal acoustic scattering from an elastic plate with spatiotemporally modulated material properties. Benjamin M. Goldsberry (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, bgoldsberry@utexas.edu), Samuel P. Wallen, and Michael R. Haberman (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Acoustic and elastic metamaterials with space- and time-dependent material properties have received great attention recently as a means to realize nonreciprocal wave propagation. The nonreciprocal behavior of propagating waves in a spatiotemporally modulated infinite medium is usually characterized by directional bandgaps present in the frequency-wavenumber spectrum. However, less attention has been given to acoustic scattering from spatiotemporally modulated media. In this work, we consider nonreciprocal reflection and transmission from a spatiotemporally modulated, infinite elastic plate excited by a plane wave at oblique incidence. A semi-analytical approach is developed that considers the coupling between the acoustic waves and the displacement of the plate. The reflection and transmission response of the plate for each generated frequency harmonic as a function of the incident angle are reported. Finally, we find conditions on the modulation parameters that yield a large degree of nonreciprocity. The present analysis leads to potential applications in acoustic communications, such as directional wave sensing.

10:00

3aEA7. Polarized source model for the design of bulk active metamaterials. 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 have the potential for the broadband realization of complex property distributions, such as those prescribed by transformation acoustics. Especially promising are metamaterials composed of unit cells consisting of paired sensors and drivers that generate a programmed acoustic response to the local pressure and particle velocity. Previous experimental works have used this approach to implement active metamaterials of theoretically controllable effective bulk modulus and mass density, but were limited to only one or a few non-interacting cells and were not applicable to general bulk media. Here, we further develop some of the theory necessary to overcome these limitations and enable the design of bulk active metamaterials with desired properties, including steep gradients

and mass density anisotropy. We present a highly simplified model of sensor-driver unit cells as interacting polarized sources that allow us to derive closed-form expressions directly relating the effective acoustic properties to the required gains, or polarizabilities. We validate our model for several example geometries and property distributions by comparing the scattered fields of the theoretical metamaterials with those of the equivalent continuous media. In particular, we show compatibility with transformation acoustics and demonstrate cylindrical cloaks using polarized sources.

WEDNESDAY MORNING, 25 MAY 2022 DIRECTORS ROW H, 9:00 A.M. TO 10:20 A.M.

Session 3aMU

Musical Acoustics: General Topics in Musical Acoustics II

Kurt R. Hoffman, Cochair
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Chair's Introduction—9:00

Contributed Papers

9:05

3aMU1. Modeling musician diffraction for artificially excited clarinet directivity measurements. Samuel D. Bellows (Phys. and Astronomy, Brigham Young Univ., N247 ESC Provo, UT 84602, sbellows@byu.edu) and Timothy W. Leishman (Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Directivity measurements of musical instruments have many applications in musical, audio, and architectural acoustics. Typical measurement methods include artificially excited instruments and instruments played by live musicians. While recent advances in directivity measurement techniques enable higher resolutions for played instruments, the results are still limited in bandwidth and repeatability compared with directivity results from artificially excited instruments. However, artificially excited instruments typically neglect musician diffraction and absorption. This work compares possible approaches for representing musician diffraction in artificially excited clarinet measurements to improve their directivity results for room simulations or auralizations.

9:20

3aMU2. Measured high-resolution directivities of guitar amplifiers. Rachel C. Edelman (Phys. & Astronomy, Brigham Young Univ., 475 w 1720 n, Apt 1-303, Provo, UT 84604, rachel.edelman17@gmail.com), Brian E. Anderson (Phys. & Astronomy, Brigham Young Univ., Provo, UT), Samuel D. Bellows (Phys. & Astronomy, Brigham Young Univ., UT), and Timothy W. Leishman (Phys. & Astronomy, Brigham Young Univ., Provo, UT)

Guitar amplifiers comprise specialized powered loudspeakers with distinct configurations and directivity characteristics. Recently, the authors measured the directivities of six guitar amplifiers using a new measurement system. The system employs a rotating semicircular microphone array in an anechoic chamber, with 2522 unique microphone positions over a surrounding sphere (i.e., 5-deg resolution in both the polar and azimuthal angles). This presentation will describe the measurement and processing procedures and compare the measured results to those produced by theoretical directivity models. An online archival database provides easy access to the guitar amplifier data and other directivity data from the human voice, played musical instruments, and other sound sources. The data are available in several commonly used file formats and should provide valuable information for many applications, including spatial audio, architectural acoustics modeling, and microphone placement.

9:35

3aMU3. Augmenting a single-point laser Doppler vibrometer to perform scanning measurements. Mark Rau (Music, Stanford Univ., 660 Lomita Court, Stanford, CA 94305, mrau@ccrma.stanford.edu), Julius O. Smith (Music, Stanford Univ., Stanford, CA), and Doug L. James (Comput. Sci. and Music, Stanford Univ., Stanford, CA)

Laser Doppler vibrometers (LDV) are used for non-contact vibration measurements of various structures and are frequently used for stringed instrument measurements. Single-point LDVs can be used with the roving hammer or LDV method for mode shape measurements, but this is time-consuming and requires constant attention. Scanning LDVs exist but are expensive and often out of reach of musical acoustics researchers. An inexpensive apparatus to modify a common single-point LDV such that it can perform automated scanning measurements is presented. The augmentation consists of a mirror galvanometer, impact hammer controller, and 3D printed mounting hardware. The scanning system is controlled by a microprocessor and can be easily automated. The total cost of the system, excluding the LDV and impact hammer, is under two hundred dollars. Measurements of guitars are presented to validate the scanning system and discuss any shortcomings.

9:50

3aMU4. A quantitative assessment of uncertainty in the measurement of violin impact response. Seth Lowery (Phys., Central Washington Univ., 400 E. University Way, Dept. of Phys., Ellensburg, WA 98926, seth.lowery@cwu.edu) and Andrew A. Piacsek (Phys., Central Washington Univ., Ellensburg, WA)

It is a commonly stated belief among violin players and luthiers that new violins require a period of "playing in" for the tone to develop. Several studies have worked towards an answer to this question, such as measuring the change in tone according to the human ear or the vibrational response of stimulated wood. As the effects of sustained excitation on the mechanical response of violins will likely be subtle, it is necessary to create a consistent method of measurement and to quantify the expected range of deviation among repeated measurements. We measure admittance (velocity/force) by tapping the bridge with a small modal impact hammer and recording the velocity response of the top plate near the opposite side of the bridge using a laser doppler vibrometer. The acoustic response is also measured in an anechoic chamber. Since measurements of the same violin on different days do not produce identical response curves, several methods of characterizing the deviation were developed and compared. These uncertainty metrics will be used in the second phase of the experiment to determine the significance of the results, and ultimately work towards a better understanding of the effects of breaking in violins.

10:05

3aMU5. Timbral effects the Paulstretch audio time-stretching algorithm. Colin Malloy (Music, Univ. of Victoria, 3330 Richmond Rd., Victoria, BC V8P 4P1, Canada, malloyc@uvic.ca)

The Paulstretch algorithm is a procedure for achieving aurally pleasing extreme time-stretches while avoiding the "phasiness" issues common to phase-vocoder-based implementations. The algorithm accomplishes this by randomizing the phase information where a typical phase-vocoder approach would perform phase-unwrapping or another method to preserve phase alignment. When performed without time-stretching, phase randomization is often referred to as whisperization. This allows for more extreme time-stretches than other approaches. Where most audio time-stretching is usually used for small adjustments, Paulstretch is regularly used to stretch audio by a factor of 5, 10, 20 or much more. This effect is popular for its aesthetic timbral effects and is regularly used in soundscapes, film/television scoring, and more. When time-stretching is performed at such extremes, however, it is unclear how randomizing the phase, different FFT window sizes, the stretch factor, and other variables combine to affect the reconstructed audio output. This study employs multiple audio analysis methods to examine the spectral, timbral, and perceptual effects of the Paulstretch algorithm.

WEDNESDAY MORNING, 25 MAY 2022 GOVERNORS SQUARE 11, 8:05 A.M. TO 12:00 NOON

Session 3aPA

Physical Acoustics and Noise: Meteorological Acoustics I

Roger M. Waxler, Chair
Univ. of Mississippi, P.O. Box 1848, University, MS 38677

Contributed Papers

8:05

3aPA1. Azimuthal deviation for outdoor sound propagation in a moving medium with a crosswind. David Norris (National Security Solutions, ENSCO, Inc., 4849 N. Wickham Rd., Melbourne, FL 32940, norris.david@ensco.com) and Vladimir Ostashev (Cooperative Inst. for Res. in Environ. Sci., Univ. of Colorado Boulder, Hanover, NH)

Current codes for ray tracing in a moving atmosphere involve solutions of a set of differential equations. In this presentation, ray tracing is suggested, which is based on a closed-form expression for the sound propagation path in a stratified moving atmosphere. This approach can complement existing ray tracing codes, perform with improved computational efficiency, and serve as a benchmark solution. The closed-form expression for the sound propagation path enables other analytical formulations in geometrical acoustics such as the prediction of the azimuthal deviation as driven by the crosswind. This result also allows estimation of the crosswind provided that the azimuthal deviation between the true and apparent source location is measured. The result is validated by numerical simulations of sound propagation in the atmospheric surface layer with different meteorological regimes.

8:20

3aPA2. On the possibility of electrostatically generated infrasound during thunderstorms. Roberto Sabatini (Embry-Riddle Aeronautical Univ., 1 Aerosp. Blvd., Daytona Beach, FL 32114, sabatinr@erau.edu), Jeremy Riousset (Florida Inst. of Technol., Melbourne, FL), Jonathan B. Snively (Embry-Riddle Aeronautical Univ., Daytona Beach, FL), and Ningyu Liu (Univ. of New Hampshire, Durham, NH)

Acoustic emissions from lightning discharges are usually attributed to two mechanisms. The audible part of their spectrum mainly results from the shock wave generated by the heating of the lightning channel. On the other hand, the infrasonic component is associated with the conversion to sound of the electrostatic energy stored in the thundercloud. While there is a broad scientific consensus on the former mechanism, the latter process is still controversial. The electrostatic mechanism was proposed by C. T. R. Wilson in 1921 and can be summarized as follows. Due to the electrostatic repulsion of the charged particles, the pressure within a thundercloud is lower than outside. At discharge, the electrostatic field collapses, and the sudden air volume contraction produces an acoustic pulse. This work examines the existing theoretical models of the electrostatic mechanism with the data observed for thundercloud dimensions, charge densities, and total charges. Our results show that, although possible, this mechanism, as currently described, cannot explain observations. Our findings might support the hypothesis that the heating of the lightning channel is responsible for the generation of both infrasound and audible sound. However, a definitive answer remains precluded and requires a better understanding of cloud formation and electrification processes.

Invited Papers

8:35

3aPA3. Anomalous vertical sound propagation and the ground-blocking of turbulence. D. Keith Wilson (U.S. Army Engineer Res. and Development Ctr., U.S Army ERDC-CRREL, 72 Lyme Rd., Hanover, NH 03755-1290, D.Keith.Wilson@usace.army.mil), Vladimir Ostashev, Matthew J. Kamrath, and Carl R. Hart (U.S. Army Engineer Res. and Development Ctr., Hanover, NH)

Scattering by turbulence leads to random scintillations and coherence loss of sound waves, among other impacts. In a recent experiment involving transmissions from a 130-m tower to microphones on the ground [Kamrath *et al*., J. Acoust. Soc. Am. 149(3), 2055–2071 (2021)], the turbulence impacts on phase variances were shown to be considerably smaller for vertical and slanted paths than theoretically predicted. The primary cause is likely *ground blocking* of the turbulence: since the sound waves are impacted most strongly by the turbulent velocity fluctuations in the direction of propagation, and the ground blocks large vertical motions, the turbulence impacts are relatively weakened for vertical paths. An approach is described for including the ground blocking in the turbulence spectrum and, thus, in acoustic scattering calculations. The anisotropic propagation effect could possibly be used to measure the relative strengths of vertical and horizonal velocity fluctuations in the near-ground atmosphere.

8:55

3aPA4. The use of infrasound from repeating explosion sequences in Oklahoma to probe the atmosphere. Stephen Arrowsmith (Southern Methodist Univ., 819 Lake Terrace Dr., Dallas, TX 75218, sarrowsmith@smu.edu), Gil Averbuch, Miro R. Giannone (Earth Sci., Southern Methodist Univ., Dallas, TX), and Roberto Sabatini (Embry-Riddle Aeronautical Univ., Daytona Beach, FL)

Infrasound waves are sensitive to atmospheric processes that are unresolved in numerical weather prediction models, and to atmospheric properties at altitudes that are poorly constrained by observations. We report on the use of infrasound observations to probe atmospheric phenomena occurring over temporal scales from less than one hour to days, and at altitudes that extend up to the stratopause. Two measurement campaigns were conducted to capture infrasound signals from repeating explosion sequences from the McAlester Army Ammunition Plant in Oklahoma. The first campaign captured infrasound signals propagating in a tropospheric waveguide, while the second captured signals propagating in a stratospheric waveguide. In both cases, we resolve sub-hour variations in infrasound signal characteristics that we show are related to unresolved atmospheric phenomena. We further resolve day-to-day variations for the two different waveguides and assess how well these longer-term variations can be explained using meteorological models. Using a normal-mode code based on a Chebyshev collocation method, we also compared ground infrasonic recordings and numerically synthesized pressure waveforms. The results provide insight into the value that infrasound measurements can provide for meteorology over different temporal and spatial scales.

9:15

3aPA5. Microbarometer measurements to probe turbulent kinetic energy in the atmospheric boundary layer. Jelle D. Assink (R&D Seismology and Acoust., KNMI, Utrechtseweg 297, Utrecht 3731GA, Netherlands, jelle.assink@knmi.nl)

Microbarometer networks are typically in place for the detection of atmospheric infrasound waves. The monitoring of such waves can be of interest for source characterization, which include a wide array of geophysical and man-made sources, as well as atmospheric infrasound propagation studies. In such studies, the presence of turbulence and other wind-induced effects are considered a nuisance. For this reason, wind noise filters are typically in place for suppression. As Cuxart *et al*. [BLM (2013)] pointed out, microbarometer measurements indeed have a value for detection of local and remote turbulence and measurements can be used to estimate turbulent kinetic energy (TKE). In this study, we compare microbarometer observations at the Cabauw atmospheric research site to co-located in-situ TKE measurements. Moreover, we compare turbulence timeseries to predictions from the high-resolution HARMONIE weather model in which this quantity is parameterized. This approach has two foreseen applications: (1) modeled TKE fields could possibly help in identifying regions that are most appropriate for infrasound monitoring due to low TKE values and (2) microbarometer observations could possibly be of use in the further refining of the modeled TKE parameter.

9:35

3aPA6. Wind, sea swell, and excitation of atmospheric waveguides by underwater earthquakes. Oleg A. Godin (Phys. Dept., Naval Postgrad. School, Phys. Dept., Naval Postgrad. School, Monterey, CA 93943, oagodin@nps.edu)

Powerful underwater sources, such as volcano eruptions and large explosions, can generate infrasonic waves, which remain detectable at distant receivers on shore. Evers *et al*. [Geophys. Res. Lett. **41**, 1644–1650 (2014)] described observations of acoustic signals in air from the 2004 Macquarie Ridge submarine earthquake, by the infrasonic array IS05AU of the International Monitoring System. Infrasound propagated in the tropospheric and stratospheric waveguides and was received 1325 km from the epicenter. Specific physical mechanisms of the excitation of guided infrasonic waves by underwater sources are not fully understood. Here, we emphasize the potential role of local meteorological conditions. Scattering of ballistic waves from the earthquake epicenter by wind waves and sea swell on the ocean surface is a potent mechanism of excitation of normal modes in hydroacoustic waveguides (T-waves) [Godin, J. Acoust. Soc. Am. **150**, 3999–4017 (2021)]. We investigate the hypothesis that the scattering of ballistic waves at the ocean surface also explains the earthquake excitation of acoustic normal modes in the tropospheric and stratospheric waveguides. Relative significance of the surface

scattering and the previously proposed evanescent coupling mechanism is studied as a function of the wind speed, wave frequency, and the source depth.

9:55

3aPA7. Infrasound generated by meteorological fronts: Observations and modeling. 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 results of study of the parameters of infrasound waves (amplitudes, coherences, grazing angles, azimuths, and horizontal phase speeds) detected during a passage of warm and cold fronts through the networks of microbarometers in Moscow region are presented. The observed effect of internal gravity waves on the coherence and phase speed of infrasound from meteorological fronts is analyzed. The differences found in the time dependences of the parameters of infrasound from warm and cold fronts that must be taken into account when detecting infrasound precursors of atmospheric storms are discussed. A possible mechanism for the generation of infrasound by the turbulent airstream flowing around the geometric irregularities of the meteorological front is proposed. [This work was supported by the Russian Science Foundation, Grant No. 21-17-00021.]

10:15–10:30 Break

10:30

3aPA8. Comparison of infrasound emissions from tornadic and non-tornadic storms. Garth Frazier (NCPA, Univ. of Mississippi, NCPA, University of MS, P.O. Box 1848, Oxford, MS 38677, frazier@olemiss.edu), Roger M. Waxler (Univ. of Mississippi, University, MS), Carrick Talmadge (NCPA, Univ. of Mississippi, Oxford, MS), Claus Hetzer (NCPA, Univ. of Mississippi, Tempe, AZ), and Hank Buchanan (NCPA, Univ. of Mississippi, Oxford, MS)

At a previous ASA meeting (Spring 2019), some results corresponding to application of maximum likelihood estimation with the complex Wishart distribution to direction-of-arrival of infrasound from tornadic storms were presented. In this presentation, updated results are presented and compared with estimation of direction-of-arrival of infrasound from two non-tornadic storms that were measured with the same infrasound arrays. One of these storms was similar to the tornadic storm in that it consisted of numerous isolated storm cells with frequent lightning. The other storm was significantly different in that it was a well-organized squall line that produced only limited lightning. The tornadic storm yielded very well-defined direction-of-arrival tracks in contrast to the non-tornadic storms.

Contributed Paper

10:50

3aPA9. Signal versus noise: Propagation modeling applied to the detection of the tornadoes produced in Northern Alabama during the passage of a storm front in March, 2018. Roger M. Waxler (NCPA, Univ. of Mississippi, P.O. Box 1848, University, MS 38677, rwax@olemiss.edu), Garth Frazier (NCPA, Univ. of Mississippi, Oxford, MS), and Claus Hetzer (NCPA, Univ. of Mississippi, Tempe, AZ)

On March 19, 2018, a storm front that produced at least eight confirmed tornadoes passed across northern Alabama. During that period, we had a network of infrasound arrays deployed throughout the region. Infrasonic signals were detected from each of the confirmed tornadoes; however, detections were not made on all of the arrays simultaneously. To understand why certain tornadoes were, or were not, detected on a given array, an infrasound signal propagation analysis was undertaken. A parabolic equation (PE) propagation algorithm was employed with atmospheric temperature and wind profiles input from the weather research and forecasting (WRF) model. To complement the propagation studies, a signal verses noise analysis was performed for each array. Taking the noise analysis into account, the correlation between detection versus non-detection and the predictions of the propagation model was quite good.

Invited Papers

11:05

3aPA10. How does knowledge of acoustics guide the parameterizations of gravity waves? Christophe Millet (CEA, CEA, DAM, DIF, Bruyères-le-Châtel 91297, France, christophe.millet@cea.fr), Francois Lott (LMD, PSL Res. Inst., ENS, Paris, France), and Alvaro de la Camara (Dept. Earth Phys. and Astrophysics, Universidad Complutense de Madrid, Madrid, Spain)

Describing the statistics of gravity wave (GW) fields represents a major motivation for both current research on atmospheric GWs and long-range infrasound propagation. In practice, the probability density functions (PDF) of the momentum fluxes are estimated combining observations, numerical modelling, and theory. Numerical models (such as WRF) show that the PDFs vary in a robust way relative to the background local wind speed. For non-orographic GWs, these PDFs are approximated as lognormal distributions, with characteristics found to depend on the background wind speed. Studies show that some trends are not observed using a state-of-the-art stochastic parameterization of GWs, unless the phase velocities of GW sources (essentially tropospheric) are dramatically changed. As the vertical wavelength and the phase velocity are related to each other, such changes also affect the interaction between infrasound and lower-stratospheric GWs. Consequently, significant efforts have been made to use ground-based acoustic sensors for characterizing the GW sources, including the use of neural networks. This approach provides a promising way to describe the statistics of GW sources from ground-truth infrasound events and an additional constraint to tune stochastic parameterizations of GWs.

11:25

3aPA11. Modeling of acoustic and gravity wave interactions, coupling, and observables above meteorological systems. Jonathan B. Snively (Ctr. for Space and Atmospheric Res., Embry-Riddle Aeronautical Univ., 1 Aerosp. Blvd., Daytona Beach, FL 32114, snivelyj@erau.edu), Roberto Sabatini (Ctr. for Space and Atmospheric Res., Embry-Riddle Aeronautical Univ., Daytona Beach, FL), Donna A. Calhoun (Dept. of Mathematics, Boise State Univ., Boise, ID), Christopher J. Heale, Pavel A. Inchin, and Matthew D. Zettergren (Ctr. for Space and Atmospheric Res., Embry-Riddle Aeronautical Univ., Daytona Beach, FL)

Meteorology, especially strong tropospheric convection, is widely appreciated to generate broad spectra of acoustic and gravity waves (AWs and GWs or, together, AGWs). These include GWs with scales of tens to hundreds of kilometers and periods of ~5 min to hours, that readily propagate upward, reach high altitudes (often to the lower-thermosphere), and grow to large amplitudes so that they may evolve nonlinearly prior to being overcome by dissipation. Strong convective dynamics, e.g., thunderstorms and tornadoes, are also known to radiate AWs at very low infrasonic frequencies (e.g., 0.1 Hz down to ~4 mHz) that reach high altitudes and may be detectable in fluctuations of the atmosphere and ionosphere [e.g., Nishioka *et al*. (2013); Heale *et al*. (2019)]. The breaking of strong GW fields may also generate secondary AWs and, more generally, AGWs [Snively (2017); Heale *et al*. (2021)]. Together, convection and secondary AGW processes contribute to a broad spectrum of AWs with ~mHz periods that are readily detectable at high altitudes and in pressure signals also measured at ground. Although AWs are excluded from traditional numerical weather prediction models, we report on models and simulation experiments designed to capture AW/AGW evolutions and their resulting observable signatures. In particular, we review and highlight scenarios by which ~mHz AWs may reveal source processes of interest, as well as the opportunities to use atmospheric and ionospheric signals of AWs/AGWs as complement to ground-based infrasound recordings.

Contributed Paper

11:45

3aPA12. Lighthill's theory applied to tornadoes as an infrasonic source. Bin Liang (Univ. of MS, 145 Hill Dr., Oxford, MS 38677-1848, bliang@go.olemiss.edu), Roger M. Waxler (Univ. of Mississippi, University, MS), and Paul Markowski (Penn State Univ., University Park, PA)

It has been shown that tornadoes radiate infrasound, which can be detected at great distances. The physical mechanism underlying the radiation is not well understood. In this work, we use Lighthill's acoustic analogy to predict the pressure signals emitted from numerically simulated tornadoes. Several simulations will be considered including simulations produced using an LES model and provided by researchers from the Penn State Department of Meteorology and Atmospheric Science. The acoustical source produced from the LES tornado simulations shows interesting patterns in a narrow low-frequency band. By comparing different numerical models of tornadoes, we hope to determine what in a tornado is causing the infrasound radiation.

Session 3aPPa

Psychological and Physiological Acoustics: Binaural Hearing

Daniel J. Tollin, Cochair
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Mathieu Lavandier, Cochair
University De Lyon Entpe, Lyon France

Chair's Introduction—7:50

Contributed Papers

7:55

3aPPa1. Front-back reversals for tones heard in a room by moving listeners. William Hartmann (Michigan State Univ., 749 Beech St., East Lansing, MI 48823, wmh@msu.edu), Jack C. Magann (Johns Hopkins Univ. Appl. Phys. Lab, Laurel, MD), and Eric Macaulay (Michigan State Univ., East Lansing, MI)

The tone localization experiments reported by Macaulay and Hartmann [J. Acoust. Soc. Am. **149**, 4159–4179 (2021)] were highly biased in favor of sources in front of the listener. Nevertheless, listeners sometimes localized tones in back, both when they were stationary and when they were required to move. The experiments provided extensive data relevant to the matter of in-back localization, because they measured head location and orientation as well as ear canal signals throughout each experiment trial. Because the experiments were done in a room, the data were irregular functions of source azimuth. Analysis revealed the following effects: For stationary trials, back responses were strongly correlated with weaker tone levels in the ear canals for 500, 2000, and 4000 Hz. Correlation was greatly reduced at 1000 Hz. That result is consistent with head-related transfer functions, where ear-canal intensity is greater for sounds in front except near 1000 Hz. For moving trials, back responses at 500 and 1000 Hz were strongly correlated with the slopes of interaural level differences and interaural time differences as functions of head-pointing direction, as naturally occur in free field. Correlations decreased dramatically at higher frequencies.

8:10

3aPPa2. Is acoustic space "learned" in the buildup of the precedence effect? M. Torben Pastore (College of Health Solutions, Arizona State Univ., PO Box 870102, ASU, Tempe, AZ 852870102, m.torben.pastore@gmail.com), William A. Yost (College of Health Solutions, ASU, Tempe, AZ), and Yi Zhou (Speech and Hearing Sci., Arizona State Univ., Tempe, AZ)

Most of what is known about sound source localization in reverberant environments rests upon knowledge of how the auditory system processes a pair of brief stimuli presented over headphones that simulate a direct sound and a single reflection. In everyday environments, however, sound sources often emit relatively continuous sounds or repeat them often in succession—for example, speech. Also, listeners often move. Dynamic changes in source/listener positions may present a unique opportunity to investigate the effects of "learning" that seem to result in the "buildup of the precedence effect—listeners' ability to effectively localize reverberant sounds after repeated presentation that would be poorly localized after only one presentation. The implicated learning may constitute learning of the spatial acoustics of the environment or learning of the temporal order and timing of reflections (or both). In this study, we sought to disambiguate these two possible learning strategies by comparing listener behavior in response to presentation of repeated pairs of lead/lag noise stimuli presented in a soundfield. Listeners' perceived sound-source localization and the rate of fusion were measured.

8:25

3aPPa3. A sex difference and learning in the discrimination of interaural-level differences but not interaural-time differences in naïve listeners. Beverly A. Wright (Commun. Sci. and Disord., Northwestern Univ., 2240 Campus Dr., Northwestern University, Evanston, IL 60208, b-wright@northwestern.edu) and Huanping Dai (Dept. of Speech, Lang., and Hearing Sci., Univ. of Arizona, Tucson, AZ)

The two primary cues to sound-source location on the horizontal plane are interaural differences in time (ITDs) and in level (ILDs). Here, we asked whether the ability to discriminate small changes in each of these interaural cues differs between the sexes. We tested males and females who had no prior experience with any psychoacoustic task on either ILD discrimination at 4 kHz or ITD discrimination at 0.5 kHz. For ILD discrimination, the overall mean threshold, as well as the threshold for each 50-trial block, was significantly lower for males ($n = 80$) than for females ($n = 166$). Both males and females learned over the course of testing, but there was no sex difference in learning rate. In contrast, for ITD discrimination, thresholds did not differ significantly between the sexes for the overall mean or for any block ($n = 43M/94F$). There also was no learning across blocks for either group. For both tasks, the individual thresholds spanned a wide range in each group. The presence of sex differences and learning for ILD but not ITD discrimination suggests that the factors responsible for these outcomes influenced a relatively peripheral ILD-specific pathway, and not an ITD-specific pathway, or a more central locus that is not cue specific.

8:40

3aPPa4. A novel paradigm to define functional spatial boundaries. Erol J. Ozmeral (Commun. Sci. and Disord., Univ. of South Florida, 4202 E. Fowler Ave., Tampa, FL 33620, eozmeral@usf.edu) and Nathan Higgins (Commun. Sci. and Disord., Univ. of South Florida, Tampa, FL)

One approach to studying an individual's binaural function is to measure speech reception ability for spatially separated compared to co-located competing speech streams. This spatial-release-from-masking (SRM) task has been widely adopted due to its capacity to detect abnormal binaural function, but it has its limitations in interpreting individual spatial abilities. Here, we introduce a modified SRM task that directly measures the needed spatial separation between target and maskers to achieve a specific SRM.

Stimuli were presented from 24 speakers in the free field using an equal power panning method. Normal-hearing listeners ($n = 10$) were tasked with identifying digits in a target stream of monosyllabic words always presented from the front. Initial measures were made using the classic SRM paradigm in which two opposite-gender masker streams were either presented from the front (co-located), or symmetrically from 30° (fixed-separation), to determine a speech reception threshold (SRT; in dB target-to-masker ratio). Based on the co-located SRT, the task then adaptively tracked the separation angle needed to achieve 6-dB SRM. Results showed for most listeners, a 10-dB SRM at 30° separation and an average spatial boundary less than 30 degrees when fixed at 6-dB SRM. [Work supported by NIH NIDCD R21DC017832 to EJO.]

8:55

3aPPa5. Visual capture and changes in response times in multisensory localization tasks. Colton Clayton (Commun. Sci. and Disord., Grand Valley State Univ., 975 S. Mytle Ave. Tempe, AZ 85287, ctclayto@asu.edu) and Yi Zhou (Speech and Hearing Sci., Arizona State Univ., Tempe, AZ)

In the real-world, environmental objects are often both seen and heard. Visual stimuli can influence the accuracy, variability, and timing of the listener's responses to auditory targets. One well-known example of this visual influence is the Ventriloquist Illusion or visual capture of the perceived sound source location. However, less is known about how vision affects the timing of sensorimotor output driven by auditory events. We hypothesize that response time may manifest the influences of two different visual factors—stimulus features (e.g., spatial congruence) and environmental features (e.g., contextual reference). We measured eye saccades, a natural orienting behavior, in response to auditory and visual stimuli with different degrees of spatial alignment in two visual environments (with or without contextual reference). Results showed both visual factors can influence the timing of saccades to auditory targets. Response time was longer when auditory, and visual signals are spatially separated. Response time was also longer in the non-referenced than referenced environments, but the effect size was small. These results demonstrate response time as a valuable metric for understanding multisensory perception. Further studies are needed to investigate the patterns of changes in both response time and response accuracy to gain new insights into auditory localization in real-life settings.

9:10

3aPPa6. Spectral weighting functions for localization of complex sound: Effects of sensorineural hearing loss. Monica L. Folkerts (Hearing and Speech, Vanderbilt Univ., 1215 21st Ave. South, Nashville, TN 37232, monica.l.folkerts@vanderbilt.edu), Erin M. Picou (Hearing and Speech, Vanderbilt Univ., Nashville, TN), and G. Christopher Stecker (Ctr. for Hearing Res., Boys Town National Res. Hospital, Omaha, NE)

When localization cues are distributed across the spectrum of a complex sound, listeners with normal hearing (NH) utilize cues near the 600-800 Hz "ITD dominance region" [Bilsen and Raatgever, Acustica **28**, 131–132 (1973)]. Recent work has quantified this pattern using spectral weighting functions (SWFs) [Folkerts and Stecker, ARO **42**, 505 (2018)] in NH listeners. However, the salience of the ITD dominance region in listeners with sensorineural hearing loss (SNHL) remains unexamined. Folkerts and Stecker [ARO **45**, 171 (2022)] used simulated SNHL in NH listeners to measure impacts of elevated hearing thresholds, loudness recruitment, and reduced high-frequency ITD sensitivity [Steinberg and Gardener, J Acoust. Soc Am **9**, 11–23 (1937)] on SWFs. The current study measured SWFs in listeners with bilateral sloping high-frequency SNHL. Direct comparisons to SWFs measured with simulated SNHL test the hypothesis that reduced audibility of high-frequency components shifts the ITD dominance region to lower frequencies (i.e., 400–600 Hz). Individual SWFs measured in listeners with SNHL are compared to their headphone and free field thresholds to assess whether audibility can account for weighting changes. [Work supported by NIH R01-DC016643.]

9:25

3aPPa7. A creeping-wave elliptic-head model that explains non-monotonic envelope interaural time differences. Paul G. Mayo (Hearing and Speech Sci., Univ. of Maryland-College Park, 0100 LeFrak Hall, 7251 Preinkert Dr., College Park, MD 20742, paulmayo@umd.edu), Andrew D. Brown (Speech and Hearing Sci., Univ. of Washington, Seattle, WA), and Matthew J. Goupell (Hearing and Speech Sci., Univ. of Maryland-College Park, College Park, MD)

Interaural time and level differences (ITDs and ILDs) created by interactions of impinging sound with the head facilitate horizontal-plane sound localization. ILDs are known to be non-monotonic functions of azimuth, resulting in predictable mislocalization of tonal sources at some frequencies, while ITDs are typically construed as monotonic functions of azimuth. However, recent measurements using a binaural mannequin revealed envelope ITDs (ENV-ITDs) that were non-monotonic functions of azimuth, with the extent of nonmonotonicity dependent on carrier frequency. This study proposes that delayed sound traveling around the back of the head causes this phenomenon. ENV-ITDs created by a time-dependent model of incident sound waves propagating around the front and back of an elliptical head were compared to ENV-ITDs computed using publicly available HRTF libraries for a broadband speech excitation signal. Results from the model align with data from the HRTF libraries, with nonmonotonicities in the ENV-ITD-azimuth functions occurring at nearly identical azimuths and frequencies. These data suggest the veracity of this phenomenon and may have implications for spatial hearing abilities. They also suggest that time-independent models of ITDs that only consider frequency and incident angle, such as spherical head or circular arc length models, could be updated to include such time dependence.

Session 3aPPb

Psychological and Physiological Acoustics: Clinical Populations and Devices II

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Chair's Introduction—9:55

Contributed Papers

10:00

3aPPb1. Spectral processing deficits appear to underlie developmental language disorders. Susan Nittrouer (Speech, Lang., and Hearing Sci., Univ. of Florida, 1225 Ctr. Dr., Gainesville, FL 32610, snittrouer@phhp.ufl.edu), Joanna Lowenstein, Kayla Tellez, Priscilla O'Hara, and Donal Sinex (Speech, Lang., and Hearing Sci., Univ. of Florida, Gainesville, FL)

More than 50 years of research has suggested something is awry in the auditory functioning of children with language disorders, yet there remains no consensus on exactly what the problem is. To address this gap in knowledge, we have proposed a model termed the Auditory Bases of Phonological Acquisition. This model is based on the language feature known as dual patterning, which describes two levels of structure: one consisting of meaningful words and how they can be combined and the other consisting of word-internal, meaningless units and how they can be combined. We hypothesized that acquisition of this second level—phonological structure—is most affected by deficits in auditory functions, especially those involving the spectral domain. To test this hypothesis, we assessed children with and without language disorders between 7 and 10 years old on their abilities to detect spectral and temporal modulation and to perform relevant language tasks. The largest group differences were found at younger ages for spectral modulation detection (SMD). SMD thresholds were also found to correlate most strongly with language measures, especially those based on phonological structure. Outcomes suggest that keen sensitivity to spectral structure in the speech signal is necessary for acquiring precise phonological representations.

10:15

3aPPb2. Factors influencing performance on the modified rhyme test by noise-exposed service members. Douglas S. Brungart (Walter Reed, 4401 Holly Ridge Rd., Rockville, MD 20853, dsbrungart@gmail.com)

Two speech-in-noise experiments using stimuli from the modified rhyme test (MRT) were conducted on service members (SMs) in conjunction with their annual hearing tests. The first experiment, involving roughly 3200 SMs, compared performance in the standard diotic headphone-presented MRT test to the performance level that could be achieved if the same MRT stimuli were presented from a single wall-mounted speaker in one of two different clinical sound booths. The second experiment, involving roughly 2000 SMs, examined performance in a headphone-presented MRT paradigm where the target talker was presented monaurally and the ITD of the masker alternated between -800 μs and $+800$ μs at a 4 Hz rate. Pure-tone audiometric thresholds, N_0S_π detection thresholds, and subjective surveys of hearing difficulty and noise exposure were collected for all participants in both studies. The results show that MRT scores and response times both vary systematically with subjective hearing difficulty, suggesting that a composite time-accuracy score might be the most effective way to evaluating performance on the MRT. The results also provide insights into the hearing difficulties frequently experienced by noise- and blast-exposed service members with normal or near-normal hearing thresholds. [The views expressed in this abstract are those of the author and do not reflect the official policy of the Department of Defense or U.S. Government.]

10:30

3aPPb3. The effect of masker type on the sensitivity of the digits in noise test for detecting hearing loss. Coral E. Dirks (National Military Audiol. & Speech Pathol. Ctr., Walter Reed National Military Medical Ctr., 4954 N Palmer Rd., Bethesda, MD 20889, coral.e.dirks.ctr@mail.mil), Joshua G. Bernstein, and Douglas S. Brungart (National Military Audiol. & Speech Pathol. Ctr., Walter Reed National Military Medical Ctr., Bethesda, MD)

The Digits-in-Noise (DIN) test, a quick hearing-loss screening tool with high test-retest reliability and good sensitivity and specificity, can substitute for an audiogram when calibrated audiometry is unavailable. Traditionally, listeners hear three digits in speech-shaped noise, but other masker types might more effectively distinguish normal hearing thresholds (NHTs) from elevated hearing thresholds (EHTs). This study compared the ability of different energetic and informational maskers to detect low-frequency pure-tone averages >20 dB HL. First, DIN perception was adaptively measured ($N = 1149$ subjects) to determine the 80% correct speech-reception threshold (SRT80) for each of six masker types: female, male, and multi-gender multitalker babble; positive and negative Schroeder-phase complex tones; and speech-shaped noise. Then, for each masker type, a different group of subjects was tested at the corresponding SRT80. As expected, preliminary results suggest that a positive Schroeder-phase masker produces the greatest differences in DIN scores between NHTs and EHTs. Sensitivity and specificity rates for EHT detection for each masker, along with intersubject variability and order effects, will be explored. [The views expressed in this abstract are those of the authors and do not reflect the official policy of the Department of Army/Navy/Air Force, Department of Defense, or U.S. Government.]

10:45

3aPPb4. Contralateral unmasking for single-sided-deafness cochlear-implant users with shifted frequency assignments to reduce interaural place mismatch. Joshua G. Bernstein (National Military Audiol. & Speech Pathol. Ctr., Walter Reed National Military Medical Ctr., 4954 N Palmer Rd., Bethesda, Bethesda, MD 20889, joshua.g.bernstein.civ@mail.mil), Megan M. Eitel, Sandeep A. Phatak, Kenneth Jensen, Elicia M. Pillion, Coral Dirks (National Military Audiol. & Speech Pathol. Ctr., Walter Reed National Military Medical Ctr., Bethesda, MD), and Matthew J. Goupell (Hearing and Speech Sci., Univ. of Maryland-College Park, College Park, MD)

For cochlear-implant (CI) users with single-sided deafness (SSD), standard clinical programming yields interaural place-of-stimulation mismatch, because the electrode array does not reach the apex. This mismatch might degrade spatial-hearing abilities. This study examined whether acutely presented alternative frequency-to-electrode assignments ("remapping"), designed to reduce mismatch, could improve the use of two ears together to perceptually separate competing talkers. Remapped frequency assignments were derived from computed-tomography scans of intracochlear electrode locations or psychophysical tuning curves for interaural time-difference discrimination. Contralateral unmasking was measured by presenting target speech (closed-set corpus) to the acoustic ear and two same-sex competing talkers to just the acoustic ear or to both ears. Preliminary results ($N = 8/15$ planned subjects) show that for seven subjects with small ($\leq$3-dB) initial binaural benefit, remapping yielded a small but significant (0.5-dB mean) increase in binaural benefit. Remapping was detrimental for the one subject with large (6-dB) initial binaural benefit. Possible longitudinal effects and tradeoffs with other SSD-CI hearing benefits that could be affected by remapping are discussed. [The views expressed in this abstract are those of the authors and do not reflect the official policy of the Department of Army/Navy/Air Force, Department of Defense, or U.S. Government.]

11:00

3aPPb5. Comparison of laboratory and clinical computational metrics of acoustic measures of hearing-aid processed speech. Emily Lundberg (Speech, Lang., and Hearing Sci., Univ. of Colorado, 409 UCB, Boulder, CO 80309, Emily.Lundberg@colorado.edu), Ramesh Muralimanohar (Speech, Lang., and Hearing Sci., Univ. of Colorado, Portland, OR), James M. Kates, and Kathryn Arehart (Speech, Lang., and Hearing Sci., Univ. of Colorado, Boulder, CO)

Computational auditory metrics are used to characterize hearing-aid fittings in the laboratory and the clinic. The purpose of this study is to compare the speech intelligibility index (SII) as a clinical metric to the laboratory based SII and the hearing-aid speech perception index (HASPI) and hearing-aid speech quality index (HASQI) laboratory metrics. These comparisons are drawn from a comprehensive dataset of hearing-aid fittings for 120 hearing-aid recipients from a hospital-based audiology clinic. Hearing-aid devices are drawn from multiple manufacturers and multiple technology levels, and a total of nine hearing-aid devices are included. Acoustic recordings are made with the hearing-aids positioned on the KEMAR manikin and include multiple hearing-aid settings for each patient (manufacturer's recommended fit, first fit by the audiologist, and the final fitting selected by the patient). Metric values are computed for each of the settings, and these metric comparisons highlight the advantages and challenges of each metric in characterizing hearing-aid signal processing. In addition, the results provide insight regarding hearing-aid fittings from the manufacturer's fitting versus clinician/patient driven fittings. [Work supported by NIH R01 DC012289, NIH NRSA T32DC012280, and GN Hearing OCG6790B.]

11:15

3aPPb6. Deep neural network models of speech-in-noise perception for hearing technologies and research. Agudemu Borjigin (Purdue Univ., 715 Clinic Dr., West Lafayette, IN 47907, aagudemu@purdue.edu), Kostas Kokkinakis (MED-EL, Durham, NC), Hari Bharadwaj (Purdue Univ., West Lafayette, IN), and Josh Stohl (MED-EL, Durham, NC)

Widespread adoption of artificial intelligence has yet to occur in the hearing field. Hearing technologies, such as cochlear implants (CIs), provide limited benefits of noise reduction, even with current state-of-the-art signal processing strategies. Recent developments in machine learning have produced deep neural network (DNN) models achieving remarkable performance in speech enhancement and source separation tasks. However, there are currently no commercially available CI audio processors that utilize DNN models for noise suppression. Furthermore, the current research community lacks a computational tool to match the complexity of natural auditory processing. To address these gaps, we implemented two DNN models: a recurrent neural network (RNN)—a lightweight template model for speech enhancement, and the SepFormer—the current top-performing speech-separation model in the literature. The DNN models resulted in significant improvements in terms of objective evaluation metrics, as well as intelligibility scores obtained with CI users at different signal-to-noise ratios. Given their flexibility and good performance on complex tasks, these models can also be used to generate hypotheses about speech-in-noise perception and serve as richer substitutes for models commonly used in research. This work serves as a proof-of-concept and a guide for the next steps towards integrating DNN technology into hearing technologies and research.

11:30

3aPPb7. Development of an electromechanical test system and acoustical metrics to predict impacts of hearing protection devices on sound localization. Theodore F. Argo (Rocky Mountain Div., Appl. Res. Assoc., Inc., Littleton, CO), David A. Anderson (Rocky Mountain Div., Appl. Res. Assoc., Inc., 1250 S Monaco PkwyUnit 42, Denver, CO 80224, danderson@ara.com), Andrew D. Brown (Speech and Hearing Sci., Univ. of Washington, Seattle, WA), Nathaniel Greene (Otolaryngol., Univ. of Colorado School of Medicine, Aurora, CO), and Jennifer Jerding (Rocky Mountain Div., Appl. Res. Assoc., Inc., Littleton, CO)

Hearing protection devices (HPDs) such as earplugs and earmuffs can protect users from dangerously high acoustical pressures but also distort cues important for the spatial localization of sounds, likely through spectral distortions associated with occlusion of the pinnae. Evaluation of the degree to which an HPD distorts localization cues is essential for critical situations where users must be protected from high pressures but maintain spatial awareness. Automated testing offers several prospective advantages over human subject testing including cost, speed, and repeatability. In this paper, we describe an electromechanical system using a rotating manikin test fixture and a speaker on a track to simulate a hemispheric speaker array. The sound source localization impact of an HPD is estimated by comparing test signals recorded by the manikin with and without an HPD in place. Test results, as a function of sound source azimuth and elevation in the virtual speaker array, and as an average across locations, are shown for a variety of HPDs. Results are compared to an initial set of parallel measurements of sound localization during HPD use in human subjects.

11:45

3aPPb8. Turn the music down! Repurposing assistive listening broadcast systems to remove nuisance sounds. Ryan M. Corey (Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, 456 Coordinated Sci. Lab, 1308 West Main St., Urbana, IL 61801, corey1@illinois.edu) and Andrew C. Singer (Elec. and Comput. Eng., Univ. of Illinois, Urbana, IL)

Many public venues, such as theaters and houses of worship, use assistive listening systems that broadcast a copy of the sound being played over a public address system to the listening devices of patrons with hearing loss. While these wireless systems are generally intended to amplify sound for listening device users, they can also be used to cancel nuisance sounds, such as loud background music that makes conversation difficult. In this work, we demonstrate a binaural listening device that uses an assistive listening system transmitter and receiver to attenuate music played through loudspeakers while preserving speech sounds from nearby talkers. A set of adaptive filters tracks the acoustic transfer functions from the loudspeakers to the earpiece microphones, predicts the music signal received by each earpiece, and subtracts that predicted signal from the mixture signal of the corresponding earpiece. The binaural adaptive system tracks changes in the unwanted music signal as the user moves while preserving the acoustic and spatial cues of the desired speech sounds. [Work supported by the Intelligence Community Postdoctoral Research Fellowship Program, National Science Foundation, and Discovery Partners Institute.]

Session 3aSA

Structural Acoustics and Vibration, Computational Acoustics, and Engineering Acoustics: Dynamic Substructuring Techniques and Their Application to Structural Acoustics

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

Matthew Luu, Cochair
Penn State, State College, PA 16803

Chair's Introduction—10:00

Invited Papers

10:05

3aSA1. A dynamic substructuring technique for coupling and decoupling structure-cavity systems. Ryan Schultz (Sandia National Labs., PO Box 5800, M.S. 0557, Albuquerque, NM 87185, rschult@sandia.gov) and R. Benjamin Davis (Univ. of Georgia, Athens, GA)

Flexible structures adjacent to acoustic cavities are commonly encountered. In some instances, coupling between the structure and cavity causes non-negligible changes to the system frequency response. Here, we describe the use of modern dynamic substructuring techniques in coupling and decoupling structure-cavity systems. As input, free-interface *in vacuo* natural frequencies and modes of the structure are used along with an uncoupled modal representation of the cavity. In the acoustic model, the fluid-structure interface is typically modeled as pressure release, though faster frequency convergence is possible by enriching the representation with rigid-wall modes. The subsystems are assembled in modal coordinates with continuity at the interface enforced via Lagrange multipliers. This approach circumvents the calculation of coupling coefficients, and results in real-valued system natural frequencies and modes; however, the approach presents challenges when large numbers of interface degrees of freedom are present. Attempts to overcome these challenges with interface reduction techniques are described. Finally, this research led to the derivation of a formula approximating the *in vacuo* natural frequencies of test structures that are well coupled to adjacent acoustic cavities. The formula was successfully applied to an air-filled cylindrical test structure for which the frequency response was confounded by structure-cavity coupling.

10:25

3aSA2. Vibroacoustic subtractive modelling: Principle of the reverse condensed transfer function approach and application to the prediction of the noise radiated by a partially coated cylindrical shell. Florent Dumortier (Laboratoire Vibrations Acoustique, INSA Lyon, 25bis Ave. Jean Capelle, Villeurbanne 69100, France, florent.dumortier@insa-lyon.fr), Laurent Maxit (Laboratoire Vibrations Acoustique, INSA Lyon, Villeurbanne, France), and Valentin Meyer (Naval Group Res., Ollioules, France)

Substructuring methods are commonly used to solve vibroacoustic problems, as they allow studying complex systems by dividing them into smaller subsystems that can be studied separately. If they are generally used to couple subsystems, we are interested here in the decoupling procedure, consisting in subtracting a subsystem from a global system. Decoupling procedures can be used to perturbate an existing model in order to investigate the impact of a default on a given system, or to study problems with complex geometries, starting from a canonical system. In the framework of subtractive modelling, the reverse condensed transfer function (rCTF) method has been developed to decouple vibroacoustical subsystems that are initially coupled along lines or surfaces. The aim of this approach is to predict the response at any point of the final decoupled subsystem. The method will be presented on an academic test case consisting in the scattering problem of a plane wave by a rigid sphere immersed in an infinite water medium. Then, an example of industrial application will be proposed by studying the acoustic radiation of a partially coated cylindrical shell using the rCTF method.

10:45

3aSA3. A structural modification analysis framework leveraging the Lagrange multiplier frequency based substructuring Method. Anthony L. Bonomo (Naval Surface Warfare Ctr., Carderock Div., 9500 MacArthur Blvd., West Bethesda, MD 20817, anthony.bonomo@gmail.com) and Robert T. Cullen (Naval Surface Warfare Ctr., Carderock Div., West Bethesda, MD)

In 2006, de Klerk, Rixen, and de Jong introduced the Lagrange multiplier frequency based substructuring method (LM FBS) as an alternative to the classical frequency based substructuring method (FBS). LM FBS has several advantages over FBS, as it easier to implement and more efficiently handles problems with more than two substructures. LM FBS has also been shown to be quite adaptable and has recently been extended to include flexible connection elements at the interfaces between substructures. One other advantage of the LM FBS that has not been fully exploited to date is the fact that the dimension of the mobility matrix of the coupled system remains the same as the block-diagonal uncoupled system. This feature allows the LM FBS to be more easily interpreted in terms of structural modification and offers the analyst the option to use the normal modes of the uncoupled system as a basis for tracking and quantifying the influence of substructure coupling on the dynamic response of the coupled system. This talk will demonstrate such a structural modification framework utilizing LM FBS.

11:00

3aSA4. Substructured optimization for a speakerphone design task. Matthew Luu (Appl. Res. Lab., Penn State Univ., 446 Bluecourse Dr (Apt907), State College, PA 16803, mbl5743@psu.edu) and Andrew S. Wixom (Appl. Res. Lab., Penn State Univ., State College, PA)

This work explores the use of substructures and parallelized optimization techniques in order to design speakerphone casings, the same problem considered by Berggren *et al*. [in 10th World Congress on Structural and Multidisciplinary Optimization (2013)]. For this work, a simplified model consisting of a 1D beam model with enforced vibration at one end is used to represent the supporting structure of the speaker and microphone of the device. The thickness of the supporting structure is varied in order to reduce coupling between the speaker and microphone. Minimization of the structural vibration in the microphone region over frequencies between 300 and 3400 Hz is examined for optimization. A substructuring approach using spectral elements and Legendre polynomials for the thickness profile reduces the computation cost such that many evaluations of the model may be obtained in a reasonable time. The Python optimization library PyGMO is used to distribute optimization tasks over multiple CPUs in order to further accelerate the design process. The approach is shown to successfully optimize a thickness distribution targeting the frequencies of interest while reducing computation costs and is compared to previously published results for this example problem.

11:15

3aSA5. Radiation of thin complex plates estimated with the landscape of localisation theory. Carlos García A. (Roberval (Mech., energy and electricity), Ctr. de recherche Royallieu, Université de technologie de Compiègne, Rue du Docteur Schweitzer, Compiègne 60200, France, carlos.garcia@utc.fr), Nicolas Dauchez, and Gautier Lefebvre (Roberval (Mech., energy and electricity), Ctr. de recherche Royallieu, Université de technologie de Compiègne, Compiègne, France)

The landscape of localisation is a practical tool that enables the prediction of the geographical localisation of localized modes and helps us to understand the transition between localized and delocalised states. Moreover, approximations based on the Rayleigh quotient and on a variant of Weyl’s law are employed to predict the eigenfrequencies for the Schro¨dinger operator in quantum mechanics, but they are also valid for the Laplace and biharmonic operators, which characterize the behaviour of most dynamical systems in acoustics and vibrations. When studying the acoustic radiation from a vibrating structure, three global parameters are key indicators: the mean squared velocity, the acoustic radiated power, and the radiation efficiency. The literature on this subject is very vast for the plate case, where for simple geometries, it is still possible to derive analytical solutions or, at least, very useful approximations. For more complex structures, numerical simulations seem to be appropriate for lack of a simpler solution. In this context, this work aims to give some light to create a direct relationship between these global parameters and the landscape of localisation function, based on the multipolar radiation behaviour presented by localized modes and estimated by geometrical means.

Session 3aSC

Speech Communication: Children's Speech Intelligibility I

Pasquale Bottalico, Cochair
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Mary M. Flaherty, Cochair
Speech and Hearing Science, University of Illinois at Urbana-Champaign, 901 S Sixth Street, Champaign, IL 61820

Invited Papers

8:30

3aSC1. Effects of F0 contour depth differences on school-age children's speech recognition in a two-talker masker. Mary M. Flaherty (Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, 901 S Sixth St., Champaign, IL 61820, maryflah@illinois.edu), Kelsey Libert (Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, Champaign, IL), and Emily Buss (Univ. of North Carolina-Chapel Hill, Chapel Hill, NC)

This study investigated the extent to which children (ages 5–17) can take advantage of differences in fundamental frequency (F0) contour between target and masker speech to improve speech-in-speech recognition. A number of studies show that differences in F0 and F0 contour between talkers support better speech-in-speech recognition for adults, suggesting it is an important cue for recognizing speech in multitalker environments. However, children's use of F0 contour differences during speech-in-speech recognition is less well understood. In the present study, sentence recognition was measured adaptively in a two-talker speech masker. Target and masker sentences were recorded with either normal, flat, or exaggerated F0 contours. Results revealed that children's sentence recognition was impacted by differences in F0 contour depth in both target and masker speech, but the pattern of results differed between children and adults in some conditions. For children, the consistent nature of the flat F0 contour often had a greater impact on speech recognition thresholds (SRTs) than target/masker differences in F0 contour depth. Observed age effects did not appear to be due to limitations in children's ability to use F0 contour differences in general but are likely related to the perceptual salience of the target contour relative to the masker.

8:50

3aSC2. The use of piecewise linear regression to explore development of children's listening-in-noise ability. Rochelle S. Newman (Hearing & Speech Sci., Univ. of Maryland, 0100 Lefrak Hall, Dept of Hearing & Speech Sci., College Park, MD 20742, rnewman1@umd.edu), Emily Shroads (Hearing & Speech Sci., Univ. of Maryland, College Park, MD), and Monita Chatterjee (Ctr. for Hearing Res., Boys Town National Res. Hospital, Omaha, NE)

To explore the development of speech-in-noise comprehension, we tested children 15–48 months in a preferential-looking paradigm; children saw two images and were told which to attend to (*"Find the keys!"*). The target voice was masked either by a 9-voice blend (primarily energetic masking) or a single talker (providing informational masking; IM). We used piecewise linear-mixed-effects modeling to identify phases of developmental growth and their corresponding changepoints. Based on 51 participants to date, performance in multitalker babble was best described by a linear model with gradual improvement over time, but the 1-voice-masker condition was well-fit by a combination of three phases: a period of rapid growth until 25.3 months, followed by a period of relative stagnation until 43.4 months, then a second growth period. Bonino *et al.* (2021) argued that substantial development in IM ability occurs prior to 30 months, which corresponds well with our first phase. The second growth period may represent an increase in the ability to integrate "acoustic glimpses"; prior studies found that 5-year-olds, but not 30-month-olds, can restore interrupted speech, and our elbow at 43 months fits with this timeline. This demonstrates the usefulness of piecewise linear-mixed-effects as an approach to studying development.

9:10

3aSC3. Pediatric speech understanding in noise and the role of working memory andcomprehension. Jessica Sullivan (Communicative Sci. Disord., Hampton, 100 signature way, 823, Hampton, VA 23666, jessica.sullivan@hamptonu.edu)

Working memory is vital to a child's cognitive processing and speech understanding in noise. More recent findings suggest that the relationship between speech recognition in noise and auditory comprehension may be capturing difficulties in understanding speech in noise which have been linked to cognitive and linguistic skills. Children's auditory-cognitive performance is highly sensitive to processing load. The extent that children engage cognitive resources depends on demands of task difficulty and intensity level of four-talker babble. The combined costs of performing a difficult task in adverse acoustical conditions are greater than the sum of the individual costs. This suggest that the relationship between speech recognition in noise and auditory comprehension may be capturing difficulties

in language which may have effects in academic success. Data will be presented (1) show that audibility alone does not account for difficulties in speech perception in noise (2) showing the complex relationship between working memory and auditory comprehension. (3) Implications for clinical speech in noise screener for children. While app-based interventions for speech and language exist, to our knowledge, none explicitly screen for perception of speech in the presence of noise.

9:30

3aSC4. Effects of two-talker child speech on novel word learning in preschool-age children. Tina M. Grieco-Calub (Rush Univ. Medical Ctr., 4711 Golf Rd., Ste. 1100, Skokie, IL, tina_griecocalub@rush.edu), Katherine R. Gordon, Stephanie L. Lowry (Boys Town National Res. Hospital, Omaha, NE), and Diana M. Cortez (Rush Univ. Medical Ctr., Skokie, IL)

Preschool-age children's recognition of speech is more susceptible to the presence of background noise than that of older children. Because younger children rely on access to speech to learn their native language, children who are poorer at resolving degraded speech may be at risk for delayed vocabulary development. To test this relation directly, we are implementing a retrieval-based word learning task in preschool-age children. Children are taught four novel disyllabic words either in a quiet condition or in the presence of two-talker child speech (TTS) at $+10$ dB-signal-to-noise ratio. Children's performance is quantified by the number of words and the number of phonological features recalled immediately after training and after a 5-min delay. Children also complete tests of familiar word recognition in TTS, verbal working memory, and vocabulary knowledge. In this presentation, we will compare children's word-learning outcomes between the quiet and TTS conditions and evaluate the influence of children language skills on their performance. We predict that children will have greater phonological precision of the novel words in quiet than in the presence of TTS. We also predict that children's verbal working memory skills and vocabulary knowledge will relate to greater learning, regardless of listening condition.

9:50

3aSC5. School-age children show poor use of spatial cues in reverberation for speech-in-speech perception. Z. Ellen Peng (Boys Town National Res. Hospital, 555 North 30th St., Omaha, NE 68131, ellen.peng@boystown.org)

Understanding speech is particularly difficult for children when there is competing speech in the background. When the target and masker talkers are spatially separated, as compared to co-located, the access to corresponding auditory spatial cues can provide release from masking, resulting in an intelligibility gain for speech-in-speech perception. When tested in free-field environments, previous work showed that children demonstrate adult-like spatial release from masking (SRM) by 9–10 years of age. However, in indoor environments where most critical communications occur such as classrooms, reverberation distorts the critical auditory spatial cues that lead to reduced SRM among adults. Little is known about how children process distorted auditory spatial cues for SRM to aid speech-in-speech perception. In this work, we measure SRM in children in virtual reverberant environments that mimic typical learning spaces. We show free-field measurements overestimate SRM maturation in realistic indoor environments. Children show a more protracted development of SRM in reverberation, likely due to immaturities in using distorted auditory spatial cues.

10:10–10:30 Break

10:30

3aSC6. Intelligibility of dysphonic speech in primary schools. Silvia Murgia (Dept. of Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, 901 South Sixth St., Champaign, IL 61820, smurgia2@illinois.edu), Taylor Mekus, and Pasquale Bottalico (Dept. of Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, Champaign, IL)

School children need clear auditory signals and low background noise to learn. When classroom acoustics are poor, teachers often compensate by raising their voices, usually with limited effect against background noise, and, long-term, this makes vocal overuse the primary cause (60%) of the high prevalence of voice problems in teachers. Speech intelligibility tests were performed in primary schools with normal hearing students using words produced by an actor with normal voice quality and simulating a dysphonic voice. The speech was played by a Head and Torso Simulator. Artificial classroom noise and classrooms with different reverberation times were used to obtain a range of Speech Transmission Index from 0.2 to 0.7 (from bad to good). Results showed a statistically significant decrease in intelligibility when the speaker was dysphonic with a maximum of 15% intelligibility loss. This study extends an important pairing of problems related to student learning: classroom acoustics and teachers with voice disorders. It provides important insights into the enormous variability in speech intelligibility in classrooms by characterizing students' intelligibility when students receive degraded auditory input. The degraded auditory input results from the intersection of classroom acoustics and poor teacher voice quality

10:50

3aSC7. Classroom acoustics and the inclusion of hard of hearing children, helping the data be heard. Frank Iglehart (None, 14 Ryans Hill Rd., Leverett, MA 01054, frank.iglehart@gmail.com), Cheryl DeConde Johnson (The ADE-vantage Consulting, Leadville, CO), and Stephen Wilson (Quality Control Architect, The Collaborative, Ann Arbor, MI)

Data on speech perception in children have led to acoustic accommodations in built schools for children with typical hearing but, despite compelling data, not for children who are hard of hearing. The ways to meet the acoustic needs of hard of hearing children are well researched and established in standards. After many years, however, the message is still not disseminating to architects, school districts, and building officials, and thus rarely makes its way into classroom construction. To help bring classroom accessibility to hard of hearing children, a team from the fields of acoustical engineering, architecture, and audiology is using speech perception data and computer simulations to promote inclusion in school design. This multi-year, multi-disciplinary effort began with earlier federal and foundation fundings for research on speech perception in hard of hearing children, which led to development of a new voluntary acoustic standard for schools by the American National Standards Institute. The goal of this work now is to get this standard into the hands of architects and school districts, and ultimately into building codes to have classroom acoustics designed for all students including those hard of hearing.

11:10

3aSC8. Do cochlear-implanted children benefit from voice cue differences for understanding speech in speech masker? Deniz Baskent (Otorhinolaryngology, Univ. of Groningen, Univ. Medical Ctr. Groningen (UMCG), Hanzeplein 1, PO BOX 30.001, Groningen 97100RB, Netherlands, d.baskent@rug.nl)

Differences in voice cues of mean fundamental frequency (F0) and vocal-tract length (VTL) between a target and masker speech can help a listener to segregate the two, and enhance speech intelligibility. Postlingually deafened and implanted adult CI users have difficulties in perceiving F0 and VTL, and this possibly contributes to challenges in understanding speech in presence of a speech masker. Previous research showed that speech-on-speech understanding is not yet adult-like in normal-hearing (NH) children of school age, as this skill seems to develop over many years. Yet, like NH adults, NH children do seem to benefit from the voice cue differences in some situations. In the case of CI children, a number of factors could influence intelligibility of speech with speech maskers. Prelingually deafened and implanted CI children develop auditory skills primarily based on spectrotemporally degraded speech input, which could lead to different developmental patterns compared to NH children. On the other hand, the CI children would benefit from the optimal brain plasticity period for learning, which could lead to different perceptual patterns compared to CI adults. In this talk, I will present data from multiple projects with children and adults at our lab to investigate these potential expectations.

11:30

3aSC9. How do noise and face masks affect speech intelligibility for cochlear implanted children? Giuseppina E. Puglisi (Dept. of Energy, Politecnico di Torino, Torino, Torino, Italy), Michele Di Iulio (Dept. of Energy, Politecnico di Torino, Torino, Italy), Pasquale Bottalico (Dept. of Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, 901 South Sixth St., Champaign, IL 61820, pb81@illinois.edu), Silvia Murgia (Speech and Hearing Sci., Univ. of Illinois - Urbana Champaign, Champaign, IL), Patrizia Consolino (Ospedale Martini di Torino, Torino, Italy), Massimo Spadola Bisetti (Audiologia e Foniatria, Azienda Ospedaliera Universitaria Città della Salute e della Scienza di Torino, Torino, Italy), Giuseppe Pitta (CIAO Ci Sentiamo - ONLUS, Torino, Italy), Louena Shtrepi, and Arianna Astolfi (Dept. of Energy, Politecnico di Torino, Torino, Italy)

Due to the recent pandemic event related to COVID-19, many countries are mandating the use of face masks to reduce the spread of the virus. Unfortunately, those masks can have a detrimental effect on speech communication, particularly for vulnerable listeners like children with hearing loss. This work investigates which type of face mask yields the highest level of speech intelligibility for children with cochlear implants. Fourteen children with cochlear implants aged 7–15 years have been involved, together with a control group of six normal-hearing children. Online speech intelligibility tests (simplified Italian matrix sentence test) were performed testing different masks at different signal to noise ratios. The listening difficulty was also evaluated on a 5-points scale. The results of this study showed which type of face masks are recommended to maximize intelligibility for children with cochlear implants.

Session 3pAA

Architectural Acoustics, ASA Committee on Standards, Biomedical Acoustics, Physical Acoustics, and Noise: Advanced Measurement and Modeling of Sound Absorption and Scattering II

Mélanie Nolan, Cochair
School of Architecture, Rensselaer Polytechnic Institute, Greene Bldg., 110 8th Street, Troy, NY 12180

Ning Xiang, Cochair
School of Architecture, Rensselaer Polytechnic Institute, 110 Eighth Street, Troy, NY 12180

Peter D'Antonio, Cochair
RPG Acoustical Systems LLC, 99 South Street, Passaic, NJ 07055

Contributed Papers

1:30

3pAA1. Modeling parallel arranged transparent micro-perforated panels to enhance low-mid frequency sound absorption. Semiha Yilmazer (Purdue Univ., Ray W Herrick Lab 177, S Russell St., West Lafayette, IN 47907, syilmaze@purdue.edu), Ela Fasllija (Interior Architecture and Environ. Design, Bilkent Univ., Ankara, Cankaya, Turkey), and Cengiz Yilmazer (CSY R&D and Architecture Eng., West Lafayette, IN)

Passive absorbers such as fibrous or porous materials are indispensable for high-frequency damping noises; still, problems are increasingly found at low frequencies. Resonant structures, such as membranes, Helmholtz resonators, and micro-perforated panels (MPPs), are usually used for treating the latter issues. These reactive structures are mass-spring systems with damping to absorb the system's resonant frequency, offering only narrowband absorption. For the sake of broadening the absorption bandwidth, a series of structures based on MPP networks in series and parallel have been explored to introduce multiple resonances. The current study examines the computational modeling of four parallel arranged transparent MPPs with different hole sizes, perforations, and back cavities. A prototype of this absorber is machined by laser and CNC techniques and tested in an impedance tube. The experimental data obtained from using the transfer function are in good agreement with the absorption performance calculated by the electro-acoustic equivalent circuit model (ECM). Results show a wider half absorption bandwidth of the overall system for more than 3 octaves (from 160 Hz to 1120 Hz). The initial findings enhance the possibility of designing single-layer absorbers that are wideband, tunable, and free from porous/fibrous materials.

1:45

3pAA2. Modeling of reverberation chambers for sound absorption measurements using a diffusion equation. Jiahua Zhang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St., Graduate Program in Architectural Acoust., Troy, NY 12180, zhangj45@rpi.edu), Mélanie Nolan, Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY), and Juan Navarro (Polytechnic Sci. Dept., UCAM Universidad Católica de Murcia, Murcia, Spain)

The sound field in reverberation chambers used for measuring sound absorption coefficients is usually non-diffuse, leading to inaccuracies and disagreements in the results. In this paper, a diffusion equation model is applied to simulating reverberation chambers in order to obtain reverberant energy distributions and sound energy flows of the chamber under investigation in a more efficient way than wave-based simulation models. The computational efficiency of the diffusion equation lies in the fact that the meshing condition of the simulation domain is dictated by the room's mean-free path length (MFPL). Further investigation shows one-twelfth of MFPL as the meshing condition is considered practically sufficient for obtaining random incidence absorption coefficient of standardized sample sizes. With the computational efficiency given by a meshing of up to twelfth MFPL, an inversion calculation of random incident absorption coefficients of highly absorptive materials is possible with a mixed boundary condition [Jing and Xiang, JASA 123, pp. 145–153 (2008)]. Based on analysis of both chamber-based measurements and simulated data, ways to scrutinize measurement results of absorptive materials are investigated.

2:00

3pAA3. An iterative approach for estimating the sound speed and attenuation constant of air in a standing wave tube. Zhuang Mo (Ray W. Herrick Labs., Purdue Univ., Ray W. Herrick Labs., Purdue University, 177 S. Russell St., West Lafayette, IN 47907-2099, mo26@purdue.edu), Guochenhao Song, Hou Kang, and J. S. Bolton (Ray W. Herrick Labs., Purdue Univ., West Lafayette, IN)

In the present work, an iterative method based on the four-microphone transfer matrix approach was developed for evaluating the sound speed and attenuation constant of air within a standing wave tube. In particular, when the air inside the standing wave tube is treated as the material under test, i.e., as if it were a sample of porous material, the transfer matrix approach can be used to identify the air's acoustic properties. Note that the wavenumber within the tube is complex owing to the formation of a visco-thermal boundary layer on the inner circumference of the tube. Starting from an assumed prior knowledge of the air properties, an iterative method can be applied in the post-processing stage to estimate the complex wavenumber accurately. Experimental results presented here show that although the results are sensitive to ambient temperature, a formula previously proposed by Temkin matches closely with the measured sound speed and attenuation constant. Furthermore, it is shown that the Temkin prediction accurately represents the variation of sound speed with frequency, in contrast to the formula recommended in the ASTM E1050 standard, in which the sound speed is assumed to be independent of frequency.

2:15

3pAA4. Why noise reduction coefficients aren't enough. Richard L. Lenz (RealAcoustix LLC, 2361 B Ave., Ogden, UT 84401, RL@RealAcoustix.com)

Noise reduction coefficients have been around for decades. The use, and misuse, of the data supplied by these tests has been well-documented over that time. The desire by some to bring changes to standards like ASTM C423 and ISO 354 have been met with tremendous resistance by both manufacturers, laboratories and others over the years. This paper will discuss the use of these standards, what they tell us about sound and where they fall short in the assessment of the quality and assessment of sound absorption. Discussion of inappropriate use of standards in testing materials will also be presented. A discussion on the subject of timbre and how different materials may sound different while having similar NRC ratings will also be presented. The need for different test data and higher standards in the industry of acoustics will also be discussed. A presentation of the use of a free-field test method for assessing the quantity and quality of absorption will be presented along with test data from this method.

2:30

3pAA5. Spatial room impulse response measurement and post-processing for virtual reproduction of a stereo sound system. Zane T. Rusk (Dept. of Architectural Eng., The Penn State Univ., 104 Eng. Unit A, University Park, PA 16802, ztr4@psu.edu) and Michelle C. Vigeant (Graduate Program in Acoust., The Penn State Univ., University Park, PA)

Spatial room impulse responses (SRIRs) characterize the spatial, temporal, and spectral properties of the sound field received at a particular location in a room for a given source-receiver configuration. SRIRs can be captured using spherical microphone arrays and are useful both for evaluating the acoustic performance of a space and in applications where it is desired to auralize the sound field being characterized with varying signals being played from the source. In this work, SRIRs were measured in a living room (mid-frequency RT approximately 0.4 s) using an mh acoustics Eigenmike em32 as the receiver and both a studio monitor loudspeaker and a subwoofer as the sources. Studio monitor placements replicated a left-right-center stereo monitoring configuration, with the subwoofer on the floor below the center loudspeaker, while attempting to avoid source and receiver placement on estimated modal planes in the room. Details of the post-processing used to finalize the SRIRs for use as a virtual stereo sound system will be presented, including cross-over filtering, tuning, and correction of the overall frequency response to a curve that is favorable for in-room loudspeaker listening. Measurement system details, SRIR data visualizations, and RT estimates will also be discussed.

2:45

3pAA6. Raising the bar for jack-up floating floors—Understanding laboratory test results and proposed future test methodologies. Adam P. Wells (None, 530 7th Ave., Ste. 902, New York, NY 10018, adam.prescott.wells@icloud.com)

The jack-up concrete floating floor has been a staple in the field of architectural acoustical design for decades. Compatible with both spring and elastomeric isolators, the jack-up concrete floating floor system is often specified when the highest level of airborne and structure-borne vibration isolation separation is desired, and assurance of a decoupled floor is needed. Recent laboratory testing for jack-up floating floors under various configurations has yielded interesting and perhaps unexpected results. This paper will review the results from laboratory testing and dissect which factors appear to influence the airborne and/or the structure-borne isolation performance. A preview of future testing will be discussed, including new testing methodologies to validate existing conclusions and identify the influence of additional factors that may influence the airborne and/or structure-borne isolation performance.

Session 3pBAa

Biomedical Acoustics and Physical Acoustics: Bubble-Cell Interaction II

Eleanor P. Stride, Cochair
University of Oxford, Institute of Biomedical Engineering, Oxford, OX3 7DQ, United Kingdom

Klazina Kooiman, Cochair
Erasmus MC, Wytemaweg 80, Room Ee2302, Rotterdam, 3015 CN, Netherlands

Invited Paper

1:00

3pBAa1. Characterization of microbubble-enhanced molecular delivery to cells in acoustofluidic channels. Jonathan A. Kopechek (Bioengineering, Univ. of Louisville, 2301 S. Third St., Paul C.Lutz Hall, Rm. 419, Louisville, KY 40292-0001, jonathan.kopechek@louisville.edu)

Cell therapies are rapidly emerging as a promising approach for treatment of many diseases, such as cancer and cardiovascular diseases. Non-viral methods for molecular delivery are currently in development to increase safety and reduce varability during manufacturing of cell therapies. Ultrasound-mediated microbubble cavitation has been utilized as an effective approach for non-viral molecular delivery via sonoporation or other mechanisms. However, prior ultrasound studies have primarily utilized static sample chambers, which have limited throughput and are generally not practical for cell therapy manufacturing processes. To address these limitations, we are developing novel acoustofluidic platforms to enable intracellular delivery of biomolecules as cells continuously pass through an ultrasound field in a flow chamber. We have investigated a range of fundamental parameters that influence acoustofluidic-mediated molecular delivery to human cells, including microbubble properties, channel geometry, and ultrasound parameters. In addition, we investigated biological factors that influence acoustofluidic-mediated molecular delivery to cells, including cellular properties and extracellular metabolite levels. The results of these studies will be presented to provide new insights into the key parameters and mechanisms that affect the efficiency of molecular delivery to human cells in acoustofluidic channels. Further development of acoustofluidic platforms may enable improved processes for manufacturing of cell therapies.

Contributed Papers

1:30

3pBAa2. Use of static overpressure to assess the role of acoustic cavitation in ultrasound-mediated gene transfection. Chun Kiat Ng, Jason L. Raymond, Claudia Driscoll, Sophie Oldroyd, Wei E. Huang, Ian P. Thompson (Dept. of Eng. Sci., Univ. of Oxford, Oxford, United Kingdom), and Ronald A. Roy (Dept. of Eng. Sci., Univ. of Oxford, Dept. of Eng. Sci., 17 Parks Rd., Oxford OX1 3PJ, United Kingdom, ronald.roy@eng.ox.ac.uk)

Ultrasound-mediated DNA delivery (UDD) is a technique that utilizes ultrasound to aid the transfer of genetic materials into bacterial cells of interest (either in their planktonic or biofilm modes), resulting in new functionalities being displayed by the target microorganisms *in-situ*. The physical mechanism of UDD is poorly understood, which hampers its widespread adoption and scaling. To investigate the role of cavitation in UDD, we utilized a static pressure vessel designed to suppress cavitation while leaving other acoustic effects unaltered. *Pseudomonas putida UWC1*, a soil bacterium that is not naturally competent, was suspended in water and exposed to 40 kHz burst-mode ultrasound (400 cycle burst, 10 Hz repetition frequency, 60 s total) at an ambient pressure of either 1 bar or 9.5 bars. Noise diagnostics yielded inertial cavitation levels and transformed cells were quantified using a doubly selective reporter plasmid DNA. Samples exposed at 9.5 bars overpressure exhibited a drastic reduction in both cavitation activity and the number of transformed cells compared to those exposed at 1 bar. (Cell viability was not affected by pressure change.) The fact that transformation rates are reduced significantly when cavitation is suppressed suggests that acoustic cavitation is the major contributing factor in UDD.

1:45

3pBAa3. Ultrasound-assisted membrane permeabilization of endothelial cells under flow conditions. Elahe Memari (Phys., Concordia Univ., Montreal, QC, Canada), Fiona Hui (Biology, Concordia Univ., Montreal, QC, Canada), and Brandon Helfield (Phys., Concordia Univ., 7141 Sherbrooke St. West, L-SP 365.04, Montreal, QC H4B 1R6, Canada, brandon.helfield@concordia.ca)

Ultrasound-stimulated microbubbles have been shown a feasible approach for localized therapeutic delivery. As applications of this technique span many anatomical sites, so too do the local fluid dynamics experienced by the circulating microbubbles and the adjacent endothelial cells. Our objective was to assess the relative effectiveness of endothelial cell sonoporation as a function of flow conditions. Human umbilical vein (HUVECs) or human brain endothelial cells (HBECs) were cultured as a monolayer in flow chamber slides connected to a fluidic system and placed upon an acoustically-coupled microscope. A suspension of diluted lipid-encapsulated microbubbles and propidium iodide (PI), used as a sonoporation marker, was constantly perfused over the monolayer at either 5 or 30 ml/min. Cells were treated with 1 MHz ultrasound (PRI= 1 ms, 20 cycles, duration = 2 s), and the video-microscopy data were quantified offline to assess the number of PI-positive cells. Our results demonstrate a marked increase in sonoporation efficiency at 30 ml/min as compared to 5 ml/min in both endothelial cell lines under identical acoustic conditions (9.7-fold increase and 2.3-fold increase for HUVECs and HBECs respectively, $p < 0.001$). Our results suggest the local fluid flow environment plays

3p WED. PM

a role in US-mediated endothelial perforation efficiency and can modulate treatment strategies.

2:00

3pBAa4. Microbubble-endothelial cell interactions in 3D: Internalization of microbubbles and pore or tunnel formation for drug delivery. Ines Beekers, Simone A. G. Langeveld (Biomedical Eng., Erasmus MC, Rotterdam, Netherlands), Bram Meijlink (Biomedical Eng., Erasmus MC, Rotterdam, Zuid-Holland, Netherlands), Antonius F. van der Steen, Nico de Jong (Biomedical Eng., Erasmus MC, Rotterdam, Netherlands), Martin D. Verweij (Imaging Phys., Delft Univ. of Technol., Delft, Netherlands), and Klazina Kooiman (Biomedical Eng., Erasmus MC, Wytemaweg 80, Rm. Ee2302, Rotterdam 3015 CN, Netherlands, k.kooiman@erasmusmc.nl)

Ultrasound-activated microbubbles can locally enhance vascular drug delivery, but fully understanding the mechanism requires further investigation. The aims of this *in vitro* study were to (1) assess the initial single microbubble-endothelial cell (n = 301) 3D morphology, (2) determine whether the ligand type on the targeted microbubble affected this morphology, and (3) investigate the morphology's influence on microbubble oscillation and drug delivery outcome, all using high-resolution 3D confocal microscopy in combination with ultra-high-speed imaging ($\sim$17 Mfps). Non-targeted and IgG1-κ control microbubbles were not internalized by endothelial cells, while targeted microbubbles were internalized. The internalization depth was ligand-dependent, since microbubbles having $\alpha_v\beta_3$ as target were significantly more internalized than those with CD31. Although the internalized microbubbles (n = 246) had a damped oscillation upon ultrasound insonification (2 MHz, 200 kPa, 10 cycles), their ability to sonoporate cells (i.e., PI uptake) was increased. Upon sonoporation (n = 230), either pores or tunnels (i.e. transcellular perforation) were formed in the cell membrane for respectively intracellular or transcellular delivery. Fewer transcellular perforations and smaller pore areas were observed for internalized microbubbles. In conclusion, receptor-mediated microbubble internalization, its effect on microbubble oscillation, and resulting membrane perforation were revealed using a state-of-the-art imaging system, thereby providing novel insights.

2:15

3pBAa5. The OrganoPlate® as vessel-on-a-chip model to investigate increased microbubble-mediated vascular permeability. Bram Meijlink (Biomedical Eng., Erasmus MC, Wytemaweg 80Room Ee2322, Rotterdam, Zuid-Holland 3015 CN, Netherlands, k.meijlink@erasmusmc.nl), Ines Beekers, Simone A. G. Langeveld (Biomedical Eng., Erasmus MC, Rotterdam, Netherlands), Kristina Bishard (Mimetas, Leiden, Netherlands), Antonius F. van der Steen, Nico de Jong (Biomedical Eng., Erasmus MC, Rotterdam, Netherlands), Sebastiaan J. Trietsch (Mimetas, Leiden, Netherlands), and Klazina Kooiman (Biomedical Eng., Erasmus MC, Rotterdam, Netherlands)

The vessel wall is an important barrier modulating drug delivery to the underlying diseased tissue. Oscillating microbubbles can be used to locally enhance vascular permeability and sonoporate cells. As the mechanism is not fully understood, our aim was to grow 3D human vessels-on-a-chip in the OrganoPlate® 40 and use this model to investigate the effect of $\alpha_v\beta_3$-targeted microbubble and different ultrasound pressures (2 MHz, 100–850 kPa peak negative pressure) and cycle lengths (10$\times$10 or 10$\times$1000 cycles) on vascular permeability and sonoporation. The vascular permeability of 122 microvessels in 14 different conditions was quantified by microscopy imaging using the leakage pattern of a 150 kDa FITC-dextran dye. Furthermore, sonoporation was assessed using propidium iodide (PI). Upon microbubble and ultrasound treatment, an increase in vascular permeability was observed. Higher pressures and longer cycle length treatment showed a significantly higher vascular permeability and significant increase in PI uptake compared to all control groups (sham, ultrasound only, microbubble only), suggesting a simultaneous increase in vascular permeability and sonoporation correlating with higher pressure and longer cycle insonifications. In conclusion, the vessel-on-chip model is a suitable model to investigate how insonification with different ultrasound settings affects the microbubble-mediated vascular permeability increase and sonoporation.

2:30

3pBAa6. Correlating high-speed optical imaging and passive acoustic mapping of cavitation dynamics. Qiang Wu, Michael Gray, Cameron Smith, Luca Bau (Inst. of Biomedical Eng., Univ. of Oxford, Oxford, United Kingdom), Constantin Coussios (Inst. of Biomedical Eng., Univ. of Oxford, Headington, Oxford, Oxfordshire, United Kingdom), and Eleanor P. Stride (Inst. of Biomedical Eng., Univ. of Oxford, Inst. of Biomedical Eng., Oxford OX3 7DQ, United Kingdom, eleanor.stride@eng.ox.ac.uk)

The biological effects of acoustic cavitation are mediated by a range of phenomena associated with different types of bubble activity, e.g., microstreaming, micro-jetting, and shockwave generation. The acoustic emissions generated by cavitation are also correlated to bubble dynamics. Hence, monitoring these emissions during ultrasound therapy is desirable, to maximise treatment safety and efficacy. The precise relationship between the spectral content of acoustic emissions and bubble dynamics is, however, less well understood. The aim of this study was to use simultaneous ultra-high-speed optical imaging (1–10 MHz) and passive acoustic mapping to characterise the behaviour of individual and clusters of microbubbles over a range of ultrasound exposures. As expected from the literature and numerical modelling, both the number of discrete harmonic components and amplitude of broadband content in the emissions increased with increasing amplitude of bubble oscillation. This suggests that passive acoustic mapping provides a useful indicator of spherical bubble behaviour. Frequently, however, complex bubble behaviour, such as fragmentation and coalescence was observed, which could produce substantially different effects in tissue compared with spherical bubble collapse. Future work will focus on determining whether these differences can be adequately captured in defining a future cavitation "dose" based on acoustic emissions for different applications.

2:45–3:15
Panel Discussion

Session 3pBAb

Biomedical Acoustics: Biomedical Acoustics Best Student Paper Poster Session

Kevin J. Haworth, Cochair
Internal Medicine, University of Cincinnati, 231 Albert Sabin Way, CVC 3939, Cincinnati, OH 45267-0586

Kenneth B. Bader, Cochair
University of Chicago, 5835 South Cottage Grove Ave., Department of Radiology, MC 2026, Q301B, Chicago, IL 60637

The ASA Technical Committee on Biomedical Acoustics offers a Best Student Paper Award to eligible students who are presenting at the meeting. Each student must defend a poster of her or his work during the student poster session. This defense will be evaluated by a group of judges from the Technical Committee on Biomedical Acoustics. Additionally, each student will give an oral presentation in a regular/special session. Up to three awards will be presented to the students with USD $500 for first prize, USD $300 for second prize, and USD $200 for third prize. The award winners will be announced at the meeting of the Biomedical Acoustics Technical Committee.

Below is a list of students competing, with abstract numbers titles. Full abstracts can be found in the oral sessions associated with the abstract numbers. All entries will be on display, and all authors will be at their posters from 1:00 p.m. to 3:00 p.m.

1aBA1. Spatiotemporal decomposition methods for nanobubble contrast enhanced ultrasound
Student author: Dana Wegierak

1aBA2. Sensitivity of the subharmonic responses from contrast microbubbles to ambient pressure
Student author: Roozbeh Hassanzadeh Azami

1aBA7. The effect of stiffness and impurities on bubble nucleation in polyacrylamide hydrogels
Student author: Ferdousi S. Rawnaque

1aBA8. Histotripsy bubble dynamics in tendon and anisotropic gel phantoms
Student author: Jacob C. Elliott

1aBA9. The effect of surface tension on the color Doppler ultrasound twinkling artifact
Student author: Eric Rokni

1aBA11. Passive cavitation detection with a needle hydrophone array
Student author: Zheng Jiang

2aBAa2. Investigating the link between intensity of lung ultrasound vertical artifacts and penetration depth of ultrasound waves, in silico study
Student author: Federico Mento

2aBAa6. Localizing pulmonary nodules for surgical resection using ultrasound multiple scattering
Student author: Roshan Roshankhah

2aBAa7. Quantifying severity of lung fibrosis in rodents using random matrix theory
Student author: Azadeh D. Cole

2aBAb8. Optimization of cavitation-mediated mRNA delivery to cancer and non-cancer cells in vitro and in vivo
Student author: Alexander Martin

2pBAa3. Effect of microbubble and acoustic radiation force parameters on $\alpha v \beta 3$-microbubble targeting
Student author: Jair I. Castillo

2pBAa4. Inactivation of gram positive or gram negative microbes using different histotripsy regimes
Student author: Pratik Ambekar

2pBAa5. Ultrasound contrast agents: from buckling dynamics to swimming
Student author: Georges Chabouh

2pBAa7. A novel ultrasound-assisted laser technique to remove atherosclerotic plaques
Student author: Rohit Singh

2pBAa9. Application of Koopman operator theory to the control of nonlinear bubble dynamics
Student author: Andrew J. Gibson

2pBAa10. Analysis of cavitation induced stresses on blood vessel wall during photo-mediated ultrasound therapy using finite-element based numerical models
Student author: Rohit Singh

2pBAa12. Combining ultrasound and endovascular laser for thrombolysis
Student author: Rohit Singh

2pBAb5. On the impact of pixel resolution on automated scoring of lung ultrasound images from Coronavirus disease 2019 patients
Student author: Umair Khan

2pBAb6. Neuro-symbolic Interpretable AI for automatic COVID-19 patient-stratification based on standardised lung ultrasound data
Student author: Leonardo L. Custode

3aBA3. Ultrasound-targeted microbubble cavitation increases paracellular gaps in an in vitro blood brain barrier model
Student author: Grace E. Conway

3aBA4. Microbubble dynamics in brain microvessels at 1 MHz and 330 kHz driving frequencies
Student author: James H. Bezer

3aBA10. Ultra-high speed quantification of cell strain during cell-microbubble interactions
Student author: Oliver Pattinson

3aBA11. Exploration of ultrasound-mediated microbubble-cell membrane interactions using novel protein-loaded microbubbles and their role in immunomodulation
Student author: Veerle A. Brans

3pBAa5. The OrganoPlate® as vessel-on-a-chip model to investigate increased microbubble-mediated vascular permeability
Student author: Bram Meijlink

4aBA2. Viscoelasticity inversion for arterial shear wave elastography
Student author: Tuhin Roy

4aBA3. Acoustic and viscoelastic characterization of hydrogel scaffolds to optimize preparation parameters for tissue engineering applications
Student author: Megan S. Anderson

4aBA5. Full-waveform shear wave elastography for imaging tumors
Student author: Abdelrahman M. Elmeliegy

4aBA7. The evolution of microwave-induced thermoacoustic signal characteristics generated during pulsed microwave ablation
Student author: Audrey L. Evans

4aBA9. Comparison between focused ultrasound and dry needling treatments in a murine Achilles tendinopathy model
Student author: Molly Smallcomb

4aBA10. Investigation of a noninvasive method for monitoring intracranial pressure using sheep skulls
Student author: Nicholas Cameron

4aBA15. Side lobe reduction using null subtraction imaging
Student author: Zhengchang Kou

4pBA10. Design of an electronic radiological clip for improved breast cancer imaging with ultrasound
Student author: Jenna Cario

4pBA11. Implementation of real-time high-speed ultrasound communications through tissue
Student author: Zhengchang Kou

Session 3pCA

Computational Acoustics, Signal Processing in Acoustics, and Structural Acoustics and Vibration: Application of Model Reduction Across Acoustics

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

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

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

Chair's Introduction—1:00

Invited Papers

1:05

3pCA1. A numerical approach to estimate bulk wave speeds and dispersion. Matthew S. Allen (Mech. Eng., Brigham Young Univ., 350B EB, Brigham Young University, Provo, UT 84602-4119, matt.allen@byu.edu), Jonathan Blank, Lesley Arrant, Josh Roth, and Darryl Thelen (Mech. Eng., Univ. of Wisconsin-Madison, Madison, WI)

In many applications, one is interested in estimating the wave speed in materials or structures for which the wave equation cannot be solved analytically. This work is concerned with studying wave propagation in tendons and ligaments, where material anisotropy, nonlinearity, prestress, and non-ideal boundary conditions make an analytical solution impractical. In these cases, the finite element method can be used to simulate the structure in time and then extract wave motion, but this is computationally expensive, both in the simulation and for post processing. This work exploits the duality between modes of vibration and traveling waves to show how one can perform a semi-analytical dispersion analysis. The proposed method is then applied to models of tendons and ligaments of varying complexity and is found to provide significant insights into the nature of wave propagation and the expected dispersion behavior.

1:30

3pCA2. Real-time acoustic modeling in the 2021 New England Shelf Break Acoustics experiment and beyond. Brendan J. DeCourcy (Appl. Ocean Phys. & Eng., Woods Hole Oceanographic Inst., 86 Water St., Falmouth, MA 02543, bdecourcy@whoi.edu), Ying-Tsong Lin, and Weifeng G. Zhang (Appl. Ocean Phys. & Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

Real-time acoustic modeling in a shipboard context emphasizes the element of computation time in pursuit of *in situ* comparison between observed ocean data and model output. The importance of computation time highlights two competing factors in efforts to produce real-time acoustic predictions: the quantity of models to run and model accuracy. During the 2021 New England Shelf Break Acoustics (NESBA) experiment, a four-dimensional (4D) model of acoustic signals in the experiment network comprised of multiple stationary sound sources and receivers was produced using shipboard high-performance computing (HPC) resources. This effort involved hundreds of acoustic model runs across four separate source and receiver pairings and allowed for *in-situ* model adjustments, ultimately producing favorable comparisons between data and models in a matter of days. This presentation will examine the HPC approaches taken during the NESBA science cruise and address the planned improvements to the system for future scientific field experiments. [This work is supported by the Office of Naval Research.]

1:55

3pCA3. Efficient band-structure calculations of non-classically damped phononic materials by Bloch mode synthesis in state space. Abdelaziz Aladwani (Manufacturing Eng. Technol. Dept., College of Technolog. Studies, Shuwaikh, Kuwait), Mostafa Nouh (Mech. and Aerosp. Eng., Univ. at Buffalo (SUNY), Buffalo, NY), and Mahmoud I. Hussein (Aerosp. Eng. Sci., Univ. of Colorado Boulder, 3775 Discovery Dr., AERO 354, 429 UCB, Boulder, CO 80303, mih@colorado.edu)

Bloch mode synthesis (BMS) techniques enable efficient band-structure calculations of periodic media by forming reduced-order models of the unit cell. Rooted in the framework of the Craig-Bampton component mode synthesis methodology, these techniques decompose the unit cell into interior and boundary degrees-of-freedom that are nominally described, respectively, by sets of normal modes and constraint modes. In this paper, we generalize the BMS approach by state-space transformation to extend its applicability to

generally damped periodic materials that violate the Caughey-O'Kelly condition for classical damping. In non-classically damped periodic models, the fixed-interface eigenvalue problem may, in general, produce a mixture of underdamped and overdamped modes. We examine two mode-selection schemes for the reduced order model and demonstrate the underlying accuracy-efficiency trade-offs when qualitatively distinct mixtures of underdamped and overdamped modes are incorporated. The proposed approach provides a highly effective computational tool for analysis of large models of phononic crystals and acoustic/elastic metamaterials with complex damping properties. This investigation does not only extend the applicability of BMS techniques to the most generally damped models of periodic media, it also advances our understanding of the nature of damping modes and the non-trivial manner by which they contribute to the wave propagation properties.

2:20

3pCA4. Asymptotic modal analysis application to a cavity-plate system to predict sound transmission loss. Shung H. Sung (SHS Consulting, LLC, 4178 Drexel Dr., Troy, MI 48098, ssung1972@gmail.com) and Donald J. Nefske (DJN Consulting, LLC, Troy, MA)

The asymptotic modal analysis (AMA) method has previously been developed to predict the high-frequency dynamic response of structural and acoustical systems [Sung *et al.*, *Asymptotic Modal Analysis of Structural and Acoustical Systems* (Morgan & Claypool Pub., 2021)]. In the AMA method, the dynamic response is obtained as the asymptotic limit of the classical modal analysis (CMA) solution as the averaged response in large frequency bands. In previous work, the AMA method has been applied to uncoupled individual structural and acoustical systems. In the asymptotic limit, the AMA results for individual structural and acoustical systems have been shown to approach the corresponding statistical energy analysis (SEA) results. The AMA method has also previously been applied to a coupled cavity-plate system to predict the high frequency response for piston-like excitation of the plate. In this paper, the AMA method is applied to a cavity–plate system subjected to random excitation of the plate to predict the sound transmission loss into the cavity. The AMA results in the asymptotic limit for the cavity-plate system are shown to approach the corresponding CMA and SEA results. In addition, the AMA method is able to predict the spatial distribution of the plate and cavity responses.

Contributed Paper

2:45

3pCA5. Predictions for the perturbed root mean square displacement of a vibrating structure using modal parameters. Allison Kaminski (Mech. Eng., Boston Univ., 110 Cummington Mall, Boston, MA 02215, allison9@bu.edu) and James McDaniel (Mech. Eng., Boston Univ., Boston, MA)

A harmonically vibrating structure is considered for this study and divided into segments. The objective of the proposed research is to use knowledge from the nominal structure to make predictions as to how the root mean squared (RMS) displacement of the structure will change when either the mass or stiffness of a segment is scaled. Typically, to calculate the value for the modified RMS displacement, a linear solve is required to determine the new displacement vector. Additionally, the Neumann series may be used. However, this approach requires computations of matrix-vector products, which may become expensive for larger Degrees of Freedom (DOF) systems. Here, a method is proposed to predict the RMS displacement for the modified system from scalar values of the nominal system. Since only scalar values from the nominal system are required, predictions for the perturbed system can be made cheaply. The proposed method is based on the modal displacement equation and the assumption that the system is forced near a natural frequency. This assumption is used in Rayleigh's quotient to approximate the new natural frequency as a function of the approximated modal stiffness and mass. The limitations and accuracy will be explored. [Work supported by ONR Grant N00014-19-1-2100.]

Session 3pEA

Engineering Acoustics, Structural Acoustics, and Vibration and Physical Acoustics: Smart Metamaterials and Metastructures II

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

Bogdan-Ioan Popa, Cochair
Univ. of Michigan, 2350 Hayward St., Ann Arbor, MI 48109

Chair's Introduction—1:00

Contributed Papers

1:05

3pEA1. Multi-material stimuli-responsive hydrogels with optically induced actuation. Haley Tholen (The Penn State Univ., 336 Reber Bldg., University Park, PA 16802, hmt5321@psu.edu) and Ryan L. Harne (The Penn State Univ., State College, PA)

A vision for soft, autonomous materials entails synthesis of multiple senses in multifunctional materials where material response requires sensitivity to external stimuli. Stimuli-responsive hydrogels are of particular interest for optically induced mechanical response due to the ability to transform external stimuli into large, reversible shape change. Specifically, temperature-responsive hydrogels are broadly used and can be designed to achieve deformation through the photothermal effect as a result of surface plasmonic resonance of gold nanoparticles. Here, a multi-material stimuli-responsive hydrogel network with embedded gold nanoparticles is demonstrated in a unit cell pattern with anisotropic swelling behavior in response to visible light. Reversible, anisotropic swelling leads to bending motion that contributes to the development of soft, autonomous materials.

1:20

3pEA2. Enhancing low frequency sound radiation of electrodynamic loudspeakers with acoustic metamaterials. Xiuyuan Peng (Elec. and Comput. Eng., Duke Univ., 101 Sci. Dr., Durham, NC 27705, xp21@duke.edu), Junfei Li, and Steven Cummer (Elec. and Comput. Eng., Duke Univ., Durham, NC)

Since the radiation resistance of a vibrating membrane in the air is proportional to frequency in the subwavelength range, low frequency (<100 Hz) sound radiating devices generally require a large vibrating diaphragm and a bulky enclosure to be efficient and loud enough for practical use. The tapped horn topology, where the front radiation of an electrodynamic driver adds up with the back radiation after propagating through the internal volume of a speaker box, has been widely used in custom audio to provide low-frequency enhancement to the bass unit. Recently, people have demonstrated the application of acoustic metamaterials based on multiple Helmholtz resonators (HRs) in sound absorption, sound insulation, directivity control, impedance matching, etc. In this presentation, we show that by branching the sound path of a tapped horn speaker box with multiple Helmholtz resonators and fine-tuning the geometry of each individual element, we can produce high and relatively constant sound pressure levels at subwavelength frequencies, making our methodology a good candidate for compact subwoofer design. Genetic algorithm (GA) is used to bring forth optimal performance within the geometric framework. Both transfer matrix method and numerical simulation are employed to derive the sound output of a particular subwoofer.

1:35

3pEA3. Quiet power: Exploring the feasibility of a noise-mitigating, thermoacoustic energy harvester. Samarjith Biswas (Oklahoma State Univ., 201 General Academic Bldg., Stillwater, OK 74078, samarjith.biswas@okstate.edu) and James M. Manimala (Oklahoma State Univ., Stillwater, OK)

The thermoacoustic effect provides a means to convert acoustic energy to heat and vice versa without the need for moving parts. This enables the realization of mechanically robust, noise mitigating energy harvesters, although there are limitations to the power-to-volume ratio achievable. The mechanical, thermal, and geometric properties of the porous stack that forms a set of acoustic waveguides in thermoacoustic devices are key to its performance. In this feasibility study, first, various 4-in. diameter ceramic and polymeric stack designs are evaluated using a custom-built thermoacoustic test rig. Influence of stack parameters such as material, length, location, porosity, and pore geometry are correlated to simulations using DeltaEC, a software tool based on Rott's linear approximation. An acousto-thermo-electric transduction scheme is employed to harvest useable electrical power using the best performing stack. Steady-state peak voltage generated was 33.5 mV for a temperature difference of 34 °C between the hot and cold sides of the stack at an acoustic excitation frequency of 117.5 Hz. Further investigations are underway to establish structure-performance relationships by extracting scaling laws for power-to-volume ratio and frequency-thermal gradient dependencies.

1:50

3pEA4. Design of multi-material sound transmission blocker using additive manufacturing. Trigun Maroo (Systems Eng., Univ. of Arkansas at Little Rock, Little Rock, AR) and Andrew Wright (Systems Eng., Univ. of Arkansas at Little Rock, 2801 South University Ave., Little Rock, AR 72204, abwright@ualr.edu)

Additive manufacturing is most commonly done through 3D printing, which can print in multiple materials. This enables the design and construction of new structures that can block sound transmission, such as a periodic array of TPU plastic cylinders embedded in a matrix of PLA plastic. The difference in wave speed between the plastics can cause reflections and interference inside the material. However, printing in multimaterials provides distinct challenges for the current 3D printing technology. When printing with multiple nozzles, the inactive nozzle can clog, and getting the two materials to fuse can yield warping and separation. We addressed these problems to fabricate a sample of TPU cylinders (1.8 cm diameter, 8.6 cm height) embedded in PLA (20 cm × 10 cm × 2 cm) and measured the change

3p WED. PM

in STL relative to a block of PLA. We discovered an improved attenuation of about 5.5 dB at 6300 Hz.

2:05

3pEA5. Application of acoustic metasurfaces for reduction of broadband noise generated by tire-pavement interaction. Hyeonu Heo (Phys., Univ. of North Texas, 1155 Union Circle # 311427, Denton, TX 76203, hyeonu.heo@unt.edu), Jaehyung Ju (UM-SJTU Joint Inst., Shanghai Jiao Tong Univ., Shanghai, China), Arup Neogi, and Arkadii Krokhin (Phys., Univ. of North Texas, Denton, TX)

Noise pollution by traffic is the most widespread environmental problems that cause sleep disturbance, hearing damage, even cardiovascular disease. The primary noise sources of the vehicle are the engine, exhaust, aeroacoustics, and tire-pavement interaction. Among them, a dominant noise within automobiles occurs from the tire-pavement interaction. Most noise suppression efforts aim to use sound absorbers or cavity resonators to narrow the bandwidth of acoustic frequencies using acoustic foams or Helmholtz resonators. However, the effectiveness of existing methods of noise reduction is limited by the design constraints and material itself. In this study, we propose artificially designed reflecting acoustic metasurfaces to control and isolate sound generated by a moving car. The proposed design can significantly reduce the noise arising from tire-pavement interaction over a broadband of acoustic frequencies under 2 kHz and over a wide range of vehicle speeds. A set of experiments with lab-scale and field tests have demonstrated that the lightweight metamaterial displaced inside the tires gives 2—5 dB stronger reduction of 200—300 Hz noise inside the car cabin than currently used acoustic foam. The proposed approach can be extended to other objects generating low-frequency mechanical noise. [Work supported by the National Science Foundation under EFRI Grant No. 1741677.]

2:20

3pEA6. Connections between unit cells in locally resonant metamaterials and their impact on the effectiveness of noise mitigation by its base structure. Klara Juros (AGH Univ. of Sci. and Technol., Mickiewicza 30, Cracow 30-059, Poland, juros@agh.edu.pl) and Aleksander Kras (Silencions Sp. z o.o., Wroclaw, Poland)

Locally resonant metamaterials (LRS) for sound and vibration mitigation have been a widely investigated topic in the field of acoustic in recent years. Many articles and projects describe fundamental problems of LRS design, numerical simulation, and laboratory measurements. However, there is a noticeable knowledge gap between laboratory prototypes and the mass production (development for a wide-scale implementation) of metamaterials structures. In general, the LRS prototypes require specific boundary conditions, for example, equal spacing between the individual resonators and separate attachment of each resonator to its base. These conditions allow for many simplifications in calculations. At the same time, these boundary conditions create a problem when it comes to the mass production of the LRS structures. Creating a connection between adjacent resonators would allow easier postprocessing of structures ex. attaching the structure to the base and keeping specified distances between resonators. This project investigates the possibilities of creating connections between adjacent unit cells and their influence on the simulation and measurement results. The effect on the simulated band gaps and sound transmission loss is investigated at the stage of numerical simulations. The prototypes of structures are 3D printed and tested. The laboratory measurement results are compared with simulations and further discussed.

2:35

3pEA7. Phononic crystal as a homogeneous viscous metamaterial. Arkadii Krokhin (Phys., Univ. of North Texas, University of North Texas1155 Union Circle #311427, Denton, TX, arkady@unt.edu), Martin Ibarias (Instituto de Fisica, Universidad Autonoma de Puebla, Puebla, Puebla, Mexico), Yurii Zubov (Phys., Univ. of North Texas, Denton, TX), and Jesus Arriaga (Instituto de Fisica, Universidad Autonoma de Puebla, Puebla, Puebla, Mexico)

At low frequencies, a phononic crystal behaves like a homogeneous medium. Different variants of homogenization technique have been recently proposed to calculate the effective elastic parameters of periodic medium. Here, we develop a homogenization theory for a phononic crystal of solid rods in a viscous fluid. Using the plane wave expansion method, we derive analytical formula for the decay coefficient of low frequency sound propagating in a 2D Bravais lattice with arbitrary unit cell. It is shown that due to the formation of a viscous boundary layer around each cylinder, the losses are enhanced by two to four orders of magnitude as compared to the losses in the free fluid. This enhancement depends on the filling fraction of solid rod and it becomes very strong for almost touching scatterers when sound propagates through narrow slits between the neighbouring rods. Also, the decay coefficient in a phononic crystal scales with frequency as ω^2, unlike scaling $\sqrt{\omega}$ known for free viscous fluid. The enhanced viscous losses are associated with high effective viscosity of phononic crystal. Like other effective parameters the effective viscosity exhibits strong anisotropy, if the unit cell is asymmetric. The proposed theory can be used for evaluation of efficiency of acoustic devices based on phononic crystals. [This study is supported by NSF Grant No. 1741677.]

Session 3pID

Interdisciplinary: Hot Topics in Acoustics

Kevin M. Lee, Chair
Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758-4423

Chair's Introduction—2:40

Invited Papers

2:45

3pID1. Ocean acoustics in the rapidly changing Arctic ocean. Matthew A. Dzieciuch (SIO/UCSD, San Diego, CA, mad@ucsd.edu)

The Arctic ocean is undergoing rapid climate change, in fact it is the most rapidly warming area of our planet. Underwater sound is a uniquely effective tool for Arctic monitoring, because it can be used under the ice where satellites are blind. The most striking change has been the reduction of summer ice cover, but recent acoustic experiments have started to gather evidence for other changes as well, such as the reduction of ice scattering from the loss of multiyear ice and the formation of subsurface ducted propagation in the Beaufort Sea. The ambient soundscape is also of great interest as human activity increases in the region and as biological activity responds. Due the strategic importance of the Arctic during the cold war, there is a history of experimental work in the area but this short talk will try to show a few recent examples.

3:10

3pID2. New perspectives for active acoustic metamaterials. Bogdan-Ioan Popa (Univ. of Michigan, 2350 Hayward St., Ann Arbor, MI 48109, bipopa@umich.edu)

It is generally accepted that active acoustic metamaterials could, in principle, control impinging sound in ways not possible with passive media. One way to realize this potential is to employ structures composed of sensor-driver unit cells. In this approach, sensors are detecting the impinging wave and actuators are driven coherently with the sensed fields in order to generate almost arbitrary and programmable scattering profiles for the cells. This presentation will briefly review some of the rich and unusual dynamics of metamaterials based on these cells and, most importantly, will highlight the untapped potential of this architecture. In particular, the sensor-driver cells are not limited to generating a response in the same physical field as the excitation. Here, I will focus on metasurfaces that respond with light to acoustic waves. One possible application of these acousto-optical metamaterials will be discussed, namely, the realization of new acoustic imaging devices with potentially much larger resolutions than traditional imaging devices. [Work supported by the National Science Foundation under Grant No. CMMI-1942901.]

3:35

3pID3. Lung ultrasound: State of the art and future directions. Libertario Demi (Dept. of Information Eng. and Comput. Sci., Univ. of Trento, Via sommarive 9, Trento, Italia 38123, Italy, libertario.demi@unitn.it)

Following the COVID-19 pandemic, lung ultrasound (LUS) received growing attention as the need for a safe, point-of care, cost-effective, and widely available imaging solution was crucial. However, when performed with clinical scanners, LUS remains limited to the subjective and qualitative interpretation of artifacts and imaging patterns. This negatively affects accuracy and reproducibility. During the COVID-19 pandemic, we have developed a standardised LUS acquisition protocol and scoring system, aiming at minimizing the impact of several confounding factors. This approach has proven prognostic value and is capable of stratifying patients based on their risk of worsening. Moreover, the availability of an extensive and standardised dataset (772,780 LUS frames) allowed us to develop AI approaches capable of analysing LUS data automatically. Results from a multicentric study conducted on 220 patients will be presented. Next, beyond what can be achieved with clinical scanners, we are also pioneering the development of novel ultrasound solutions dedicated to the lung. To provide with a forward-looking view on the future of LUS, the latest results from numerical and *in vitro-* studies focused on unveiling the physical phenomena at stake will be discussed in detail, together with the latest results from the first clinical studies on quantitative lung ultrasound spectroscopy.

Session 3pMU

Musical Acoustics: Evolution and Maturation of Musical Acoustics

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

James P. Cottingham, Cochair
Physics, Coe College, 1220 First Avenue NE, Cedar Rapids, IA 52402

Chair's Introduction—1:00

Invited Papers

1:05

3pMU1. Why did wind instruments stop evolving? Jonas Braasch (School of Architecture, Rensselaer Polytechnic Inst., School of Architecture, 110 8th St., Troy, NY 12180, braasj@rpi.edu)

Adolphe Sax invented the last widely spread orchestral wind instrument in 1846, and the saxophone has hardly changed since its inception. Also, outside of classical music, wind instruments have not evolved with very few exceptions in popular music, like the melodica. The latter was popular in the mid 20th century, because it was one of the cheapest instruments with a piano-style keyboard before mass-produced electronic keyboards. Since its inception, jazz drew from traditional orchestral wind instruments that were invented long before jazz came up. They were easily available as they were widely spread in military ensembles. This presentation looks into the factors that stalled the evolution of mainstream wind instruments during the 19th century, such as instrument affordability, practice habits, conservatory education practices, standardization, and cultural identification. While individual instrument makers and musicians continue to develop fundamentally new wind instruments, they no longer exceed experimental status. Instead, widespread innovations now focus on electronic and digital musical instruments.

1:25

3pMU2. The evolution and maturation of the electric guitar as a system. M. Torben Pastore (College of Health Solutions, Arizona State Univ., PO Box 870102, ASU, Tempe, AZ 852870102, m.torben.pastore@gmail.com) and Nikhil Deshpande (Eventide, Inc., Austin, TX)

The electric guitar came to its initial prominence in the 1940s when its volume allowed jazz guitarists like Charlie Christian to step out in front of the rhythm section like other soloists, competing with the brass, wind, and piano. Further changes in the playing and design of the instrument came with exploitation of the interaction of the electric guitar, tube amplifiers, and analog effects as a larger system. Especially in the rock and funk idioms, the playing and construction of the overall instrument evolved and expanded rapidly with the mainstream embrace of digital technology. This talk will consider the evolution of the electric guitar as a system up through today and consider why it seems the instrument has fully evolved and is unlikely to experience further seismic shifts, especially as the overall thrust of music has shifted to digitally-manipulated sound that is often entirely independent of the physical playing of any instrument.

1:45

3pMU3. Life cycles of free reed instruments: The accordion-concertina family and the reed organ-harmonium family as case studies. James P. Cottingham (Phys., Coe College, 1220 First Ave. NE, Cedar Rapids, IA 52402, jcotting@coe.edu)

In 1800, the free reed instruments of European origin did not exist. Beginning in the 1820s, there was very rapid invention and innovation in the development and production of instruments like the reed organ and accordion so that within a few decades these instruments had reached something approaching their final form. By 1900 instruments with hand-driven bellows (accordions and concertinas) and instruments with foot-driven air supplies (reed organs and harmoniums) of a wide variety of types were being manufactured and played worldwide in large numbers. From around 1900 through the 1920s, there was a sharp drop in the number of new reed organs and harmoniums manufactured and sold. This number decreased to near zero by mid-century and has remained there. Meanwhile, the manufacture of the "squeeze box" instruments has remained robust to the present day. Factors involved in the discrepancy between these two families of instruments of almost identical acoustical design will be discussed. These include changes in musical styles and taste, socio-economic factors, and competition from other musical instruments.

2:05

3pMU4. Drums speak too: An examination of modern percussion instruments in the jazz idiom. E. K. Ellington Scott (School of Architecture, Rensselaer Polytechnic Inst., Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, scotte3@rpi.edu)

Jazz has been a leading force in the evolution of music in the United States. The complexity of rhythmic and harmonic languages has grown from the birth of jazz improvisation. Modern percussion instruments played a fundamental role in the development of rhythmic language. Improvised figures between the soloist and sectionals have cultivated a vocabulary still utilized by modern drummers and percussionists today. Moreover, evolving styles matured into a more sophisticated language, demanding a more refined instrument. This presentation explores the development of modern percussion instruments within jazz and improvised music, focusing on the historical and cultural aspects impacting the evolution of percussion instruments. Investigations of stylistic approaches and influences on musical styles will also be examined.

2:25

3pMU5. A look back at the development of the steelpan and related instruments. Andrew C. Morrison (Natural Sci., Joliet Junior College, 1215 Houbolt Dr., Joliet, IL 60431, amorriso@jjc.edu)

One of the most significant acoustic instruments developed in the 20th century was the Caribbean steelpan. The steelpan family originated from the islands of Trinidad and Tobago when the islands were a British colony. Because musical traditions on the island were heavily reliant on percussion, local musicians quickly adopted the new drum type. Other factors contributing to the development of the steelpan family include the oil drilling industry and the use of the islands as naval facilities leading up to the second world war. The steelpan has primarily always been tuned by hand by individual tuners, as attempts to mass-manufacture the instruments have largely failed. The most recent innovations in steelpan construction contributed to the development of the instrument known as the Hang, a hand-played instrument. The popularity and scarcity of the Hang have given rise to a new percussion instrument known as the handpan.

Contributed Paper

2:45

3pMU6. Investigating the effect of drumstick diameter on drum sound level. Richard Silva (Veneklasen Assoc., 1711 16th St., Santa Monica, CA 90404, rsilva@veneklasen.com) and Wayland Dong (Veneklasen Assoc., Santa Monica, California, CA)

Acoustic drum sets typically generate high sound levels when played. Over decades, players have developed numerous techniques and manufacturers have created various products to reduce sound levels. Many of these approaches and products not only lower sound levels but also alter the drum sound, which can impact sound quality for the drummer and listener. For example, multi-rods, a collection of wooden rods wrapped together, have been created as a quieter alternative to traditional drumsticks. Perceptively, multi-rods do not replicate the sound of a drumstick, however, creating their own unique sound and vibe. The same can be said for brushes, a collection of retractable metal bristles held together by a hollow case. Reducing the diameter of a traditional drumstick design may generate lower sound levels while maintaining a traditional drum sound. This study investigates the effect on sound level due to changes in drumstick diameter.

3:00–3:20
Panel Discussion

Session 3pPA

Physical Acoustics and Noise: Meteorological Acoustics II

Roger M. Waxler, Chair
Univ. of Mississippi, P.O. Box 1848, University, MS 38677

Contributed Papers

1:00

3pPA1. Influence of the stratopause wind and temperature stratification on the waveform of the infrasound signal from the August 4, 2020 beirut explosion. 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 waveform of the infrasound signal received at a distance of 2398 km (infrasound station IS48) from the epicenter of the powerful explosion in Beirut, which occurred on August 4, 2020, is analyzed and modeled by using ray trace and pseudo-differential parabolic equation (PDPE) methods. Given a high temporal variability of the effective sound speed in the stratopause predicted by the European ECMWF model, we assumed that within the stratopause layer the increase in effective sound speed with increasing height is very small, on the order of 1 m/s. We also took into account the fine-scale layered structure of wind velocity and temperature in the real atmosphere, which is not taken into account in the ECMWF model. Accounting for the scattering of infrasound by strongly anisotropic (layered) inhomogeneities of the effective sound speed by using the PDPE wave method allowed us to explain for the first time the appearance of "fast" stratospheric arrivals, their time durations, time period between successive arrivals (about 110 s) and the waveform of the entire observed signal. [This work was supported by the Russian Science Foundation, Grant No 21-17-00021.]

1:15

3pPA2. Characteristics of Lamb waves from 2022 Hunga Tonga volcano eruption detected in the Moscow region. Sergey Kulichkov (Obukhov Inst. of Atmospheric Phys., 3 Pyzhevsky Per., Moscow 119017, Russian Federation, snik1953@gmail.com), Igor P. Chunchuzov, Oleg Popov, and Vitaly Perepelkin (Obukhov Inst. of Atmospheric Phys., Moscow, Russian Federation)

We present characteristics (waveform, coherence, phase speed, and propagation directions) of Lamb waves from 2022 Hunga Tonga volcano eruption detected by the network of microbarographs in the Moscow region. The pressure waves were detected both on the day of the volcano eruption (January 15 UTC) and after the time interval during which they circled the entire globe after the first detection and returned to the receiving point (on January 17 UTC). This made it possible to study the change in the waveform and duration of the Lamb wave depending on the distance from the volcanic eruption site and the influence of stratification of wind speed and atmospheric temperature on the signal waveform. [This work was supported by the Russian Science Foundation, Grant No. 21-17-0002.]

1:30

3pPA3. Frequencies of atmospheric Schumann resonances correlate with the frequencies of acoustically detected stellar massive particles. Igor Ostrovskii (Phys. and Astronomy, Univ. of Mississippi, 108 Lewis Hall Phys., University, MS 38677, iostrov@phy.olemiss.edu)

The Schumann resonances (SR) in Earth's atmosphere are electromagnetic waves of extremely low frequencies (ELF). The first four modes propagate at 7.4 Hz–26 Hz. Recently, the massive particles emitted by the Sun/stars have been detected by two acoustical methods including the acousto-electric properties of 10 MHz quartz oscillators and ELF vibrations of suspended lightweight crystals. In this work, the frequencies of solar/stellar massive particles propagating through the atmosphere and landing on Earth are revealed and compared with SR frequencies. The measurements of acousto-electric admittance Y of free vibrating quartz oscillators reveal that Y(F) are changed from known resonance-antiresonance curve to a pulse type dependency Yp(F) with a number of deep minima. They occur when piezoelectric crystal axis is oriented toward a star emitting massive particles. The waveforms (WF) of vibrating suspended light-weight quartz crystals were detected by a laser Doppler vibrometer. The Fast Fourier transform spectra obtained from the Yp(F) and WF time dependencies reveal the spectra of massive particles M(F) with which they were hitting the crystals. The M(F) spectra extend from 2 to ~30 Hz, which correlates with SR spectra. In addition, the maxima in SR modes and M(F) spectra are very close or coincide.

1:45

3pPA4. Mars soundscape: Review of the first sounds recorded by the Perseverance microphones. Baptiste Chide (Space and Planetary Exploration Team, Los Alamos National Lab., P.O. Box 1663, Los Alamos, NM 87545, bchide@lanl.gov), Ralph Lorenz (Space Exploration Sector, Johns Hopkins Appl. Phys. Lab., Laurel, MD), Naomi Murdoch, Alexander Stott, David Mimoun (ISAE-SUPAERO, Toulouse, France), Xavier Jacob (Institut de Mécanique des Fluides de Toulouse, Toulouse, France), Tanguy Bertrand (LESIA, Observatoire de Paris, Meudon, France), Nina Lanza (Space and Planetary Exploration Team, Los Alamos National Lab., Los Alamos, NM), Sylvestre Maurice (Institut de Recherche en Astrophysique et Planétologie, Toulouse, France), and Roger Wiens (Space and Planetary Exploration Team, Los Alamos National Lab., Los Alamos, NM)

On February 18, 2021, NASA's Perseverance rover landed in Jezero Crater carrying the two first microphones operating on the surface of Mars: the SuperCam microphone, positioned on top of the rotating rover's mast and the EDL microphone fixed on the body of the rover. Working flawlessly since then, they provide the first characterization of Mars' acoustic environment in the audible range and beyond, from 20 Hz to 50 kHz. Recorded sounds originate from three main sources: the atmosphere (turbulence, wind), the shock-waves generated by the Supercam pulsed laser ablating rocks, and hardware-induced artificial sounds such as the signal generated by the high-speed rotating blades of the Ingenuity helicopter. After one year, the Perseverance playlist features more than 5 hours of martian sounds. In addition to providing an unprecedented short timescale characterization of the wind, temperature fluctuations, and the turbulence dissipative regime,

this dataset highlights the unique sound propagation properties of the low-pressure CO2-dominated Mars atmosphere: acoustic impedance varying with the season, large intrinsic attenuation of the high frequencies, and the dispersion of the sound speed in the audible range. This presentation will review these results to date.

2:00

3pPA5. Spatial coherence of vertical atmospheric sound propagation. Matthew J. Kamrath (U.S. Army Engineer Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755-1290, matthew.j.kamrath@erdc.dren.mil), Vladimir Ostashev, D. Keith Wilson (U.S. Army Engineer Res. and Development Ctr., Hanover, NH), Michael J. White (U.S. Army Engineer Res. and Development Ctr., Champaign, IL), Carl R. Hart (U.S. Army Engineer Res. and Development Ctr., Hanover, NH), and Anthony Finn (Defence and Systems Inst., Univ. of South Australia, Mawson Lakes, South Australia, Australia)

Atmospheric turbulence causes the amplitude and phase of sound waves to fluctuate, which reduces the coherence of acoustic signals. Spatial coherence describes the similarity of two signals at different points in space, and a better understanding of acoustic coherence could lead to improved target detection, tracking, and identification. This presentation compares theoretical predictions and measurements of the acoustic spatial coherence. The theoretical coherence is derived by combining sound propagation theory with turbulence models that include the effects of atmospheric shear and buoyancy instabilities. To be applicable to vertical and slanted propagation, the turbulence models use height-dependent variances and length scales for the fluctuations in temperature, shear-produced velocity, and buoyancy-produced velocity. Instrumentation on a 135-m meteorological tower at the National Wind Technology Center (Boulder, CO) provided the required model input data. The coherence measurements used a ground-based source and nine microphones attached to the same meteorological tower. Overall, the theoretical model accurately approximated the measured spatial coherences, especially when the atmospheric turbulence was fully developed (e.g., in the afternoon on sunny days). The largest disagreement occurred for measurements taken at dawn. For a source frequency of 3.4 kHz and microphones that are 130 m high and 1.5 m apart, the measured coherence was 0.3–0.6 for sunny conditions and 0.8–0.9 for cloudy conditions.

2:15

3pPA6. Sonic anemometry on a high altitude balloon. Robert D. White (Mech. Eng., Tufts Univ., 200 College Ave. Medford, MA 02155, r.white@tufts.edu) and Don Banfield (Astrophysics and Planetary Sci., Cornell Univ., Ithaca, NY)

Recent results demonstrate the promise of sonic anemometry for accurate, three dimensional, high update relative wind measurements in high altitude (stratospheric) balloons and for planetary science missions on Mars, Venus, and other environments [Banfield, JASA (2016); White, ASA (2020); White, AIAA (2020)]. The Tufts sonic anemometer, operating at 40 kHz with six commercial piezoelectric transducers and custom electronics and frame, flew out of Fort Sumner, New Mexico on August 30 2021, reaching a height of 106°000 feet at float. The entire duration of the flight was 4 h and 42 min, with 2 h at float. The three axis sonic anemometry system collected sound speed and three dimensional relative wind velocity in two modes, at 3.4 Samples/sec or 1.5 Samples/sec. Results reported include comparison to balloon inertial and GPS motion to attempt to establish accuracy. Analysis is ongoing. [Work supported by NASA-NNX16AJ24G and NASA-80NSSC20M0007. Thanks to Chris Yoder and NASA Wallops Balloon Programs Office for technical support.]

WEDNESDAY AFTERNOON, 25 MAY 2022 PLAZA BALLROOM D, 1:20 P.M. TO 2:25 P.M.

3p WED. PM

Session 3pPP

Psychological and Physiological Acoustics: Auditory Neuroscience Prize Lecture

Tom Yin, Chair
University of Wisconsin-Madison, Madison, WI

Chair's Introduction—1:20

Invited Paper

1:25

3pPP1. Coincidences and delays in disguise. Philip X. Joris (Neurosciences, KU Leuven, Herestraat 49, bus 1021, Leuven B-3000, Belgium, Philip.Joris@kuleuven.be)

Binaural hearing is alluring not only because of its acuity, but particularly because it is more than twice as tractable as monaural hearing. It provides fertile space for the interplay of anatomical, physiological, behavioral, and computational approaches, for which the Acoustical Society is a major platform. The binaural and by extension the auditory community is a particularly pleasant one to get bashed by and we are grateful for the recognition by our peers of our research, through this award. I will review some of our findings: those that have been most surprising, rewarding, or plain fun, and some of which even have relevance for the understanding of monaural hearing. Coincidences and delays have been major ingredients of our research and illustrate this weird but—despite much current negative press covfefe—marvellous group effort we call science, which we are privileged to be part of.

Session 3pSC

Speech Communication: Children's Speech Intelligibility II

Pasquale Bottalico, Cochair
Department of Speech and Hearing Science, University of Illinois at Urbana-Champaign, 901 South Sixth Street, Champaign, IL 61820

Mary M. Flaherty, Cochair
Speech and Hearing Science, University of Illinois at Urbana-Champaign, 901 S Sixth Street, Champaign, IL 61820

Invited Papers

1:00

3pSC1. Assessing bottom-up and top-down effects on speech-in-noise recognition by adolescents with normal hearing or cochlear implants. Susan Nittrouer (Speech, Lang., and Hearing Sci., Univ. of Florida, 1225 Ctr. Dr., Gainesville, FL 32610, snittrouer@phhp.ufl.edu), Joanna Lowenstein, and Donal Sinex (Speech, Lang., and Hearing Sci., Univ. of Florida, Gainesville, FL)

The terms "bottom-up" and "top-down" effects are commonly used when measuring speech recognition in competing sounds to refer to listeners' abilities to exploit details of the sensory input and to use linguistic constraints, respectively. The magnitude of each effect has been assessed for listeners with cochlear implants, but typically in separate studies. We varied the availability of bottom-up and top-down information in a single experiment by generating sentences with three kinds of clause structure, from less to more restrictive, in three different babbled backgrounds, with sex and number of talkers varied. Next we examined the factors that might explain abilities to use bottom-up and top-down information in speech-in-noise recognition. Predictor variables included spectral modulation detection abilities, lexical-syntactic knowledge, verbal working memory, and phonological sensitivity. Participants were 14-year-olds with normal hearing (NH; 46) or with cochlear implants (CIs; 46). Adolescents with cochlear implants showed deficits in their abilities both to segregate the target speech from background babble and to benefit from restrictive syntactic constructions. For adolescents with NH, only verbal working memory explained variability in abilities to use bottom-up or top-down information in speech-in-noise recognition. For adolescents with CIs, only spectral modulation detection abilities explained their abilities to apply these effects.

1:20

3pSC2. Spoken word recognition by bilingual and monolingual children. Susannah V. Levi (Communicative Sci. and Disord., New York Univ., 665 Broadway, 9th Fl., New York, NY 10012, svlevi@nyu.edu) and Nandita Karthikeyan (Communicative Sci. and Disord., New York Univ., Singapore, Singapore)

Previous research has demonstrated a bilingual advantage in the domains of executive function and social processing, including a type of social processing tied to speech perception—listening to differences between speakers' voices. Despite numerous studies showing an advantage, the existence of such an advantage has also been widely challenged. In the current study, we test whether monolingual and bilingual children differ in spoken word recognition for speech produced by foreign-accented speakers, as it may be the case that bilingual children have more experience with foreign-accented speech. Children completed a spoken word recognition task in which they heard high and low familiarity words (e.g., "hair" vs "loom") mixed with noise. Logistic mixed-effects models revealed the expected effects of word familiarity (high > low) and of age (older > younger). No effect of bilingual status or of language ability were found. Although the findings do not demonstrate a benefit of bilingualism, we argue that it is important to report these findings given the controversy surrounding other bilingual advantage studies. [Work supported by NIDCD 1R03DC009851.]

1:40

3pSC3. Masked speech recognition in school-age children with Down syndrome. Lori Leibold (Hearing Res., Boys Town National Res. Hospital, 555 North 30th St., 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)

Masked speech recognition is poorer for children who are typically developing compared to adults, particularly when the masker is two-talker speech. Audibility, receptive vocabulary size, and executive function contribute to these age effects, all areas of weakness for children with Down syndrome. This study sought to determine whether these weaknesses put children with Down syndrome ($n = 15$, 5–17 yrs) at a disadvantage when listening to masked speech compared to age-matched children who are typically developing. Speech-in-noise and speech-in-speech was assessed using an adaptive, forced-choice procedure with a picture-pointing response. Audiological testing and standardized assessments of receptive vocabulary, non-verbal cognition, and executive function were also completed. Overall, children with Down syndrome performed more poorly than their typically developing peers. While the relationship between age and SRTs differs between the noise and speech maskers for children who are typically developing, this interaction was not observed for

children with Down syndrome. Controlling for age, SRTs for children with Down syndrome were associated with vocabulary and executive function scores with the noise masker but not the speech masker. Results suggest that masked speech recognition is more challenging and is limited by different factors for children with Down syndrome compared to those who are typically developing.

2:00–2:30
Panel Discussion

WEDNESDAY AFTERNOON, 25 MAY 2022 GOVERNORS SQUARE 16, 1:00 P.M. TO 3:35 P.M.

Session 3pSP

Signal Processing in Acoustics: General Topics in Signal Processing II

Trevor Jerome, Chair
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

3pSP1. Development of software for performing acoustic time reversal with multiple inputs and outputs. Adam D. Kingsley (Phys. & Astronomy, Brigham Young Univ., Provo, UT 84602, adamkingsley@gmail.com) and Brian E. Anderson (Phys. & Astronomy, Brigham Young Univ., Provo, UT)

At Brigham Young University, acoustic time reversal experiments are conducted in solids and fluid media. The experimental setup involves synchronized generation and acquisition hardware. The synchronized hardware allows for multi-channel generation, and in some cases, multi-channel acquisition of time reversal focusing. A LabVIEW application has been compiled to increase ease of use and repeatability for the students conducting experiments. Forward and backward steps of time reversal are conducted through this simple user interface. The software is also able to control a 2D positioning system that allows for the measurement of the spatial extent of a time reversal focus. Modifications to traditional time reversal processing, such as inverse filtering and one-bit processing, may be easily implemented in the software. This presentation describes the hardware and software that help to facilitate time reversal research. Ideas from this presentation may help others develop time reversal data acquisition systems.

1:20

3pSP2. Time reversal focal amplitudes as the focal position approaches boundaries. Jay M. Clift (Phys. & Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, j@clift.org), Brian D. Patchett (Phys. & Astronomy, Brigham Young Univ., Orem, UT), Adam D. Kingsley, Lucas A. Barnes, McKayla Townsend, and Brian E. Anderson (Phys. & Astronomy, Brigham Young Univ., Provo, UT)

Time reversal (TR) is a signal processing technique that may be used to generate a focusing of waves at selected positions in reverberant environments. This study looks at the increase in the focal amplitude as the distance between the focal location and a boundary is decreased. Previous studies with audible sound in rooms have shown experimentally and numerically that there is a 3 dB increase in focal amplitude if the focal location is positioned 1 cm away from a wall; the numerical model used was a modal summation simulation. It has since been found that using an image source model simulation does not show any increase in focal amplitude as the focal location approaches boundaries. Experiments conducted in solid media with ultrasound showed that focal amplitudes increase by 6 dB at a free boundary. Experiments and modal summation modeling conducted with audible sound in a reverberation chamber also yielded similar results as the focal location reached the wall. The presentation will discuss why the image source model does not yield these results.

1:35

3pSP3. Super-resolution, time-reversal focusing using path-diverting scatterers. Emily D. Golightly (Phys. & Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, 23emilyg@gmail.com), Adam D. Kingsley (Phys. & Astronomy, Brigham Young Univ., Provo, UT), Rebekah Higgins (Civil Eng., Brigham Young Univ., Provo, UT), and Brian E. Anderson (Phys. & Astronomy, Brigham Young Univ., Provo, UT)

This presentation will discuss the possibility of using path-diverting scatterers in time reversal focusing to achieve super resolution. In particular, use of a one-dimensional pipe system with varying lengths of diverting pipes is shown to decrease the effective wave speed. This provides insight, albeit in an extreme case, of how scatterers can force sound to travel a longer path as the waves converge to the focal location. As the effective wave speed decreases, the spatial extent of the focusing decreases, creating an apparent super resolution when compared to the speed of sound in an unaltered pipe system. Previous work achieved super resolution with what they described as scatterers, but it is likely that their scatterers are better classified as resonators, and consequently, it has yet to be shown that scatterers can be used to obtain super resolution.

1:50

3pSP4. Imaging watermelons. David J. Zartman (None, 3441 S Garfield Ave. Loveland, CO 80537, zartman.david@gmail.com)

Watermelons are enjoyed by many all over the world. The challenge to find the perfect ripeness desired is one most easily answered using acoustic imaging, though most would commonly refer to this as a thump test. Basically, impact from a flicked finger does not do damage to the fruit, but can create an impulse in the watermelon, generating a response within the fruit that can be both felt and heard. How can this be interpreted for best results though? Watermelons start out firm and fibrous, and as they ripen fluid becomes more prevalent and sweeter, until there is more fluid than fiber and the fruit is deemed old and squishy. Thus, the two extremes are pure fibrous solid and a bag of fluid. The actual quest is in-between the two, finding the optimal balance. When there is both fiber and fluid in the fruit, it is very resonant. The amount of resonance directly corresponds to the balance between fiber and fluid. Some prefer more fiber for more crunch, and some prefer more fluid for a sweeter juicier product. Knowledge of how watermelons ripen and appropriate acoustic imaging techniques can allow the common grocery shopper to acquire their optimal product.

2:05

3pSP5. Extended imaging applications. David J. Zartman (3441 S Garfield Ave. Loveland, CO 80537, zartman.david@gmail.com)

Going back to the very basics of imaging techniques, all imaging requires a source, a target, and a receiver. Modern technology has expanded these capabilities in ways that might not be expected. A few basic principles still apply though. Taking a measurement affects the system is a well-known concept illustrated by the life or death of Heisenburg's cat. Tom Clancy's executive orders also defines a way to trace information pathways by using unique enticing differentiability. In 2015, OPM was hacked twice, compromising assumedly everyone's information—is there thus a way to determine who is now intimately familiar with all of our information that wasn't previously? If everything is compromised, are there ways to defend privacy anyway and compromise those believed to be incognito while compromising others? How does this apply to acoustics? A person practices a speech by themselves in an empty room. They are determined to be crazy talking to themselves by the person monitoring the room discreetly onsite. Then the person practicing a speech gets a critique of their speech by someone discreetly monitoring offsite unknown to the onsite personnel. Knowledge of the actual audience, both as the source speaking, and the receiver in the audience, is impactful.

2:20–2:35 Break

2:35

3pSP6. Numerical modeling and verification of room impulse response inference from the coherence properties of speech and music signals. Erin Driscoll (Elec. and Comput. Eng., Univ. of Rochester, 120 Trustee Rd., Comput. Studies Bldg., Rochester, NY 14620, edrisco5@ur.rochester.edu), Mark Bocko, and Sarah R. Smith (Elec. and Comput. Eng., Univ. of Rochester, Rochester, NY)

In previous work, we presented theory and motivating examples to demonstrate that the coherence properties of the harmonic partials of speech and music signals recorded in an acoustic space give information about the impulse response of the space. By analyzing auto- and cross-correlations between pairs of harmonics of the signal, we may "blindly" obtain the frequency-dependent reverberation time of the acoustic space. Assuming that the signal contains harmonically related partials with mutually correlated amplitude and phase modulations, a reasonable assumption for sources such as musical instruments or the human voice, this approach allows inference of information about the impulse response of an acoustic space. In this work, we verify this method for a variety of signals with different coherence times and amounts of pseudo-random amplitude and frequency modulations.

2:50

3pSP7. An n-order hold approach for fractional-delay interpolation in auralization. Andrew Christian (Structural Acoust. Branch, NASA Langley Res. Ctr., 2 N. Dryden St., M/S 463, Hampton, VA 23681, andrew.christian@nasa.gov), Randall Ali (Dept. of Elec. Eng., KU Leuven, Leuven, Belgium), and Eric Greenwood (Dept. of Aerosp. Eng., The Penn State Univ., State College, PA)

During the signal processing chain of an auralization simulating the propagation of a sound from a moving source to a stationary receiver, it is often necessary to interpolate between the samples of the source signal in order to arrive at uniformly-spaced samples at the receiver. In some cases, this interpolation is done in the receiver time frame—where the "input" samples of the source have become irregularly spaced due to time dilation effects. Canonical band-limited interpolation methods (i.e., sinc and sinc-derived approaches) cannot be applied in this case as they rely on having a uniformly-spaced input. The use of geometric interpolation methods that can handle irregularly-spaced input may not be grounded in signal processing principles and may produce unwanted artifacts and noise. This presentation outlines the possibility of embedding an irregularly spaced n-order hold signal within a highly over-sampled uniformly-spaced signal and then processing down to the desired sampling rate through successive decimations. Initial distortion and noise characteristics of the approach are shown for some basic propagation geometries. The possible benefits of using such an approach in an auralization scheme with a time-varying Doppler shift are discussed including: the prevention of aliasing, processing time advantages, and real-time processing.

3:05

3pSP8. Phonetic realization of vowel length and glottalization in Todos Santos Mam. Jennifer Kuo (Linguist, Univ. of California Los Angeles, 3125 Campbell Hall, Los Angeles, CA 90095-1543, jennifer.x.kuo@gmail.com) and Noah Elkins (Linguist, UCLA, Santa Monica, CA)

Mam (Guatemala: Mayan) is described as having a vowel length contrast, as well as a laryngeal contrast between glottalized and modal vowels [England 32–36 (1983)]. Glottalized vowels are described as having a falling pitch contour and being longer relative to their modal counterparts (*ibid.*). However, there are no phonetic studies explicitly confirming this. In particular, the Todos Santos dialect has received almost no phonetic investigation, and the presence and realization of the length and laryngeal contrasts is not well-established. This study is one of the first phonetic studies of vowel length and glottalization in Todos Santos Mam. The authors collected data of one native speaker reading a list of words varying in vowel quality. We find evidence for both a length contrast and laryngeal contrast. For the length contrast, short vowels are significantly shorter in duration and lowered relative to long vowels. For the laryngeal contrast, glottalized vowels are found to have a falling pitch contour, while modal vowels have level/rising pitch. In addition, a lengthening effect was found such that short glottalized vowels are longer than their modal counterparts, but it is unclear whether the same effect is present for long glottalized vowels.

3:20

3pSP9. Comparison of waveform estimated using inverse filtering with direct measurement of the volume flow at the glottal exit. Jacob J. Michaud (Dept. of Biomedical Eng., Univ. of Cincinnati, 231 Albert Sabin Way, MSB, Rm. 6303, Cincinnati, OH 45267, michaujj@mail.uc.edu), Liran Oren, Charles Farbos de Luzan (Otolaryngol., Univ. of Cincinnati, Cincinnati, OH), Ephraim Gutmark (Dept. of Aerosp. Eng., Univ. of Cincinnati, Cincinnati, OH), and Sid M. Khosla (Otolaryngol., Univ. of Cincinnati, Cincinnati, OH)

Inverse filtering is a signal processing technique used to estimate the glottal waveform from the speech signal. Although this technique is regularly used in research studies, it has never been formally validated because of the difficulties in obtaining direct measurements of the glottal airflow. The objective of this study is a first step towards validating this technique by comparing its estimated glottal waveform with flow measurements taken simultaneously at the glottal exit. The setup is based on synthetic vocal folds connected to a vocal tract model. Direct measurements of the volume flow at the glottal exit are taken using time-resolved tomographic particle image velocimetry. A circumferentially vented pneumotachograph (i.e., Rothenberg) mask is connected to the vocal tract and used with the Glottal Enterprise system to calculate the glottal waveform using inverse filtering. Effect of varying subglottal pressures and effect of near-field (i.e., the minimal gap between the false vocal folds) and far-field (i.e., oral opening) constrictions were also investigated. Results show that overall using inverse filtering gives a good approximation of the glottal flow waveform, but its accuracy can change depending on the constrictions in the vocal tract. The clinical implication of these findings will be further discussed.

WEDNESDAY AFTERNOON, 25 MAY 2022 GOVERNORS SQUARE 14, 1:00 P.M. TO 3:15 P.M.

Session 3pUW

Underwater Acoustics: General Topics in Underwater Acoustics I: Modeling and Measurements

Kevin D. Heaney, Cochair
Applied Ocean Sciences, Fairfax Station, VA 22039

Anatoliy N. Ivakin, Cochair
University of Washington, 1013 NE 40th Street, Seattle, WA 98105

Contributed Papers

1:00

3pUW1. Normal mode acoustic modeling using measurements of ocean sound speed from the Beaufort Sea. Jessica Desrochers (Ocean Eng., The Univ. of Rhode Island, 13 Gilroy St. Apt 2, Newport, RI 02840, jfothergill@uri.edu), Lora Van Uffelen (Ocean Eng., Univ. of Rhode Island, Narragansett, RI), Sarah E. Webster (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Alexander P. Muniz (Naval Undersea Warfare Ctr., Div. Newport, Newport, RI), Cristian E. Graupe, and Luis O. Pomales Velázquez (Ocean Eng., Univ. of Rhode Island, Narragansett, RI)

Over the last few decades, environmental changes in the Arctic have resulted in a subsurface acoustic duct located between 100- and 300-m depth, known as the Beaufort Duct. This subsurface duct allows for long-range acoustic transmission with little to no interaction with the sea surface or seafloor. In a 2017 long-range acoustic tomography experiment, two Seagliders traversed between five active sources moored within the duct which transmitted linear frequency modulated (LFM) sweeps centered around 250 Hz. These Seagliders were equipped with conductivity, temperature, depth (CTD) sensors as well as passive acoustic receivers. The environmental measurements were used to create sound speed profiles for input into broadband parabolic equation and normal mode acoustic propagation models. The normal mode models provide physical insight into the relationship between the peak arrival and the final cutoff of the ducted acoustic receptions. Modal group speeds from the predictions are used to interpret the acoustic arrival patterns measured on the Seagliders.

1:15

3pUW2. The Arctic underwater soundscape today and as projected for 2030. Kevin D. Heaney, Christopher Verlinden (Appl. Ocean Sci., Fairfax Station, VA), Kerri D. Seger (Appl. Ocean Sci., 2127 1/2 Stewart St., Santa Monica, CA 90404, kerri.seger.d@gmail.com), Jennifer Brandon (Appl. Ocean Sci., San Diego, CA), Leila Hatch (NOAA, Scituate, MA), Martha Schönau (Appl. Ocean Sci., LLC, Pensacola, FL), and Andrew Heaney (Appl. Ocean Sci., Fairfax Station, VA)

Canada, the United States, and the World Wildlife Fund are co-sponsoring ongoing work in the Arctic Council's Protection for the Marine Environment Working Group to evaluate shipping noise in the Arctic region. Applied Ocean Sciences has used ship tracking and sea ice data to model the region's underwater soundscape to improve understanding of radiated noise generated by shipping throughout the PanArctic. Current (2019) models have been compared with ambient noise measurements collected during time periods when vessel sounds were identifiably present and when biological sounds were not. Projections of ice cover and shipping routes along and between the northern borders of Arctic countries were used to forecast potential future (2030) Arctic soundscapes. Focused interpretation of these model results within sub-regions, time periods, and frequencies important to marine fauna and in turn to indigenous peoples will be provided to PAME and other fora seeking to guide the development of shipping practices and mitigation strategies. The final results will be incorporated as a PAME/Arctic Council product. This presentation will focus on the acoustic modeling work under projected sea ice conditions, maps of "excess noise" induced by ships in 2019 and 2030, and risk assessment for a few endemic marine mammal species.

1:30

3pUW3. Rapid soundcape generate using energy flux modeling. Kevin D. Heaney (Appl. Ocean Sci., 11006 Clara Barton Dr., Fairfax Station, VA 22039, oceansound04@yahoo.com)

The impact of anthropocentric sound on the living marine environment has been a growing area of concern. Human induced sound sources include shipping, siesmic exploration, pile driving, US Navy sonars, and miscellaneous (rare) explosions. The most ubiquitous and least regulated sound source is shipping, which dominates the ocean soundscape for frequencies below 500 Hz in much of the worlds oceans. Modeling of the ocean soundscape has been conducted using adiabatic modes, the parabolic equation and ray-tracing. In this paper, we present the modeling of the ocean soundscape (shipping and wind) with an energy flux algorithm. This analytic approach is very efficient and can handle mildly range-dependent environments. The flux model is used to generate regional sound models. With the inclusion of horizontal and vertical angle information at the model source position, this model can generate the noise directionality efficiently and can be then use

for the performance of any particular array against directional noise in the ocean environment.

1:45

3pUW4. Midfrequency sound transmission from mobile sources in a deep Arctic ocean at short and medium ranges: Time-domain modeling and data analysis. Anatoliy N. Ivakin (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, aniv@uw.edu), John E. Joseph, and D. Benjamin Reeder (Oceanogr., Naval Postgrad. School, Monterey, CA)

Midfrequency (2300–3000 Hz) LFM sweeps (3 s-long, one per minute) and CW 57 s-long signals at multiple frequencies (950, 1050, 1150, 2800, 2900, and 3000 Hz) were transmitted by two mobile sources EMATTs moving at three knots in circular patterns under ice in the Beaufort Sea on March 12, 2016. Acoustic pressure time series were recorded on five vertically and horizontally separated receivers Acousonde, which provided different combinations of source-receiver ranges (0.3–10 km) and depths (45–183 m). A time-domain model of propagation and reverberation is suggested that accounts for refraction in water column and scattering from ocean boundaries (bottom and ice). Effects of spatial variations of the environment, particularly the two shallow ducts, at 0–75 m and 75–250 m water depths, as well as bottom and ice roughness, are demonstrated. Preliminary results of the model-data comparisons are presented. Possibilities for remote sensing of the arctic environment are discussed. [Work supported by ONR.]

2:00–2:15 Break

2:15

3pUW5. Towards developing a generalized semi-coherent three-dimensional energy flux model. Mark A. Langhirt (Penn State Univ., 201 Appl. Sci. Bldg., Graduate Program in Acoust., University Park, PA 16802, mal83@psu.edu), Charles W. Holland (Elec. and Comput. Eng., Portland State Univ., Portland, OR), Ying-Tsong Lin (Woods Hole Oceanographic Inst., Woods Hole, MA), Sheri Martinelli, and Daniel C. Brown (Penn State Univ., State College, PA)

The energy flux method can be interpreted as an integration over propagation angle of a semi-classical (WKB) modal continuum, which yields an averaged incoherent intensity field with depth-dependent intensity bands that decay in range. Range dependence is typically incorporated via the adiabatic modes approximation and use of the ray invariant to map propagation angles. Though the energy flux method averages out nearly all of the field structure, it has the primary advantages of avoiding finding eigenvalues and eigenrays and its computational effort does not directly scale with frequency or range. The method has also been extended in the past decade to include a convergence factor derived from the interference of neighboring modes, producing a semi-coherent solution that captures some convergence structure analogous to high-frequency caustics. Since the development of the semi-coherent energy flux method, it has so far only been applied to Nx2D environmental models, but should theoretically be applicable in a 3D environment without the assumption of azimuthal symmetry or the exclusion of horizontal refraction. This paper will discuss the theoretical derivation and numerical implementation of a generalized semi-coherent three-dimensional energy flux model. [This study is funded by the NDSEG Fellowship program.]

2:30

3pUW6. Exploring surface source distributions for ocean ambient noise interferometry with airgun shots. John Ragland (Elec. and Comput. Eng., Univ. of Washington, 185 W Stevens Way NE, Seattle, WA 98195, jhrag@uw.edu) and Shima Abadi (Univ. of Washington, Seattle, WA)

Ambient noise interferometry utilizes the cross-correlations of ambient sound to estimate the time domain Green's function (TDGF). We have previously shown that ambient noise interferometry can resolve multi-path arrivals between two bottom-mounted hydrophones separated by 3.2 km, at a depth of 1500 m, and located 470 km off the Oregon coast. In 2019, a seismic reflection survey was conducted directly over the two hydrophones for 28 days covering a 763 km^2 area. The airgun shots occurred every 37.5 m while the ship moved at a speed of $\sim$4.5 knots, equivalent to a shooting interval of 16 s. The hydrophone recordings during this survey provide the unique opportunity to understand the effects of the surface source distribution on the noise cross correlation function (NCCF). In this talk, we show the sensitivity of the NCCF to the surface source locations using simulated and experimental data. Then, we use the image source method to analytically define the location of the sound sources that contribute to different delay times in the NCCF. [Work supported by ONR.]

2:45

3pUW7. Underwater sound characteristics from down-the-hole pile drilling for a coastal construction activity. Shane Guan (Div. of Environ. Sci., Bureau of Ocean Energy Management, 1315 East-West Hwy., Silver Spring, MD 20910, guan@cua.edu), Tiffini Brookens (Marine Mammal Commission, Bethesda, MD), and Robert Miner (Robert Miner Dynamic Testing of Alaska, Inc., Manchester, WA)

Sound generated by pile installation using a down-the-hole (DTH) hammer is not well documented and differs in character from sound generated using conventional impact or vibratory hammers. This study describes underwater acoustic characteristics from DTH pile drilling during the installation of 0.84-m shafts within 1.22-m steel piles associated with a cruise ship terminal construction in Ward Cove, Alaska. The median single-strike sound exposure levels measured at 10 m were 138 and 142 dB re 1 μPa^{2}s for each of the two piles, with cumulative sound exposure levels of 188 and 193 dB re 1 μPa^{2}s at 10 m, respectively. The sound levels measured in this study were significantly lower than previous measurements of DTH pile driving, and the sound is determined to be less impulsive in this study as compared previous studies. These differences likely result from fact that the DTH hammer used at Wade Cove did not make direct contact with the pile, as had been the case in previous studies. Further research is needed to investigate DTH piling techniques and associated sound-generating mechanisms and to differentiate the various impulsive structures from anthropogenic underwater noise in general.

3:00

3pUW8. Analysis of underwater noise due to hurricanes in the northern Gulf of Mexico. Jamie D. Hiern (Phys., Univ. of New Orleans, 2000 Lakeshore Dr., New Orleans, LA 70148, jhiern@uno.edu), Kendal Leftwich, and Juliette W. Ioup (Phys., Univ. of New Orleans, New Orleans, LA)

The proposed study will use broadband deep-water ambient sound field information collected by stationary and mobile passive acoustic monitoring platforms during a hurricane passage in the northern Gulf of Mexico. EARS buoys will be used for stationary data collection and reconnaissance flights of Slocum gliders equipped with hydrophones for mobile collection. The goal is to use these data to reconstruct storm wind speed distributions and to quantify changes in the ocean environment during hurricanes and tropical storms. The majority of this research is the signal processing involved in analyzing the collected noise data. Signal processing techniques employed include Fourier transform frequency analysis, Wavelet transform signal analysis, Bayesian signal processing, and machine learning, all of which have been applied to similar acoustic data in the past. Machine learning techniques are used to compare and correlate analyses from both bottom-moored EARS and moving glider data. Comparisons of data from gliders and from stationary EARS buoys can be made with similar analyses of recorded stationary EARS buoy data from the same and nearby locations over several recent years, now with a focus on estimating wind speeds from acoustic noise analysis. [This material is based upon the work supported by the Office of the Under Secretary of Defense for Research and Engineering under Award No. FA9550-21-1-0215.]

Plenary Session and Awards Ceremony

Maureen Stone
President, Acoustical Society of America

Presentation of Certificates to New Fellows

Julien Bonnel – For advances in time-frequency analysis of underwater sound

Rochelle Newman – For understanding of speech perception and language development in challenging listening conditions

Andi Petculescu – For exploring the acoustics of extraterrestrial environments

Erica Ryherd – For advancements to acoustics in the healthcare industry

Introduction of Award Recipients and Presentation of Awards

William and Christine Hartmann Prize in Auditory Neuroscience to Philip X. Joris

Silver Medal in Noise to Paul D. Schomer

R. Bruce Lindsay Award to Meaghan O'Reilly

Helmholtz-Rayleigh Interdisciplinary Silver Medal to George Augspurger

Gold Medal to Michael J. Buckingham

Vice President's Gavel to Joseph R. Gladden

President's Tuning Fork to Maureen Stone

OPEN MEETINGS OF TECHNICAL COMMITTEES

Technical Committees of the Acoustical Society of America will hold open meeting on Tuesday, Wednesday, and Thursday evenings.

All meetings will begin at 7:30 p.m., except for Engineering Acoustics which will meet starting at 4:45 p.m. on Tuesday and Computational Acoustics which will meet starting at 4:30 p.m. on 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 discussion.

Committees meeting on Tuesday

Engineering Acoustics (4:45 p.m.)	Governors Square 10
Acoustical Oceanography	Governors Square 14
Animal Bioacoustics	Governors Square 17
Architectural Acoustics	Plaza Ballroom A
Physical Acoustics	Governors Square 11
Psychological and Physiological Acoustics	Plaza Ballroom D
Signal Processing in Acoustics	Governors Square 16
Structural Acoustics and Vibration	Governors Square 12

Committees meeting on Wednesday

Biomedical Acoustics	Governors Square 15

Committees meeting on Thursday

Computational Acoustics (4:30 p.m.)	Governors Square 12
Musical Acoustics	Directors Row H
Noise Plaza	Ballroom D
Speech Communication	Plaza Ballroom E
Underwater Acoustics	Governors Square 14

ACOUSTICAL SOCIETY OF AMERICA

Silver Medal in

Noise

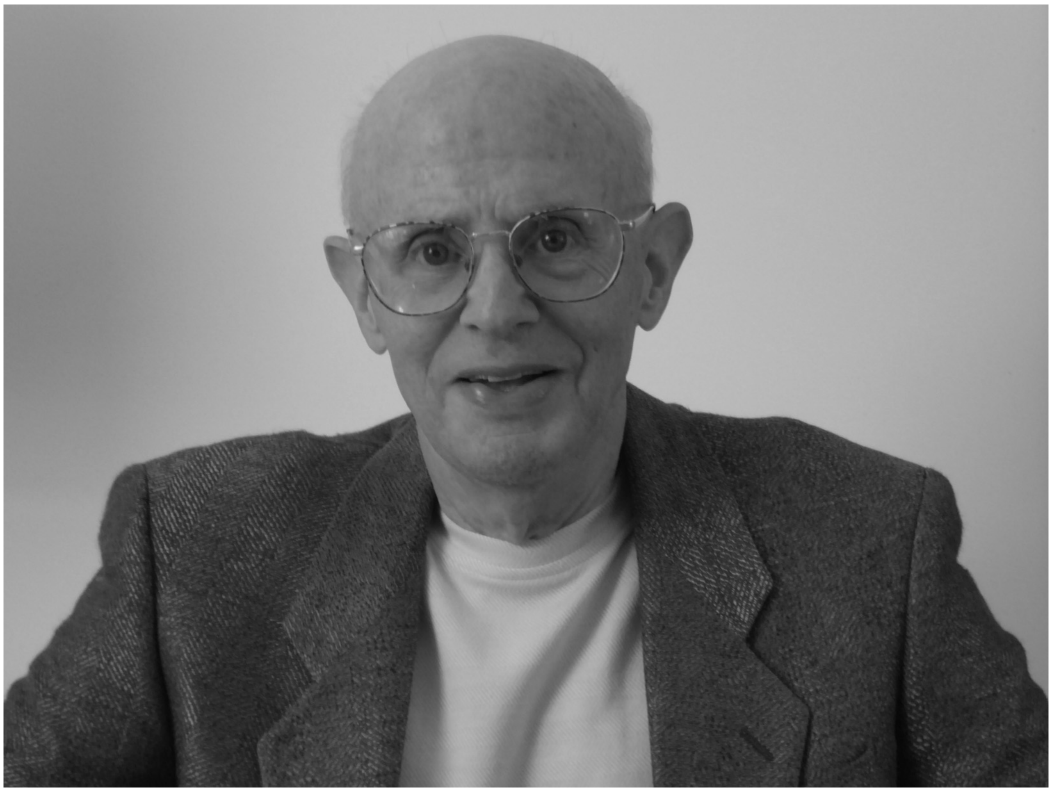
Paul D. Schomer

2021

The Silver Medal is presented to individuals, without age limitation, for contributions to the advancement of science, engineering, or human welfare through the application of acoustic principles, or through research accomplishment in acoustics.

PREVIOUS RECIPIENTS

Harvey H. Hubbard	1978	Larry H. Royster	1999
Henning E. von Gierke	1981	Louis C. Sutherland	2002
William W. Lang	1984	Alan M. Marsh	2006
Tony F. W. Embleton	1986	Michael R. Stinson	2009
William J. Galloway	1988	Keith Attenborough	2012
George C. Mailing, Jr.	1992	Scott D. Sommerfeldt	2020
Kenneth M. Eldred	1994		



ENCOMIUM FOR PAUL D. SCHOMER

...for contributions to the understanding of the sources and effects of noise and for leadership in national and international acoustical standards

25 MAY 2022 • DENVER, COLORADO

Paul Schomer is a distinguished scientist and expert with over 50 years of experience in acoustics, noise control, and measurements of the effects of noise on people and communities. He has made significant contributions to quantifying and standardizing the effects of environmental noise exposure as well as to the understanding of the effects of noise from aircraft, wind turbines, gunfire, blast noise, sonic booms, transportation, construction, and industry.

Paul has published over 50 papers in peer-reviewed journals, including 26 papers in the *Journal of the Acoustical Society of America* (JASA). He has authored chapters in books on community noise and electroacoustics, authored or co-authored 59 technical presentations, and chaired numerous special sessions at ASA meetings. Paul is a Fellow of the ASA and of the Institute of Noise Control Engineers.

Paul received his B.S. degree in Electrical Engineering from the University of Illinois in 1965, an M.S. degree in Electrical Engineering-Acoustics from the University of California, Berkley in 1966, and a Ph.D. in Electrical Engineering-Acoustics from the University of Illinois in 1971. His thesis was entitled: "Sound Transmission Loss Between Spaces Connected by Multiple Paths: A New Measurement Technique." He holds a PE license from the District of Columbia.

Paul was an Adjunct Full Professor in the Department of Electrical and Computer Engineering, University of Illinois, and Environmental Noise and Acoustics Team Leader at the U.S. Army Construction Engineering Research Laboratory (CERL) in Champaign, Illinois from 1971–2001. The goal of this division was to resolve or mitigate National Environmental Policy Act issues surrounding military training ranges. At CERL, Paul studied the effects of impulsive and low frequency noise sources, blast noise, weapons firing, helicopter, and construction noise. Paul realized the importance of community noise exposure and the variables between noise dose and the effects of noise on people. He developed computational models for prediction of ground-to-ground blast noise propagation and designed some of the first general purpose, field-portable, digital noise monitoring instruments. In 1990, he was selected US Army Corps of Engineers' Engineer of the Year, and was designated One of the Top 10 Federal Engineers of the Year by the National Society of Professional Engineers.

Paul was the seventh and longest serving Standards Director of the ASA, and received the ASA Distinguished Service Citation in 2015. Over the 14 years of his term, he chaired and served on numerous ASA national and international standards committees. Paul led the development of the ASA/ANSI S12.9 series of environmental noise standards and championed the development of the first ASA/ANSI S12.60 Classroom Acoustics standard. One of Paul's traits, which led in part to his success in standards development, was to understand the concerns of critics of a standard, to engage with them, and to turn them into advocates. When ASA/ANSI S12.60 was balloted and approved, a trade association appealed to ANSI. Although the appeal was ultimately unsuccessful, Paul worked with them, and as a result of this positive interaction, the trade association supported a revision and a new part of the standard.

From 1993–2018, Paul was convenor of the ISO TC/43/SC1 Working Group 45 on Environmental Noise, during which time several of the ISO 1996 series of environmental noise standards were developed. These international standards later became the basis of the ASA/ANSI 12.9 standards. In 1995, Paul became Chair of the U.S. Technical Advisory Groups to ISO TC/43 Acoustics and ISO TC43/SC1 Noise, with the responsibility of approving the recommended U.S. position on all committee drafts from these committees. Paul was also head of the U.S. delegations to all of the international meetings of these two ISO standards committees through 2014.

He founded his own consulting firm, Schomer & Associates, Inc. in 2003 where his work included studies of the effects of wind turbine noise and other sources, and investigation of soundscapes, especially in national parks. Paul continued his involvement in standards and worked with engineering students from the University of Illinois, encouraging them to make presentations at ASA meetings, and to co-author papers in *Proceedings of Meetings on Acoustics* (POMA) and JASA.

Another of Paul's important contributions was the development with Sanford Fidell and Vince Mestre, of the Community Tolerance Level (CTL), a figure of merit used to predict the percent of persons in a community highly annoyed by transportation noise. CTL is based on the theory that annoyance tracks loudness growth, (i.e., A-weighted rms sound pressure to the 0.3 power). Community annoyance, which is influenced by a number of factors not related to noise level, can therefore be described by CTL, which is normalized to the median day-night noise level (DNL) for the best fit to the loudness growth function for a given community. Paul shepherded the CTL method to become part of the international standard: ISO 1996-1:2016 "Acoustics — Description, measurement and assessment of environmental noise — Part 1: Basic quantities and assessment procedures". It is also included in ASA/ANSI S12.9-2005/Part 4 (R2020) "Quantities and Procedures for Description and Measurement of Environmental Sound – Part 4: Noise Assessment and Prediction of Long-term Community Response", currently under revision.

Paul was instrumental in resolving the disparity between Beranek's Noise Criteria (NC), for the assessment of the effects of room noise on speech intelligibility, and Warren Blazier's Room Criteria (RC), for the design of heating ventilation and air conditioning (HVAC) systems. While NC worked well for rooms with well-behaved HVAC systems, it did less well for turbulent fluctuating HVAC systems with high level low-frequency components. RC accommodated systems with low frequency fluctuations, but over-predicted perceived effects, leading to unnecessary or impossible design requirements. The Room Noise Criterion (RNC) metric – which Paul independently developed – combined and improved upon the best parts of the NC and RC metrics to effectively evaluate room noise while providing an accurate HVAC design method that properly addresses low frequency noise fluctuations. RNC is an integral part of the ASA/ANSI S12.2-2019 "Criteria for Evaluating Room Noise" standard.

Paul's family has always been very important to him. He married Susan in 1966, and they have two children, Beth and Jeffrey, and two grandchildren Madeleine and Russell. Paul enjoys time with his family and traveling with them. Paul is a wonderful and supportive father and grandfather, and he especially enjoys introducing new and challenging activities to them.

In recognition of his work in noise control engineering, his publications and presentations at ASA and other meetings, his improving our understanding of the effects of noise on community response, and his leadership of the ASA Standards Program, ASA awards the Silver Medal in Noise to Paul Schomer "For contributions to the understanding of the sources and effects of noise and for leadership in national and international acoustical standards."

CHRISTOPHER J. STRUCK
ROBERT D. HELLWEG, JR.

R. BRUCE LINDSAY AWARD

ACOUSTICAL SOCIETY OF AMERICA

Meaghan A. O'Reilly

2022

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

ENCOMIUM FOR MEAGHAN ANNE O'REILLY

…for contributions to biomedical ultrasound applications in the central nervous system

25 MAY 2022 • DENVER, COLORADO

Meaghan Anne O'Reilly grew up in Toronto, Canada as the fourth of six children. She decided early on that she could do anything her 5 siblings could. This philosophy was not without fail, but through her determination to meet any challenge set by her brothers and sister, she learned to pick herself back up and try again. She enthusiastically took up ice hockey, sailing (dinghies and tall ships), robotics, speech arts, chess and paddling (canoes and kayaks), among other hobbies.

Meaghan came later than some to her chosen career as an ultrasound researcher. Her intellectual curiosity and the breadth of her undergraduate training as a mechanical engineer (Queen's University, Canada) conspired to present a wide range of career path opportunities. She completed her MSc in Biomedical Engineering at the University of Oxford in 2008, with a focus on biomechanics, and with ideas about the design of prosthetics and an interest in accident reconstruction. Meaghan returned to Canada in the midst of the global financial crisis and, having been introduced to biomedical acoustics while at Oxford through courses taught by Constantin Coussios, she had just enough confidence to respond to an advertisement for a one-year contract position in the lab of Kullervo Hynynen at Sunnybrook Research Institute, Toronto. This set her off on a path towards her current position as Canada Research Chair in Biomedical Ultrasound.

In her first 6 months working in the Focused Ultrasound lab, Meaghan performed almost the same task each day: she would test a different way to assemble a PVDF receiver, sometimes spending hours on a given method, before seeing it ultimately fail in a variety of ways. Not one to give up, she successfully developed a reproducible design that could be tested *in vivo* for monitoring blood-brain barrier opening in rodents. Getting to participate in *in vivo* studies of ultrasound bioeffects sparked an even greater interest in her work. Near the end of her one-year contract, Meaghan was convinced to pursue a PhD, and in March 2010 she enrolled at the University of Eastern Finland. Making up for her late start in the field, she worked with enthusiasm and efficiency. By the time she stood for defense in June 2012, she had already published 7 first authored peer-reviewed papers.

One of the most notable contributions from Meaghan's doctoral work was the development of active treatment control for ultrasound-mediated blood-brain barrier opening based on the detection of cavitating microbubbles. Ultrasound and microbubbles hold great promise for the treatment of brain disorders in that the interactions of the acoustically stimulated microbubbles with blood vessel walls can transiently modify the permeability of the blood-brain barrier to allow drugs to reach the brain in therapeutically relevant quantities. A key challenge is calibrating the exposures such that sufficient effect is achieved without causing permanent tissue damage. The association of changes in acoustic spectra from the cavitating bubbles with successful blood-brain barrier opening was first reported in 2006. However, it was in 2012 that Meaghan was the first to demonstrate that acoustic pressure could be modulated during treatment based on the changes in recorded spectra (Radiology 263(1), 96-106 (2012)). This vastly improved the reproducibility and safety of subsequent preclinical studies, and a modified version of her control method is currently being used in clinical trials investigating blood-brain barrier opening in patients.

Following her doctoral studies, Meaghan continued making important contributions to biomedical ultrasound in the brain. She built a sparse receiver array integrated within a large aperture hemispherical transcranial array. With this device she published one of the seminal papers on ultrasound super-resolution imaging (Med. Phys 40 (11), 110701 (2013)), demonstrating super-resolved three-dimensional images of a phantom through an *ex vivo* human skull cap. She further showed the utility of such an array for mapping cavitation activity from bubble clouds during blood-brain barrier opening in rodents, and thus the potential for spatiotemporal control of treatments.

In 2015, Meaghan started her own lab at Sunnybrook Research Institute and the University of Toronto. A key focus of her independent program has been translating ultrasound therapy approaches to the spinal cord. She demonstrated the first evidence of positive therapeutic effect of ultrasound-mediated drug delivery to tumors situated in the spinal cord in a rat model (Sci. Rep 8 (1), 9013-27335 (2018)). She and her team also work extensively on developing the clinical scale approaches to translate this work to patients. They have shown the ability to focus sound through human vertebrae to the vertebral canal and have demonstrated successful ultrasound and microbubble mediated blood-spinal cord barrier opening in pigs through the intact spine (Theranostics 10 (17), 7758-7774 (2020)).

To date, Meaghan has published 48 peer-reviewed articles in the field of acoustics, and co-authored numerous issued and pending patents. In recognition of her status as an emerging international leader, in 2019 she was awarded the Tier 2 Canada Research Chair in Biomedical Ultrasound. In 2020 she received both the Frederic Lizzi Early Career Award from the International Society for Therapeutic Ultrasound, and the IEEE Ultrasonics Early Career Investigator Award from the IEEE Ultrasonics, Ferroelectrics and Frequency Control Society. Meaghan is an active member of the Acoustical Society of America. She has served on the Biomedical Acoustics Technical Committee since 2015 and serves on the Committee for Women in Acoustics and the Member Engagement Committee. She is also active in several other societies through leadership positions and committee memberships, and serves the community as a peer-reviewer for many journals, including *the Journal of the Acoustical Society of America*.

KULLERVO HYNYNEN

Helmholtz-Rayleigh Interdisciplinary Silver Medal
in
Architectural Acoustics and Engineering Acoustics

ASA
ACOUSTICAL SOCIETY
OF AMERICA

George L. Augspurger
2022

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

Interdisciplinary Silver Medal

Eugen J. Skudrzyk	1983
Wesley L. Nyborg	1990
W. Dixon Ward	1991
Victor C. Anderson	1992
Steven L. Garrett	1993

ENCOMIUM FOR GEORGE L. AUGSPURGER

…for contributions to design of recording studios, performance venues, and loudspeakers, and for decades of patent reviews

25 MAY 2022 • DENVER, COLORADO

George L. Augspurger has been active in the design of studio and other listening spaces and the design and theory of audio transducers for over 60 years. He earned a BA degree from Arizona State College in 1953 and a Master of Arts Degree from the University of California, Los Angeles in 1959, where his project was *The Influence of Stage Set Construction on Audience Intelligibility*, and did additional postgraduate work at Northwestern University.

After working in sound contracting and television broadcasting, he started his acoustic career at JBL in 1958. He first provided applications support as Technical Service Manager and wrote numerous technical explanatory articles which were published in the trade press. He was instrumental in the formation, and rose to the position of Manager, of the newly formed JBL Professional Products Division in 1968. He continued as Technical Director at JBL until 1970, leaving to start his own, still-active, consulting company, Perception, Inc., where his activities range from studio and theater design in his architectural practice to loudspeaker and transducer technology in electroacoustic engineering.

George has enjoyed a continued reputation as a skilled practitioner the design of, and for correcting acoustical flaws in, studios and listening rooms. He combines technical knowledge with a gift for what is practical and affordable for the client. Many of his studio projects are still in daily use in Los Angeles and other recording centers worldwide. His clients include Capital Records, the Dorothy Chandler Pavilion, Stanford University, the Hollywood Bowl, and over 100 others. Although he recently celebrated his ninety-second birthday, he remains an active contributor to these fields, consulting on studio projects, teaching an electrical engineering course for ten years in electroacoustics at USC, and writing monthly reviews of patents, published in both the *Journal of the Acoustical Society of America (JASA)* and the *Journal of the Audio Engineering Society* (JAES).

George introduced new knowledge into loudspeaker design. An early example was an unusual double-chamber loudspeaker enclosure in 1961. It generated considerable publicity and was a favorite of loudspeaker builders. Later in his career he became interested in transmission line loudspeakers, which had never received a thorough analysis. He developed an electroacoustic analog circuit, made extensive tests on it, and was then able to predict the performance of various pipe geometries and damping techniques. The project was published in 2000 as "Loudspeakers on Damped Pipes," in JAES 48, pp. 424-436.

He developed new analysis techniques based on his damped pipe theory by writing and offering a software program to analyze speaker and enclosure configurations. There were several unique features to his program. First, it could model arbitrary, complex geometries. Second, the damping coefficients were derived from tests on real materials at varying densities, not theoretical guesswork. Actual tests were then made on pipes of various lengths and geometries. Third, as a further confirmation of its accuracy, George worked out the alignments for three different common pipe geometries.

While his consulting, teaching, and research are notable, it is his published work, specifically his patent reviews, that are perhaps best known. They have appeared in virtually every issue of the Journal since 1980 and make him arguably the most prolific author in the history of JASA. These eagerly awaited reviews reflect his encyclopedic knowledge of prior art and serve as a learning opportunity for working professionals. To date their number is approximately 4800. The depth of his understanding and near photographic recollection of their history are truly remarkable and are eloquent teaching tools to advance or correct the fundamental principle being patented. Each draw on his encyclopedic knowledge of transducer and system history as well as basic principles and is occasionally sprinkled with an acerbic dry wit.

His additions to the technical literature are also memorable. These include 18 papers on acoustics, another 9 papers in the JAES, three JASA publications, and one *Acoustics Today* invited paper along with several invited talks to local ASA chapter meetings and national ASA conventions as well as local and national AES meetings. He has chaired Architectural Acoustic sessions at ASA national meetings, such as "Sound Recording and Reproduction, the Listening Interface" in 1980, and a session on acoustical patents in October 2005.

He authored the JBL Sound Workshop Manual, which was later incorporated by John Eargle into the JBL Sound System Reference Design Manual (2nd Ed., 1986 and 3rd Ed, 1999). A more recent contribution to the literature was an invited paper in an *Acoustics Today* issue on studio design, "Control Room Design: The Monitoring Environment," published in 2013.

An excellent example of his scholarly writings is his 1985 technical paper summarizing the progress in speaker design technology, "Theory, ingenuity, and wishful wizardry in loudspeaker design—A half century of progress?" (JASA, 77, No. 4, April 1985, 1303-1308). The paper succinctly presents Mr. Augspurger's encyclopedic knowledge of loudspeaker design, its history, its successes, and its failures, and is a must-read synopsis of the significant advances in the progress of the field.

His professional contributions have been recognized by many awards. He was promoted to the rank of Fellow of the Audio Engineering Society (1977), Fellow of the Acoustical Society of America (2003), and is a member of the U.S. Institute for Theater Technology, the National Council of Acoustical Consultants (NCAC), and more recently by the receipt of a special Technical Grammy Award from the Recording Academy (2020) to "pioneering audio engineer and designer George Augspurger" for contributions to the film and recording industries.

George Augspurger is a truly deserving member of the acoustical engineering community and a long serving contributor to the Journal and the technical body of knowledge it represents. We are honored to present this encomium for his Helmholtz-Rayleigh Interdisciplinary Silver Medal in Architectural Acoustics and Engineering Acoustics.

MARSHALL LONG
NEIL A. SHAW
MARK R. GANDER

Gold Medal

Michael J. Buckingham

2022

The Gold Medal is presented in the spring to a member of the Society, without age limitation, for contributions to acoustics. The first Gold Medal was presented in 1954 on the occasion of the Society's Twenty-Fifth Anniversary Celebration and biennially until 1981. It is now an annual award.

PREVIOUS RECIPIENTS

Wallace Waterfall	1954
Floyd A. Firestone	1955
Harvey Fletcher	1957
Edward C. Wente	1959
Georg von Békésy	1961
R. Bruce Lindsay	1963
Hallowell Davis	1965
Vern O. Knudsen	1967
Frederick V. Hunt	1969
Warren P. Mason	1971
Philip M. Morse	1973
Leo L. Beranek	1975
Raymond W. B. Stephens	1977
Richard H. Bolt	1979
Harry F. Olson	1981
Isadore Rudnick	1982
Martin Greenspan	1983
Robert T. Beyer	1984
Laurence Batchelder	1985
James L. Flanagan	1986
Cyril M. Harris	1987
Arthur H. Benade	1988
Richard K. Cook	1988
Lothar W. Cremer	1989
Eugen J. Skudrzyk	1990
Manfred R. Schroeder	1991
Ira J. Hirsh	1992
David T. Blackstock	1993
David M. Green	1994
Kenneth N. Stevens	1995
Ira Dyer	1996
K. Uno Ingard	1997
Floyd Dunn	1998
Henning E. von Gierke	1999
Murray Strasberg	2000
Herman Medwin	2001
Robert E. Apfel	2002
Tony F. W. Embleton	2002
Richard H. Lyon	2003
Chester M. McKinney	2004
Allan D. Pierce	2005
James E. West	2006
Katherine S. Harris	2007
Patricia K. Kuhl	2008
Thomas D. Rossing	2009
Jiri Tichy	2010
Eric E. Ungar	2011
William A. Kuperman	2012
Lawrence A. Crum	2013
Brian C. J. Moore	2014
Gerhard M. Sessler	2015
Whitlow W. L. Au	2016
William M. Hartmann	2017
William A. Yost	2018
William J. Cavanaugh	2019
Judy R. Dubno	2020
James F. Lynch	2021

ENCOMIUM FOR MICHAEL J. BUCKINGHAM

…for theoretical and experimental contributions to ocean acoustics and for service to the society

25 MAY 2022 • DENVER, COLORADO

Mike Buckingham was born in Oxford, England and attended the Orange Hill Grammar School for Boys where he was deputy head boy. Mike studied at the University of Reading and obtained his B.Sc with Honors in 1967 and Ph.D. in 1971, both in physics. He is a Chartered Engineer in the U.K. and Fellow of the Institute of Acoustics (U.K.), Institution of Engineering and Technology (U.K.), and Acoustical Society of America. Mike currently holds the position of Distinguished Professor in Ocean Acoustics at Scripps Institution of Oceanography, UC San Diego and was awarded the Pioneers of Underwater Acoustics Medal from the Acoustical Society of America in 2017. Mike holds 5 patents for acoustic innovations.

Mike has had a lively career in underwater acoustics, full of innovative experiments, fundamental theoretical advances, and enormous contributions to our scientific culture, including mentoring a generation of young minds and dedicated service to our society. Early in his career, Mike studied ambient sound in the Arctic, deploying vertical line arrays of his own design from Royal Air Force (RAF) flights across the marginal ice zone. He describes these campaigns in colorful terms; RAF pilots of the time had their own view of what constituted a "safe vertical distance" above the ocean and skirting with the waves on these flights left an indelible impression. Not at all put off, Mike's use of aircraft as deployment platforms for experiments and sources of sound has spanned his entire career. Indeed, Mike is himself a private pilot, with airplane and glider ratings and qualifications in aerobatics. More than one graduate student, postdoctoral scholar, or visiting colleague has experienced both the joy and terror of aerobatic glider flights with Mike!

Mike's career is built on the foundation of innovative experimental approaches integrated with new and rigorous theoretical constructs. This powerful approach has advanced the field of underwater acoustics in topics spanning an enormous range, from volcano and seafloor acoustics to the acoustics of bubble plumes and ambient sound in the deep ocean trenches. Mike introduced the name and concept of "Acoustic Daylight" to underwater acoustics, which refers to the use of ambient sound to image the ocean environment. This innovation, published in the journal *Nature*, prompted the development of Acoustic Daylight monitoring sonar systems in the United States, Australia, and Singapore roughly two decades ago and, more recently, has spurred studies of how marine life may use something analogous to acoustic daylight to sense their underwater environment. Mike developed a new model of wave propagation in marine sediments, the Viscous-Grain-Shearing theory. This elegant model is easy to implement yet predicts wave properties that are in close agreement with data and has been widely accepted by the underwater acoustics community. Such sweeping developments, stemming from a single transformative concept developed with imaginative physical insight and elegant mathematical analysis, can also be seen in Mike's analysis of seafloor acoustics and his early work on the three-dimensional propagation of sound in shallow water environments.

Mike has mentored a generation of graduate students and postdoctoral scholars, teaching them the discipline of underwater acoustics. His approach is rigorous, with full expectation of excellence in scholarship, but guided by kindness and good humor. Mike's mentoring style has proven very effective, providing young minds with a roadmap for a fulfilling and productive career in ocean acoustics. Many of Mike's mentees have gone on to have their own scintillating academic careers and winning their own accolades. In addition to passing on his knack for seeing into the heart of a problem and tackling it with physical and mathematical analysis, one of Mike's hallmarks is his patent ability to train young scientists in the art of scientific communication. A successful academic career requires the effective communication of ideas and many of Mike's students have won Best Student Paper awards at international conferences over the years. The appreciation, gratitude, and affection of these scholars toward Mike was abundantly clear at the winter 2019 Acoustical Oceanography special session in his honor, which was a well-attended and cheerful celebration of Mike's career.

Mike has made significant and lasting contributions to our society over the past 30 or more years, including election to the Executive Council for a 3-year term in 2010 and election to the position of Vice President (VP), which is also essentially a 3-year term as VP-elect, VP, and past-VP. His participation on committees includes the Membership Committee, two 3-year terms on the Medals and Awards Committee, and numerous other commitments. Mike, along with Hank Medwin, David Farmer and Van Holiday, formed the "Gang of Four" that created the Acoustical Oceanography Technical Committee (AOTC). Formed as a specialty group in 1989, AOTC was conferred full status as a Technical Committee in 1992 and Mike was its inaugural Chairman. The early days of AOTC were both tumultuous and exciting; Mike's calm and insightful leadership during this time were essential to the committee's continued success.

Mike's career in marine acoustics has been bountiful. We have all benefited from his leadership within the society. Those of us fortunate enough to be considered friends know Mike to be insightful and rigorous in his scholarship, generous with his time and energy, and encouraging in his mentorship. His contributions to us all and to the science we pursue mark him as an outstanding leader of acoustical oceanography and contributor to the mission of the Acoustical Society of America.

GRANT DEANE
DAVID FARMER

Session 4aAA

Architectural Acoustics, Musical Acoustics, Noise, and Education in Acoustics: Music Education Facilities I

Shane J. Kanter, Cochair
Threshold Acoustics, 141 W Jackson Blvd, Suite 2080, Chicago, IL 60604

Lauren Ronsse, Cochair
University of Nebraska-Lincoln, 1110 S. 67th St., Omaha, NE 68182-0816

David T. Carreon Bradley, Cochair
Occidental College, Los Angeles, CA 90041

Martin S. Lawless, Cochair
Mechanical Engineering, The Cooper Union for the Advancement of Science and Art, 41 Cooper Sq., Rm. 720, New York, NY 10003

Chair's Introduction—9:00

Invited Papers

9:05

4aAA1. The acoustical design of the Imig Music Building addition—An end user's perspective and an acoustician's perspective. David Kahn (Acoust. Distinctions, 400 Main St., Ste. 600, Stamford, CT 06901, dkahn@ad-ny.com)

The design of the Imig Music Building addition at the University of Colorado/Boulder, which started in 2017, culminated in a building opening in Spring 2021. The facility includes a dedicated recital hall, a large ensemble rehearsal room with flexibility to double as a recital hall, a choral rehearsal room, theatre rehearsal room, and dance rehearsal studio along with many music teaching studios, practice rooms, and other acoustically sensitive spaces to support the College of Music's program. The footprint of the building addition was very limited, resulting in an unusually large number of acoustically critical horizontal and vertical adjacencies. Perspectives on the design process and the successful resulting building will be presented by the end user and the acoustician. One goal of this presentation is to help attendees understand how to translate end user goals into acoustic design criteria and communicate those criteria to the other design team members (architects, engineers, and other specialty consultants such as theatre consultants).

9:25

4aAA2. Architecture and acoustics in an era of experimentation. Clifford Gayley (William Rawn Assoc., Architects, Inc., 27 School St., Second Fl., Boston, MA 02108, cgayley@rawnarch.com)

Architecture and acoustics are inextricably intertwined using the same tools (volume, shape, surface) to create places for the exchange of ideas transmitted through sound, movement, and words. While true for well-established building types such as concert halls, theaters, and opera houses, this intertwining is key for new, emerging building types that respond to changing client demands across higher education, cultural institutions, and the public sector. Several case studies (*The Pop-Up, The Cauldron, The Mash-Up, The Inversion*, and *The Immersion*) will explore: (1) How learning institutions are repositioning the arts with new types of facilities that encourage innovation, creativity, and transdisciplinary collaboration, within the Arts and across all disciplines; (2) How learning institutions are experimenting with spaces for incubation, rehearsal, and performance, blurring the line between front of house and back of house; (3) How learning institutions are reconceiving convening/performing with audience members as active participants; (4) How performance venues can be welcoming for all members of the community and deepen engagement with the world around them (whether urban or sylvan) while maintaining acoustic excellence; (5) How large and often underutilized lobby spaces can be reimagined as pop-up venues for impromptu sonic events; (6) How acoustics can enable programmatic mash-ups such as a public radio broadcast studio and café at a library front door, opening new possibilities for building inclusive communities.

9:45

4aAA3. Design thinking of the Multiform Theatre: A case study. Scott A. Crossfield (Theatre Projects, 47 Water St., South Norwalk, CT 06854, scrossfield@theatreprojects.com)

Multiform Theatres are perhaps the most complex theatres to design. They must support the artistic aspirations of musicians, actors, dancers, directors, designers, and visual and aural artists in one highly flexible, transformable space. A true Multiform Theatre seamlessly supports the creation of art through automated architecture altering the Artist/Audience relationship and adapting to the artist's spatial needs, while providing adaptable technical and acoustic opportunities that inspire new artistic forms. When done right, the architecture itself becomes part of the artistic process. How does one space change from a 625-seat Concert Hall to a 350-seat Recital Hall to a 250-seat Proscenium Theatre to a laboratory for Immersive Environments, Research and Experimentation? Join Scott Crossfield, ASTC from Theatre Projects as he guides you through the design thinking behind the new Multiform Theatre at Brown University—a radical, one-of-a-kind theatre machine designed to inspire innovative new art making, enable unprecedented artistic collaboration and serve as a hub for performance at the Brown University.

10:05

4aAA4. Audio/video consultants as interpreters and advocates. Tim Perez (Threshold Acoust., 141 W Jackson Blvd., Ste. 2080, Chicago, IL 60604, tperez@thresholdacoustics.com)

As technical designers in the architecture world, AV Consultants help architects realize their vision while preserving an appropriate and useful AV system for the people who will occupy the space for years afterward. By involving users in the design process and communicating with user-focused language, we can identify functional expectations and be free to engineer a coordinated system solution with the design team. Recognizing that many of our clients only see a big building project once or twice in a lifetime, an effective design process involves seeking input from a broad range of users, setting clear expectations, presenting design progress in a meaningful and context-based manner, and creating a sense of ownership for decisions made along the way. Users may start with lofty goals and idealistic aspirations for a project, but one that is designed and managed responsively can still be successful through expectation-setting and opening the process to those who will have to live with the results.

10:25–10:40 Break

10:40

4aAA5. Acoustical criteria and design approaches for music education and rehearsal spaces. Gary W. Siebein (Siebein Assoc.,Inc., 625 NW 60TH St., Ste. C, Gainesville, FL 32607, gsiebein@siebeinacoustic.com), Marylin Roa, Matthew Vetterick, and Jennifer Miller (Siebein Assoc., Gainesville, FL)

Case studies of three music education spaces will present acoustical measurement data with architectural features of the rooms compared to criteria from Sabine (1964), Pirn (1978), Gade (1988), Wenger (2008), and Tsaih (2011) among others to assess the sound qualities of the spaces. Among other qualities, reverberation time, room volume, hearing each other, and speech intelligibility are evaluated. Data are also compered to the Norwegian Standard NS 8178-2014. The case studies show how these criteria can be achieved given very different architectural systems employed in the rooms and describe the limits of where the qualities cannot be achieved due to the limits of program, budget, site, or space. Efforts are made to make each surface in the rooms acoustically productive in an optimized design to the extent practicable.

11:00

4aAA6. Sound isolation systems and HVAC noise control: The hidden acoustical systems in music practice and rehearsal spaces. Gary W. Siebein (Siebein Assoc.,Inc., 625 NW 60th St., Ste. C, Gainesville, FL 32607, gsiebein@siebeinacoustic.com), Matthew Vetterick, Jennifer Miller, and Marylin Roa (Siebein Assoc., Gainesville, FL)

An important part of the design and acoustical performance of music education spaces are the sound isolation systems employed to reduce sound bleed between and among different spaces and the designed control of noise and vibration from building mechanical systems. Three case studies present challenges addressed in actual projects. One is a multi-story fine arts building at a university where all of the major acoustical spaces are built as rooms within rooms and the HVAC is distributed from a large rooftop mechanical room. The second is a single storey building at a public high school where primary, secondary, and tertiary sound separations, and HVAC system zones are clearly defined and describe the basic architecture of the facility. The third describes the steps in design required to transform significant acoustical difficulties in sound transmission and HVAC system noise into an expressive architectural and acoustical space built on a very modest budget.

11:20

4aAA7. A space for teaching opera performance. Robin Glosemeyer Petrone (Threshold Acoust., 141 W Jackson Blvd, Ste. 2080, Chicago, IL 60604, robin@thresholdacoustics.com) and Scott Pfeiffer (Threshold Acoust., Chicago, IL)

Within an academic context, learning and practice of performance is as important as the performance itself. The Brockman Hall for Opera at Rice University's Shepherd School of Music has been designed to wrap the vocal performer with an intimate and immediate response without sacrificing envelopment achieved though reverberant energy. The result is a stage that allows the vocalist to stretch their sound to meet each of the 600 audience members while honing their skills of articulation. Orchestra musicians' training is expanded in a pit that can support a wide repertoire from the baroque to a grand opera of 70 musicians, providing a space that allows musicians hone their skills as they accompany voice on stage.

Session 4aBA

Biomedical Acoustics: General Topics in Biomedical Acoustics: Elastography and Therapeutics

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

Contributed Papers

8:00

4aBA1. Harmonic generation in a shear wave beam with spatially varying polarization in transversely isotropic soft tissue. John M. Cormack (Ctr. for Ultrasound Molecular Imaging and Therapeutics, and Vascular Medicine Inst., Dept. of Medicine, Univ. of Pittsburgh Medical Ctr., Pittsburgh, PA 15261, jmc345@pitt.edu) and Mark F. Hamilton (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Shear wave beams with spatially varying polarization are characterized here by stretching the medium in the plane perpendicular to the propagation direction. Such a beam is produced in a soft tissue by applying motion perpendicular to the source plane, as with a piston source, due to the quasi-incompressible nature of the medium. In the transversely isotropic (TI) soft tissue, such as muscle, this polarization causes extension and contraction in the fiber direction, resulting in nonlinear propagation effects that are unique to the TI tissue compared to propagation in an isotropic tissue [Cormack, *J. Acoust. Soc. Am.* **150**, 2566 (2021)]. A KZ-type equation is presented that describes a special case of propagation of a shear wave beam with spatially varying polarization in the TI soft tissue, accounting for material anisotropy along with leading-order effects of nonlinearity and diffraction. Solutions for harmonic generation from a time-harmonic source are presented for focused and unfocused Gaussian-derivative distributions of transverse displacement in the source plane. Of practical interest is the longitudinal displacement in the shear wave field, which possesses a second-harmonic component on the axis that results from stretching in the fiber direction. This second harmonic, which is absent in isotropic tissues, may be measured using existing elastography techniques.

8:15

4aBA2. Viscoelasticity inversion for arterial shear wave elastography. Tuhin Roy (Civil Eng., North Carolina State Univ., 3535 Ivy Commons Dr., Apt. 302, Raleigh, NC 27606, troy@ncsu.edu) and Murthy Guddati (NC State Univ., Raleigh, NC)

Arterial stiffness, an important biomarker of many cardiovascular diseases, is strongly influenced by the wall modulus. In shear wave elastography, the arterial wall is excited by acoustic radiation force, and the resulting wave propagation characteristics are used to infer arterial stiffness. In earlier ASA meeting, we presented a forward model that returns arterial wall motion for a given geometry, material properties of the waveguide, and input acoustic radiation force. In this work, we propose an inversion approach developed on this forward model to estimate arterial viscoelasticity, which is also considered an important biomarker. Recently, the phase-velocity dispersion is utilized to estimate the arterial wall elastic modulus. However, this approach fails to estimate the viscoelastic modulus since the dispersion curve is not strongly influenced by the arterial wall viscoelasticity. Conversely, the spatiotemporal representation of the wall velocity is sensitive to both moduli. Therefore, in the inverse model, we minimize the mismatch between the measured and simulated particle velocities to invert for the parameters of viscoelasticity, e.g., modulus and damping coefficient for the Voigt model, modulus, and exponent for the Spring-pot model. This talk would contain the details of the underlying formulations and numerical examples illustrating the effectiveness of the proposed approach.

8:30

4aBA3. Acoustic and viscoelastic characterization of hydrogel scaffolds to optimize preparation parameters for tissue engineering applications. Megan S. Anderson (George Washington Univ., 800 22nd St. NW, Washington, DC 20052, andersonm@gwmail.gwu.edu), Marshall McCraw, Lijie Grace Zhang, Santiago Solares, and Kausik Sarkar (George Washington Univ., Washington, DC)

Cell growth and differentiation rely upon both biochemical and mechanical cues. Though the focus has historically been biochemical, there is an increasing interest on the mechanical properties of scaffold materials within the field of tissue engineering. When the mechanical properties of a scaffold material more closely match with those of the intended native tissue, the functionality of the tissue scaffold is significantly enhanced. This study explores how an acoustic and viscoelastic characterization of a popular hydrogel, gelatin methacrylate (GelMA), can be used to inform the synthesis and fabrication parameters for a variety of tissue engineering applications. GelMA scaffolds exhibiting a range of mechanical properties were produced by varying the concentration of GelMA and by varying the curing time of the scaffolds. Pulse-echo ultrasound techniques were used to determine the sound speed and attenuation, revealing a significant dependence on the GelMA concentration. Compression tests were also performed, showing an increased stiffness with certain preparation parameters. Finally, atomic force microscopy experiments were used to obtain the storage and loss moduli of the hydrogels, revealing an increase in hydrogel stiffness at short time scales for both the GelMA concentration and curing time.

8:45

4aBA4. Breast cancer tissue stiffness assessment using transfer learning ultrasound elastography: A phantom study. Justin An, Tasneem Abdus-Shakur (Univ. of the District of Columbia, Washington, DC), and Max Denis (Univ. of the District of Columbia, 4200 Connecticut Ave. NW, Washington, DC 20008, max_f_denis@hotmail.com)

In this work, a phantom study is performed to investigate the feasibility of tissue stiffness assessment of breast cancer masses using transfer learning ultrasound elastography. A transfer learning ultrasound elastography model is developed to classify the breast masses into quantifiable Young's modulus (kilopascals, kP) values regimes. The transfer learning model combines features of ultrasound elastography elastograms and images from Google's deep learning models. The elastogram features used in the training and validation of the transfer learning model were obtained from a calibrated elastography phantom with inclusions having various Young's moduli. The model was tested on a breast phantom with spherical inclusions. Test results show that the transfer learning model yields greater than 88% validation accuracy. Further results and limitations of the transfer learning techniques will be presented and discussed.

9:00

4aBA5. Full-waveform shear wave elastography for imaging tumors.
Abdelrahman M. Elmeliegy (NC State Univ., 915 Partners Way, Raleigh, NC 27606, aaelmeli@ncsu.edu) and Murthy Guddati (NC State Univ., Raleigh, NC)

Shear wave elastography (SWE) is a method of reconstructing the stiffness of soft biological tissues by matching the observed and the simulated wavefields using an inverse optimization scheme. SWE reconstruction algorithms can be classified into two main categories, local and global methods. Global methods consider more complete physics of the waves, i.e., refraction and scattering, and, thus, have the potential to better characterize the heterogeneity of the domain. These approaches require full-waveform inversion (FWI), which is computationally expensive. More importantly, due to highly nonlinear nature, FWI has the limitation of converging to local minima, leading to erroneous reconstruction. In this work, we address this issue and propose a cost functional that not only reduces the nonlinearity of the FWI but also results in a reconstruction algorithm that is independent of push amplitudes, less sensitive to the initial guess, and has a better convergence behavior compared to the classical least-squares cost functional. In addition, we propose to utilize only a fraction of the measurements with the eventual goal of 3D reconstruction of tumors using limited ultrasound measurements. In this talk, we will present the details of the underlying formulation and examples showing the effectiveness of the proposed method.

9:15

4aBA6. Motion-corrected echo decorrelation imaging for monitoring *ex vivo* cardiac ablation.
Bahar Saremi, Peter D. Grimm, Elmira Ghahramani Z., Nicholas S. Schoenleb (Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH), Deeptankar DeMazumder (Internal Medicine, Univ. of Cincinnati, Cincinnati, OH), and T. Douglas Mast (Biomedical Eng., Univ. of Cincinnati, 3938 Cardiovascular Res. Ctr., 231 Albert Sabin Way, Cincinnati, OH 45267-0586, doug.mast@uc.edu)

Radiofrequency ablation for treatment of cardiac arrhythmia would benefit from real-time monitoring of ablation progress. Echo decorrelation imaging has been shown to predict local ablation but is susceptible to artifacts caused by heart motion. Here, a method for motion-corrected echo decorrelation imaging is tested in *ex vivo* ablation of bovine cardiac muscle with controlled motion. Experiments employed a Stockert 70 generator and a 4-mm Celsius catheter (Biosense Webster) with the ablation duration of 3 min at the target ablation temperature of 90 °C, resulting in thermal lesions of diameter about 1 cm. Cyclical motion was induced by a stepping motor system (Velmex) with velocities of 0–10 mm/s in the travel range of 1.5 cm. Pulse-echo ultrasound images were acquired by an Acuson SC2000 scanner with a Z6Ms transesophageal matrix array (Siemens). Throughout ablation at intervals of 5–6 s, sequentially paired bi-plane images were acquired as complex (IQ) beamformed echoes with an interframe time of 9–13 ms. Regions of interest (ROIs), one including the catheter location and one outside the ablation zone, were tracked using detection of the myocardial wall boundary and spatial cross-correlation. Echo decorrelation within the ablation ROI was then mapped and compensated to remove motion-induced decorrelation measured in the reference ROI. Receiver operating characteristic curves characterize accuracy for predictions of local myocardial ablation by motion-compensated echo decorrelation imaging as a function of the induced motion amplitude.

9:30

4aBA7. The evolution of microwave-induced thermoacoustic signal characteristics generated during pulsed microwave ablation.
Audrey L. Evans (Elec. and Comput. Eng., Univ. of Wisconsin-Madison, 1415 Eng. Dr. RM 3415, Madison, WI 53706, alevans3@wisc.edu), Susan C. Hagness, and Chu Ma (Elec. and Comput. Eng., Univ. of Wisconsin-Madison, Madison, WI)

Microwave-induced thermoacoustic (TA) signals are of emerging interest for monitoring microwave ablation (MWA) in real-time. TA signals can be generated using an interstitial ablation antenna with a pulsed microwave energy source. When a microsecond microwave pulse is absorbed by tissue, the tissue undergoes a small-scale temperature rise, inducing a thermoelastic expansion that leads to acoustic generation. TA signal characteristics are linked to the dielectric, thermal, and acoustic properties of the local ablation environment. These relevant properties evolve significantly during the ablation process. We conducted a simulation-based study to examine the evolution of microwave-induced TA signal characteristics generated during pulsed microwave ablation. We experimentally validated our multi-physics simulation model for a spatially uniform temperature profile. Then, using the validated simulation model, we investigated TA signals generated in tissue exhibiting spatially nonuniform temperature profiles that arise during MWA. We find that TA signal characteristics are highly influenced by the local environment temperature within the region of initial TA generation and, thus, contain rich information to be exploited for real-time ablation monitoring.

9:45–10:00 Break

10:00

4aBA8. Imaging mechanisms of therapeutic ultrasound in human islets.
Andrew W. Chen (Biomedical Eng., The George Washington Univ., 800 22nd St. NW, Ste. 5000, Washington, DC 20052, andrewchen@gwu.edu), Aleksandar Jeremic (Biological Sci., The George Washington Univ., Washington, DC), and Vesna Zderic (Biomedical Eng., The George Washington Univ., Washington, DC)

We have previously shown that therapeutic ultrasound (TUS) has a wide range of effects on rat pancreas cells. We aim to explore additional possible mechanisms of TUS on the pancreas in a more physiologically relevant human islet model with the goal of potential therapies for type-2 diabetes. Fluorescence microscopy was performed on live human islets with dyes that can label for intracellular calcium, sodium, and reactive oxygen species (ROS). During microscopy, TUS was applied at 800 kHz at 0.5 W/cm^2 with a pulse repetition frequency of 0.166 Hz and a 16.6% duty cycle. The amount of insulin released from human islets as a result of TUS application was quantified with an enzyme-linked immunoassay (ELISA). Thus far, we have noticed a trend of increased intracellular sodium and ROS generation due to TUS application; however, the percent increase in ROS appears to be less than for intracellular sodium. Our initial ELISA results show that TUS does appear to increase insulin release from human islets (0.55 ± 0.90 μU/ml min) when compared to control (-0.14 ± 0.19 μU/ml min), however, not as much as with the high glucose application at 20 mM (2.47 ± 3.54 μU/ml min). Ongoing studies aim to better characterize TUS effects.

10:15

4aBA9. Comparison between focused ultrasound and dry needling treatments in a murine Achilles tendinopathy model.
Molly Smallcomb (The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, molly.smallcomb@gmail.com), Sujata Khandare, Jacob C. Elliott, Meghan E. Vidt (The Penn State Univ., State College, PA), and Julianna C. Simon (The Penn State Univ., University Park, PA)

Tendinopathy remains a common and debilitating condition that shows mixed success rates in response to treatment. Focused ultrasound (fUS) histotripsy is a non-invasive therapy that can mechanically fractionate soft tissues through bubble creation, oscillation, and collapse. Although collagenous tissues show resistance to full mechanical fractionation, previous studies in *ex vivo* murine tendons found acoustic parameters that create microdamage while preserving mechanical properties. Our objective is to compare dry needling (DN) or fUS treatments in a murine tendinopathy model. Achilles tenotomy was performed in 26 rats. One week later, tendons were treated with DN (30G needle, five fenestrations over 20 s) or fUS (1.5 MHz, 1-ms pulses at 10 Hz for 60 s, $p_+ = 89$ MPa, $p_- = 26$ MPa). Blood was collected from the tail vein immediately before and after treatment and 1-week post-intervention. Tendons were harvested for histology or mechanical testing. Results show no differences between fUS and DN in the release of IGF-I and TGF-β healing factors. No differences were found between interventions and controls in H&E-stained histology. Stiffness and percent relaxation of DN-treated tendons were lower than controls (p = 0.0041, p = 0.0441, respectively), whereas fUS-treated properties were similar. From these results, fUS treatment may be a viable alternative to DN in the treatment of tendinopathy. [Funding: NSF-GRFP-DGE1255832 and NIH-NIBIB-EB027886]

10:30

4aBA10. Investigation of a noninvasive method for monitoring intracranial pressure using sheep skulls. Nicholas Cameron (Dept. of Phys., Central Washington Univ., 400 E. University Way, Ellensburg, WA 98926-7422, nicholas.cameron@cwu.edu) and Andrew A. Piacsek (Phys., Central Washington Univ., Ellensburg, WA)

It has been previously demonstrated that resonance frequencies of fluid-filled shells of simple geometry are shifted in proportion to fluid pressure [*J. Acoust. Soc. Am.* **131**, EL506 (2012)]. We investigate the applicability of this approach for measuring changes of intracranial pressure in ovine skulls. A catheter inserted through the foramen magnum of a complete sheep's head enables the control of hydrostatic pressure, which is measured independently using a pressure transducer inserted into the parenchyma through a drilled hole in the skull. The vibrational response is measured using a small modal impact hammer to gently tap the skull with small accelerometers mounted at different locations on skull. Acceleration normalized by impact force is computed in the frequency domain, averaged over multiple taps. Significant peaks in the response spectrum are identified and associated with vibrational modes observed using a laser Doppler vibrometer. The consistency with which peaks shift in frequency in proportion to ICP is reported. The effect of tap location and accelerometer placement is also explored.

10:45

4aBA11. Pulse wave propagation through a stenotic artery with plaque constituents of different stiffnesses: *In vitro*-experiment and fluid–structure interaction simulation. Nima Mobadersany (Dept. of Biomedical Eng., Columbia Univ., 630 West 168th St., Physicians & Surgeons 19-418, New York, NY 10032, nm3254@columbia.edu), Nirvedh H. Meshram, and Elisa Konofagou (Dept. of Biomedical Eng., Columbia Univ., New York, NY)

Vulnerable (typically, softer) carotid plaques may rupture leading to blockages downstream in the brain with high potential for ischemic stroke. Pulse Wave Imaging (PWI) developed by our group is an ultrasound-based technique for pulse wave visualization and pulse wave velocity (PWV) mapping. In this study, the accuracy of PWV estimation in stenotic vessels was assessed in simulation and phantom studies. Fluid–structure interaction simulations were performed in a stenotic (50%) vessel. Experimental stenotic (50%) polyvinyl alcohol (PVA) phantoms (2 mm thickness, 8 mm lumen diameter, and 165 mm length) with soft/medium/hard plaque constituents of different stiffness ($E_0 = 12$ kPa, 30 kPa, 94 kPa) were constructed matching the simulation geometry and parameters. Pulse waves were generated by a computer-controlled pump using a physiologic flow waveform. The RF frames were acquired by a Verasonics Vantage 256 system using a 5 MHz (L7-4) linear array. The acquired RF frames were beamformed using a delay and sum algorithm, and a GPU-based, 1-D cross correlation method was used to estimate the wall velocities. The acceleration peak of the pulse wave was tracked to estimate PWV. PWVs of 2.6 ms^{-1}, 3.4 ms^{-1}, and 4.5 ms^{-1} were, respectively, measured for the soft, medium, and stiff plaque material in experimental phantoms. The PWV of 2.7 m/s was confirmed for the medium plaque in simulation. This fundamental study demonstrated, thus, a good accuracy by PWI in the estimation of plaque PWV and its underlying stiffness for future stroke risk assessment.

11:00

4aBA12. The effects of nominal absorption in bone on the extinction lengths for scattering, absorption, and overall attenuation. Brett A. McCandless (MAE, North Carolina State Univ., Raleigh, NC) and Marie Muller (MAE, North Carolina State Univ., 911 Oval Dr., Eng. Bldg. III, Raleigh, NC 27606, mmuller2@ncsu.edu)

In cortical bone, the effect of absorption in the solid matrix on ultrasound scattering and propagation is not well understood, and for absorption, it is not easily incorporated into simulations and is difficult to measure independently from scattering attenuation. In this study, finite-difference time-domain simulations were conducted in maps derived from scanning acoustic microscopy images of human femur cross-sections. These maps were modified by controlling pore density (fixed at 10 pores/mm^2) and diameter distribution (ranging between 30 and 100 μm). Eight nominal absorption values, ranging from 0 to 100 dB/cm, were attributed to the solid matrix. Absorption in the pores was set to 0.01 dB/cm. Propagation of ultrasound pulses centered at 8 MHz was simulated in both backscattering and through-transmission configurations. The scattering, absorption, and overall attenuation extinction lengths were obtained. T-tests demonstrated that varying the absorption significantly impacts the extinction lengths for scattering, overall attenuation, and absorption. The differences in values for the scattering, attenuation, and absorption extinction lengths when absorption varied from 20 to 40 dB/cm are statistically significant (respective p-values: 9.1×10^{-14}, 3.3×10^{-5}, and 8.8×10^{-5}). It was also demonstrated that attenuation due to scattering dominates overall attenuation at low-nominal absorption values, whereas the converse is true at higher absorption values.

11:15

4aBA13. *In vivo* assessment of lymph nodes using quantitative ultrasound on a clinical scanner: A preliminary study. Cameron Hoerig (Riverside Res., 156 William St., Fl. 9, New York, NY 10038, choerig@riversideresearch.org), Kirk Wallace (GE Res., Niskayuna, NY), Jeffrey Ketterling, Ernest Feleppa (Riverside Res., New York, NY), Maoxin Wu (Pathol., Stony Brook Univ., Stonybrook, NY), and Jonathan Mamou (Riverside Res., New York, NY)

Quantitative ultrasound (QUS) methods evaluating the backscattered echo signal provide a means to infer the microstructural properties of soft tissues. High-frequency ($\geq$20 MHz) QUS has been used to detect metastases in excised human lymph nodes (LNs), but limited evidence exists about the efficacy of conventional-frequency QUS for characterizing LNs *in vivo* using a standard clinical scanner. In this study, 11 LNs from 11 patients with known primary tumors were scanned *in vivo* using a GE Logiq E9 scanner with a 10-MHz linear array. Metastatic status was determined by fine needle aspiration. Echo frames were processed using QUS methods to compute five parameters from the backscattered echo power spectrum and four from histograms of the echo envelope values. The spectral slope was found to be significantly higher in metastatic nodes ($p < 0.001$), whereas benign nodes exhibited significantly lower values of spectral intercept ($p < 0.001$), midband fit ($p = 0.004$), scatterer diameter (ESD, $p < 0.001$), and acoustic concentration (EAC, $p = 0.003$). A linear discriminator operating on ESD and EAC correctly differentiated benign and metastatic nodes with sensitivity of 0.89, specificity of 0.83, and area under the ROC curve of 0.91. These preliminary results demonstrate the feasibility of characterizing LNs *in vivo* at conventional frequencies using a clinical scanner.

11:30

4aBA14. Time-domain formulation for coherent plane wave compounding in horizontally layered media. John M. Cormack (Ctr. for Ultrasound Molecular Imaging and Therapeutics, and Vascular Medicine Inst., Dept. of Medicine, Univ. of Pittsburgh Medical Ctr., Pittsburgh, PA 15261, jmc345@pitt.edu), Marc A. Simon (Div. of Cardiology, Dept. of Medicine, Univ. of California San Francisco, San Francisco, CA), and Kang Kim (Bioengineering, Univ. of Pittsburgh, Pittsburgh, PA)

The locations of origin of ultrasonic echoes from soft tissue in conventional ultrasound beamforming are based on measured time of flight. For simplicity it is usually assumed that the region of interest is homogeneous with a constant sound speed, but in many *in vivo* and benchtop applications the imaging region is composed of horizontal layers, each with a distinct sound speed. In addition to time-of-flight complications, sound speed differences between the layers result in refraction through the interface of both the transmitted pulse and those reflected from the tissue, further influencing the fidelity of beamforming for target localization or other quantitative ultrasound metrics. A formulation is presented for delay-and-sum coherent Plane Wave Compounding (PWC) that accounts for variation in sound speed and refraction of ultrasound pulses in a region with two horizontal layers. The algorithm is formulated completely in the time domain, enabling simple implementation or extension of existing homogeneous PWC algorithms. Simulated and experimental images are investigated for a typical two-layer benchtop configuration, where the sound speeds in the upper and lower layers are relatively slow (water or gel) and fast (muscle), respectively. Extension of the formulation to a medium with N distinct layers is discussed.

11:45

4aBA15. Sidelobe reduction using null subtraction imaging. Zhengchang Kou (Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, B420 Beckman Inst., 405 North Mathews Ave., Urbana, IL 61801, zkou2@illinois.edu) and Michael L. Oelze (ECE, Univ. of Illinois at Urbana-Champaign, Urbana, IL)

In this work, we propose a novel anodization scheme where three different apodizations are introduced to create three separate images that are then combined nonlinearly to produce better image quality, i.e., null subtraction imaging (NSI). All apodizations were on receive only. The first apodization consisted of a zero mean weighting with the left half of aperture a weight of $+1$ and the right half of aperture a weight of -1. The second apodization was a DC offset version of the first apodization and the third apodization was a flipped version of the second. Images were created by combining the envelope signals of three apodizations with different weights. The images were subtracted from one another resulting in a reduction in sidelobe levels, a narrowing of the main lobe and improvement in image quality. An L14-5 array transducer and a Verasonics system were used to capture the RF channel data using seven plane waves spanning angles between $-12°$ and $12°$. Both coherent compounding and filtered incoherent compounding were used as a tradeoff between lateral resolution and contrast to noise ratio (CNR). Images were constructed from wire targets, a tissue-mimicking phantom, and rat tumors *in vivo*. Image quality was assessed through the CNR and intensity of sidelobes visible in the images. Images created using NSI were compared to images created using Hanning apodization with delay and sum (DAS) and a generalized coherence factor (GCF) approach. Improvements in image quality were found when using NSI compared to the other methods.

THURSDAY MORNING, 26 MAY 2022 PLAZA BALLROOM D, 8:00 A.M. TO 11:50 A.M.

Session 4aNS

Noise and Psychological and Physiological Acoustics: Health Effects of Noise

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

Lily M. Wang, Cochair
Durham School of Architectural Engineering and Construction, University of Nebraska-Lincoln, PKI 100C, 1110 S. 67th St., Omaha, NE 68182-0816

Chair's Introduction—8:00

Invited Papers

8:05

4aNS1. How to better design a research project to assess effects of transportation noise on humans. Ralph T. Muehleisen (Energy Systems, Argonne National Lab., 9700 S. Cass Ave. Bldg. 362, Lemont, IL 60439-4801, rmuehleisen@anl.gov)

The United States Air Force recently funded a research project to review the recent literature on noise and begin an update of the International Bibliography on Noise (IBON). Subject Matter Experts (SMEs) reviewed the literature on a variety of topics including human annoyance, non-auditory health effects, sleep disturbance, speech interference, child learning, animal effects, structural effects, and underwater effects. Each SME reviewed abstracts of thousands of papers and read hundreds of papers. As a result, the SMEs were able to see a vast range of methods for assessing effects of noise. In some of the topic areas, SMEs found that the number of really well-done research studies was actually quite small. While some papers did a good job of measuring or modeling noise, most of them did a poor job of assessing noise effects. While some papers did a good job of measuring or estimating noise effects, most of those did a poor job of measuring or modeling noise. In this presentation, one of the SMEs will present his thoughts on how to design a research project to do both a good job assessing transportation generated noise AND the effects of that transportation noise on humans.

8:30

4aNS2. Avoiding common pitfalls in health effects of noise research: Instructive observations from updating the International Bibliography on Noise. S. Hales Swift (Sandia National Labs., P.O. Box 5800, MS 1082, Albuquerque, NM 87123-1082, shswift@sandia.gov)

During a recent effort to update the International Bibliography on Noise (IBON), the present author reviewed papers involving human annoyance evoked by community noise as a subject matter expert. As part of that effort, a number of papers on physiological health effects and their relationship to annoyance were evaluated. Many of the research products reviewed relied on self-reported measures of noise or used noise annoyance as a proxy for noise exposure instead of using either predicted or measured exposure values. This approach introduces substantial uncertainty into the interpretation of the results. Indeed, a smaller set of studies that included both physical and perceptual measures of exposure shows substantial differences between the results obtained via these contrasted approaches to exposure assessment. Some of the multiple directions of causality involved in the relationship between noise sensitivity, noise annoyance, psychological traits, and health effects are discussed in an effort to provide a cautionary resource for future efforts to study these relationships in ways that avoid certain well-traveled pitfalls. [Funded by the Air Force Civil Engineer Center under an interagency agreement between the Air Force Civil Engineer Center and the U.S. Department of Energy/Argonne National Laboratory, No. SAND2022-0676 A.]

8:55

4aNS3. Long-term aircraft noise exposure and incident hypertension in National U. S. Cohort Studies. Junenette L. Peters (Environ. Health, Boston Univ. School of Public Health, 715 Albany St., Talbot 4 West, Boston, MA 02118, petersj@bu.edu), Daniel D. Nguyen, Stephanie T. Grady (Environ. Health, Boston Univ. School of Public Health, Boston, MA), Jaime E. Hart (Channing Div. of Network Medicine, Dept. of Medicine, Brigham and Women's Hospital and Harvard Med. School, Boston, MA), Eric A. Whitsel, James D. Stewart (Dept. of Epidemiology, Univ. of North Carolina, Chapel Hill, NC), and Jonathan I. Levy (Environ. Health, Boston Univ. School of Public Health, Boston, MA)

Few cohort studies have examined relationships between aircraft noise and health, none in the United States. We evaluated associations between aircraft noise and incident hypertension in three national cohorts of women, the Nurses' Health Studies (NHS/NHSII) and the Women's Health Initiative (WHI). Noise levels around 90 airports for 1995–2015 (in 5-year intervals) were modeled using the Aviation Environmental Design Tool and assigned to participants' geocoded addresses over time. Hypertension risks were estimated using time-varying Cox proportional-hazards models for day-night average sound level dichotomized at 45 and 55 decibels (dB), adjusting for fixed and time-varying covariates. Hypertension cases were 33,190, 28 256, 5299 over 0.7M,1.3M, and 0.09M person-years in NHS, NHSII, and WHI, respectively. In combined NHS/NHSII, we observed adjusted hazard ratios (HRs) of 1.03 (95% CI: 0.99, 1.07) for 45 dB and 1.07 (95% CI: 0.98, 1.15) for 55 dB cut-points. In WHI, we observed preliminary HRs of 1.03 (0.94, 1.16) for 45 dB and 0.93 (0.83, 1.05) for 55 dB cut-points. Overall, in these cohorts, we did not find clear evidence of a positive association between aircraft noise exposure and hypertension.

9:20

4aNS4. The neurophysiology of noise and health: Established and emerging neural networks. Tor H. Oiamo (Dept. of Geography and Environ. Studies, Ryerson Univ., 350 Victoria St., Toronto, ON M5B 2K3, Canada, tor.oiamo@ryerson.ca)

There is strong evidence for the increased risk of ischemic heart disease from excessive noise, but evidence also supports effects on other health outcomes such as stroke and diabetes and more generally suggests that excessive noise exposures can interfere with the homeostatic function of several physiological systems (e.g., cardiovascular, metabolic, and immune). This exposure stress-response mechanism is commonly dichotomized as direct through sleep disturbance or indirect through some form of cognitive processing due to disturbance or annoyance. In either case, chronic activation of the hypothalamic-pituitary-adrenal stress axis provides a mechanistic link between exposure and systemic stress, but there is a limited understanding of how other structures in the limbic system are involved, and consequently a limited understanding of how noise is processed and appraised among other sensory inputs and as a subjective experience. This is furthermore limited by exposure estimation methods and defining the acoustic and built environments that people experience. This paper provides a review of systemic stress activation in the context of emerging knowledge on sensory processing within the limbic system and highlights recent research on effects of sound and noise on mental health and cognition that are advancing our understanding of noise effects on health.

Contributed Papers

9:45

4aNS5. Utilizing mobile devices to facilitate collection of correlated noise data and psychological and physical effects. Kimberly A. Riegel (Phys., Queensborough Community College, 652 Timpson St., Pelham, NY 10803, kriegel@qcc.cuny.edu), Kaila Pulley (Phys., Queensborough Community College, Bayside, NY), and Robert M. Lefkowitz (none, West Nyack, NY)

Noise is a pervasive urban issue that effects cities all over the globe. Exposure to noise has been linked to both psychological and health effects including sleep disturbance, increase in cortisol levels, and psychological impacts. Determining the effects of noise on large communities can be challenging because getting accurate noise levels that directly correspond to physical and psychological feedback is difficult. This study developed a mobile device application that allows a participant to take a sound level measurement and provide some feedback about the perception of the noise as well as provide some basic health information. This crowd sources the collection of data so that the sound levels that are taken are also directly related to the psychological feedback given. Preliminary data and user feedback from the initial deployment of the application will be presented.

10:00–10:20 Break

10:20

4aNS6. Relationship between noise perception, vocal fatigue, and work-related stress among health workers in a hospital in Bogotá, Colombia. Lady Catherine Cantor-Cutiva (Universidad Nacional de Colombia, Carrera 30 Calle 45, Ciudadela Universitaria, Bogota 110110, Colombia, lccantorc@unal.edu.co) and Andrés Carrillo-Gonzalez (Universidad Nacional de Colombia, Bogota, Colombia)

To characterize the relationship between noise perception, vocal fatigue, and work-related stress among health workers. Cross sectional with the participation of 79 participants. Data regarding noise levels, voice functioning, and stress were collected through validated questionaries. Generalized Linear Models (GLMs) were used to evaluate the associations between the study variables. Our results show that more than 60% of health workers reported that noise conditions were high (47% considered that it did not affect their job performance, and 22% reported that it did). Concerning vocal fatigue, on average, health workers scored 8.30 for factor 1, 2.52 for factor 2, and 5.09 for factor 3, which suggests that participants were not feeling their voices very tired (factor 1), but they have physical discomfort associated with voicing (factor 2) and experienced improvement of symptoms after recovery (factor 3). Regarding stress levels, around 80% of participating health workers reported very high or high stress levels (61% for very high and 18% for high). Health workers are exposed to noise levels that affect their communication and mental health. Health care facilities should strive to keep noise conditions at its lowest levels to keep healthy conditions for communication and mental health.

10:35

4aNS7. Managing occupational noise from blasts. Gethin Manuel (Acoust. Res. Ctr., Univ. of Salford, Newton Bldg., The Crescent, Salford, Greater Manchester M5 4WT, United Kingdom, g.w.manuel@edu.salford.ac.uk) and David Waddington (Acoust. Res. Ctr., Univ. of Salford, Salford, Greater Manchester, United Kingdom)

The aim of this work is to examine the current best practice for the assessment of auditory hazards associated with occupational impulsive noise for a UK-based test site. A summary of the literature is presented on the measurement methodologies, hearing conservation, and relevant guidance for the selection of hearing protection within the context of near-field impulsive noise. The current U,K. best practice recommends the adapted HML method from Defence Standard 0027:2015 using a C-peak measurement of exposure to compare with the national regulations. On-site measurements were made at the DNV Spadeadam Testing and Research Site (STaR), which performs full-scale high-intensity blast testing. Analyses of the trials data show that the C-peak personnel exposures resulted in acceptable levels below the 140dBC$_{pk}$ limit within the national regulations when using the prescribed hearing protection, according to the adapted HML method. It is concluded that the adapted HML assessment is unclear on how excessive low-frequency content and multiple pressure peaks are accounted for. The implication of this work is that further research is required to improve current best practice to quantify the uncertainties introduced by the additional characteristics found within real impulsive noise exposures.

10:50

4aNS8. Peak sound pressure level decay from large C-4 detonations: Establishing a 140 dB threshold. Avery K. Sorrell (Dept. of Phys. and Astronomy, Brigham Young Univ., N281 Eyring Sci. Ctr., Provo, UT 84602, averysorrell137@gmail.com), Kent L. Gee, Reese D. Rasband (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), Steven Pemberton, and Kenneth E. Laintz (Los Alamos National Lab., Los Alamos, NM)

Far-field acoustical characterization of blast wave propagation from explosives is often carried out using relatively small shot sizes (less than 1 kg). This paper describes a series of eleven C-4 detonations, with shot sizes varying from 13.6 kg to 54.4 kg (30 to 120 lbs) that were recently measured at the Big Explosives Experimental Facility (BEEF) at the Nevada National Security Site. Pressure waveform data were recorded at up to nine different stations, ranging from 23 m to 2.7 km from the blast origin, with some angular variation. As part of examining blast overpressure decay with distance and comparing with literature, the data were analyzed from the context of human safety regulations. To provide improved guidance for BEEF personnel working distances, an empirical model equation was developed for the distance, as a function of shot size, at which the peak pressure level drops below 140 dB. A preliminary investigation into peak level variability due to wind was also conducted. [Work supported by Los Alamos National Laboratory.]

11:05

4aNS9. Occluded insertion loss from intracochlear pressure measurements during acoustic shock wave exposure. Nathaniel Greene (Otolaryngol., Univ. of Colorado School of Medicine, 12800 E 19th Ave., Aurora, CO 80045, nathaniel.greene@cuanschutz.edu), David A. Anderson (Rocky Mountain Div., Appl. Res. Assoc., Inc., Denver, CO), and Theodore F. Argo (Appl. Res. Assoc., Inc., Littleton, CO)

Auditory injuries are a common result of high intensity noise exposure, and hearing protective devices (HPDs) can mitigate this injury. Current evaluation methods use manikins to measure ear canal SPLs, but neglect alternate sound conduction pathways. We have previously reported intracochlear pressures in cadaveric human specimens to high-level impulse noise, revealing a substantial bone conducted component. Here, we estimate insertion loss during HPD use from intracochlear pressures in those same specimens. Cadaveric specimens were exposed to shock waves with peak overpressures of 7–83 kPa. Fiber optic pressure sensors were placed in the external, middle, and inner ears and responses measured with ears unoccluded and with four HPDs. Spectral insertion loss was calculated for each exposure level in the frequency domain. Insertion losses calculated from EAC pressures were comparable across levels, consistent with results from acoustic manikins. In contrast, insertion loss calculated from intracochlear pressures were generally lower in magnitude, but increased with the exposure level, likely due to substantial contributions of secondary transmission pathways. Unfortunately, variability in intracochlear pressures limit insertion loss estimate utility. As a proof of concept, we show that averaging multiple exposures increased signal-to-noise considerably, similar noise reduction strategies should be utilized in future studies.

11:20

4aNS10. Methods for validation of electromechanical evaluation of hearing protection. Theodore F. Argo (Appl. Res. Assoc., Inc., 7921 Shaffer Pkwy, Littleton, CO 80127, targo@ara.com), Andrew D. Brown, David J. Audet (Speech and Hearing Sci., Univ. of Washington, Seattle, WA), Nathaniel Greene, Caylin R. McCallick (Otolaryngol., Univ. of Colorado School of Medicine, Aurora, CO), Carol A. Sammeth (Physiol. & Biophys., Univ. of Colorado School of Medicine, Aurora, CO), David A. Anderson, and Gregory T. Rule (Appl. Res. Assoc., Inc., Littleton, CO)

Preventing hearing injury while maintaining situational awareness is of critical importance while using hearing protection devices (HPDs). Determining the optimal HPD for such varied conditions as operating machinery, riding in aircraft, and using firearms is a non-trivial task. Currently, the Noise Reduction Rating (NRR) is the only required HPD specification; other important characteristics of advanced HPDs are not reported in a standardized manner despite the availability of relevant test standards. Barriers to standardized performance reporting are the use of human subjects and the range of performance capabilities of commercially available HPDs. To complement existing test standards and support standardization of performance reporting across a broader range of performance capabilities, a suite of quantitative, sensor-based tests is being developed to quickly, inexpensively, and comprehensively evaluate candidate HPDs. The five-test battery evaluates signal quality, sound localization, level-dependent frequency response, electronic self-noise, and impulse noise response. In parallel, a set of human subject tests is being developed to validate the electromechanical metrics. Here, we will present (1) preliminary results comparing human performance with complementary electromechanical data and (2) a software tool under development incorporating the electromechanical tests to optimize HPD selection for application-specific hearing protection.

11:35

4aNS11. A comparison of methods to measure the sound attenuation of active noise reduction hearing protectors. Xiaolin Wu (Key Lab. of Noise and Vib. Res., Inst. of Acoust., Chinese Acad. of Sci., No. 21, North Fourth Ring West Rd., Haidian District, Beijing 100190, China, wuxiaolin@mail.ioa.ac.cn), Xiaodong Li, Jinqiu Sang, and Xiaobin Cheng (Key Lab. of Noise and Vib. Res., Inst. of Acoust., Chinese Acad. of Sci., Beijing, China)

Hearing protectors equipped with active noise reduction (ANR) functions have become popular and widespread in recent years. It is important to assess the sound attenuation performance of the ANR hearing protectors properly, which remains a problem. This work studies several typical measurement methods, including the real-ear attenuation at threshold (REAT), microphone in real-ear (MIRE), and the acoustic test fixture (ATF). In order to reduce the masking effect from the noise generated from the ANR electronic system when ANR is turned on during the REAT test, a modified tone sound instead of one-third octave band noise is used as the test signal. The results obtained from the different measurement methods on several ANR headphones and earphones are compared and the differences are discussed. It is found that the measurement differences among those methods are huge. The values of active insertion loss measured by the REAT method are much lower than those assessed by the MIRE procedure. This work contributes to the development of the test methods to assess the sound attenuation of the ANR hearing protectors.

THURSDAY MORNING, 26 MAY 2022 GOVERNORS SQUARE 11, 8:30 A.M. TO 12:00 NOON

Session 4aPA

Physical Acoustics: General Topics in Physical Acoustics I

Shane W. Lani, Chair
Georgia Tech, 11100 Johns Hopkins Rd., Laurel, MD 20723

Contributed Papers

8:30

4aPA1. Theory of parametric receiving arrays processing utilizing sum and difference frequencies. Christopher M. Bender (Johns Hopkins Appl. Phys. Lab., 11100 Johns Hopkins Rd., 8-220, Laurel, MD, christopher.bender@jhuapl.edu) and Shane W. Lani (Johns Hopkins Appl. Phys. Lab., Laurel, MD)

The majority of highly directive receive arrays obtains their directivity from an often fully populated (lambda/2 spacing) array in one or more dimensions. Digital beamforming allows for steering of these densely populated arrays in multiple directions; however, if there is only one primary direction of interest, a similar highly directive receiver can be generated with just a few elements. Parametric receivers, such as parametric sources, can obtain enhanced directivity via a "virtual aperture" or a region in the medium where nonlinear mixing of a high intensity pump wave and a signal wave interact. For a parametric receiver, the pump element(s) with a frequency, F, and receiver element(s) are separated by a distance, L, and the signals of interest are the sum and difference frequency ($F +/- f$) between the pump frequency and an incoming signal frequency, f. The directivity of these parametric receivers are comparable to the end fire response of a line array of similar length, L. This paper will review the highlights of parametric receivers as well as detail the analysis to extend this phenomenon to include the sum and difference frequencies for single element pairing and arrays of pumps and receivers.

8:45

4aPA2. Modeling receiver characteristics of parametric receiving arrays. Shane W. Lani (Johns Hopkins Appl. Phys. Lab., 11100 Johns Hopkins Rd., Laurel, MD 20723, shane.w.lani@gmail.com), Christopher M. Bender, and Tyler J. Flynn (Johns Hopkins Appl. Phys. Lab., Laurel, MD)

Parametric receivers, such as parametric sources, can obtain enhanced directivity via nonlinear mixing of a high intensity pump wave and an incoming signal wave. Receiver properties, such as directivity and sensitivity of a parametric receiving arrays, can be derived from the physical characteristics such as the separation distance of the pump and receive elements, the pump wave frequency, and the incoming signal wave frequency. The overall receiver characteristics are also dependent on the medium properties such as the acoustic nonlinear parameter and absorption, as the medium is key to mixing of the waves. The parametric receiver's noise floor is a function the noise around the signal band and the noise native at the resultant sum and difference frequencies. This paper will present the analysis of noise for parametric receivers as well as modeling results leveraging a k-space pseudospectral method solver of the Westervelt equation. Comparisons of the directivity of the parametric receivers will be made between the modeling to the analytic theory.

9:00

4aPA3. Acoustic scattering by smooth and rough elastic cylinders insonified by directional sonars. Miad Al Mursaline (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., 70 Pacific St., Cambridge, MA 02139, miad@mit.edu), Timothy K. Stanton, and Andone C. Lavery (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

Realistic sonars radiate spherically spreading waves and have directivity. Therefore, they insonify a target over a finite number of Fresnel zones and span a continuum of oblique incident angles, even when the center of the beam is at normal incidence. These effects strongly influence both the overall scattered pressure levels and resonances. For example, because of the spreading of the beam and associated oblique insonification within the beam, axially propagating guided waves are generated that would not have otherwise existed for an idealized incident plane wave. This investigation illustrates practical sonar properties and their effects on cylinder scattering both theoretically and experimentally. An approximate theoretical formulation for acoustic scattering by elastic cylinders is proposed in the form of a simple line integral accounting for these properties. Laboratory measurements are also presented to test the range of validity of the formulation for smooth cylinders under both monostatic and bistatic configurations. This study is further extended for the case of rough elastic cylinders.

9:15

4aPA4. Sea state induced sound transmission loss of atmospheric broadband sources. Andrea Vecchiotti (Mech. Eng., Catholic Univ. of America, Viale dei Gerani, 11, Senigallia (AN) 60019, Italy, a-vecchiotti@libero.it), Faith A. Cobb, Teresa J. Ryan (Eng., East Carolina Univ., Greenville, NC), Joseph Vignola, and Diego Turo (Mech. Eng., The Catholic Univ. of America, Washington, DC)

This work presents a numerical study conducted on atmospheric sound propagation over sea. In particular, it focuses on the sound pressure level prediction uncertainties induced by the water surface roughness. To quantify these uncertainties, the generalized terrain parabolic equation (GTPE) is used to model sound propagation above water surfaces at different sea states. Water roughness is pseudo-randomly generated using Pierson-Moskovitz ocean wave spectrum. Building on a previous result that has established a simple expression for predicting single frequency sound transmission loss as a function of sea state, this work extends the approach to broadband signal from 125 to 1000 Hz. This work presents relationships between fully developed sea states (up to sea state 4) and the uncertainties on sound pressure level predictions at distances up to 1000 m from the source. These relationships are presented for typical nocturnal thermal gradients and for different elevations from the water surface.

9:30

4aPA5. Experimental study of the level-dependent softening of carbon particle stacks. Guochenhao Song (Ray W. Herrick Laboratories/School of Mech. Eng., Purdue Univ., Ray W. Herrick Labs., 177 S., Russell St., West Lafayette, IN 47907, song520@purdue.edu) and J. S. Bolton (Ray W. Herrick Laboratories/School of Mech. Eng., Purdue Univ., West Lafayette, IN)

Recently, it has been observed that a particle stack's elastic modulus and damping change with incident levels. Hence, the measured normal incidence absorption coefficient also varies: i.e., the quarter-wave resonance peak shifts to a lower frequency and grows broader under progressively higher incident sound levels. Therefore, models with strain-dependent modulus and damping have been developed for sand particles based on a series of cyclic triaxial tests, while more recently, a velocity-dependent modulus was proposed to model the acoustically induced elastic softening for the hollow glass beads. To investigate the nature of this nonlinear behavior, the absorption coefficients of a 30 mm carbon particle stack were measured under four types of band-passed input signals (i.e., 500–1000 Hz, 500–2000 Hz, 500–4000 Hz, and 500–8000 Hz), each with 15 levels in steps of 1 dB. The input signals were designed to span the resonance frequency and to extend beyond to various degrees. The integrated sound pressure level, velocity, and displacement at the particle stack surface were calculated and were used to normalize the changing modulus and damping. It was found that only the total rms surface displacement can align the changing properties for all measurements.

9:45

4aPA6. Exploring conditions required for the subwavelength focusing of sound in the near field of an array of resonators using time reversal. Andrew Basham (Phys. & Astronomy, Brigham Young Univ., 1305 N Canyon Rd., Apt 304, Provo, UT 84604, andrewtbasham@gmail.com), Adam D. Kingsley, and Brian E. Anderson (Phys. & Astronomy, Brigham Young Univ., Provo, UT)

This presentation will discuss the time reversal focusing of audible sound over an array of Helmholtz resonators. The presence of the resonators modifies the phase speed of waves, thereby lowering the effective wavelength, in the near field of the focus. It is shown experimentally what conditions are necessary to obtain super resolution. Frequencies above the resonators' resonance frequency do not result in super resolution. In order to achieve super resolution in the plane of the resonators, the focused waves must be confined to propagate only in the plane of the resonators. When waves converged in three dimensions, super-resolution was not observed but when a 2-D waveguide was used, and super-resolution was observed.

10:00

4aPA7. Nonlinear acoustic phenomena in high amplitude time reversal focusing of airborne sound in a 1-D system. Michael M. Hogg (Phys. & Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, mikerhogg@gmail.com), Brian D. Patchett (Phys. & Astronomy, Brigham Young Univ., Orem, UT), Jay M. Clift, and Brian E. Anderson (Phys. & Astronomy, Brigham Young Univ., Provo, UT)

Time reversal (TR) is a process that can be used to generate high amplitude focusing of sound. It has been previously shown that using TR in reverberant environments to cause high amplitude foci will in fact have multiple nonlinear properties including waveform steepening and a nonlinear increase in peak pressure compressions. This study investigates the removal of one possible cause for these phenomena: free-space Mach stems. By isolating the focus into a 1-D system, the potential formation of Mach stems is eliminated so that remaining effects can be qualitatively observed. To achieve this, a system of pipes is used to restrict the focused waves to be planar in a 1-D reverberant environment. Preliminary results have shown that waveform steepening effects remain as expected but that the nonlinear increase in compression amplitudes disappears, which is consistent with the idea that free-space Mach stems cannot be formed in a 1-D system.

10:15–10:30 Break

10:30

4aPA8. COMSOL modeling of time reversal focusing in a periodic network of Helmholtz resonators. Adam D. Kingsley (Phys. & Astronomy, Brigham Young Univ., Provo, UT 84602, adamkingsley@gmail.com) and Brian E. Anderson (Phys. & Astronomy, Brigham Young Univ., Provo, UT)

This presentation will outline the physical and practical considerations for conducting a COMSOL simulation of time reversal focusing in air. Attempts to obtain sub-diffraction-limited time reversal focusing, along with physical and practical consideration, will be presented. Physical considerations include end-corrections, resonator shape and placement, computational domain and anechoic terminations. Practical considerations include mesh size and temporal length. While conducting a time domain study in COMSOL, mesh size is related to the highest frequency and the simulation length is related to the lowest frequency. To maximize computational resources, the simulation uses a restricted bandwidth. Extending the bandwidth is accomplished by combining multiple simulations. Comparison to previous super resolution studies as well as the authors' equivalent circuit model will also be discussed.

10:45

4aPA9. Comparison of high-amplitude focus creation methods with time reversal to help determine a mechanism of nonlinearity. Brian D. Patchett (Phys. & Astronomy, Brigham Young Univ., 800 W University Pkwy, MS-179, Orem, UT 84058, brian.d.patchett@gmail.com) and Brian E. Anderson (Phys. & Astronomy, Brigham Young Univ., Provo, UT)

Time reversal is a method often used to focus sound to a desired location within a reverberant environment. Recent high-amplitude focusing experiments have shown that when multiple sources are used simultaneously to generate a focus, the level reached (200 dB) is higher than if each source were focused individually with the signals being superposed computationally. An investigation into the temporal and spatial features and harmonic content of each method is presented in order to work toward an understanding of the mechanisms driving the nonlinear increases observed when the sources are combined acoustically as opposed to being summed in post-processing.

11:00

4aPA10. Complex frequency sink. Curtis Rasmussen (Univ. of Colorado Boulder, 1111 Eng. Dr. UCB 427, Boulder, CO 80309, curtis.rasmussen@colorado.edu), Matheus Rosa, Jacob Lewton, and Massimo Ruzzene (Univ. of Colorado Boulder, Boulder, CO)

As the time-reversed analog of the source, the sink is an abstraction of a device that can absorb incident symmetrical waves at a point. Such a device is useful for its ability to create an arbitrarily small focusing spot, surpassing the conventional diffraction limit. Sinks have been created using impedance matching and active cancellation; however, these techniques require the introduction of tailored loss or gain into the system. Here, we demonstrate a new method for creating a sink that requires no loss or gain but instead relies on complex frequency excitations, harmonic excitations that exponentially grow or decay in time. Recent work has shown how complex frequency excitations in Hermitian systems enable phenomena such as PT symmetry and perfect absorption that are typically only associated with non-Hermitian systems. We will discuss the theory of the complex frequency sink and analyze the case of a sink in a thin elastic plate. We will present experimental results where the scattering of a symmetric scatterer is minimized by complex frequency waves launched by piezoelectric patches. Laser-doppler vibrometer measurements of the plate shows the resonator's curious behavior as a loss and gain free sink.

11:15

4aPA11. Binary volume acoustic holograms. Michael Brown (Dept. of Medical Phys. and Biomedical Eng., Univ. College London, Malet Pl. Eng. Bldg., 3.23, Gower St., London WC1E 6BT, United Kingdom, michael.brown.13@ucl.ac.uk), Ben Cox, and Bradley Treeby (Dept. of Medical Phys. and Biomedical Eng., Univ. College London, London, United Kingdom)

Acoustic holograms or phase conjugate acoustic lenses are a simple inexpensive method for forming complex acoustic fields. They are 3D printable phase plates that can map an incident field onto a pre-defined phase distribution such that it diffracts to form a desired field. These phase plates are analogues of thin optical holograms with a thickness $d \cong \lambda$ the design wavelength. Another class of optical holograms are "thick" or "volume" holograms, composed of weak periodic variations in the refractive index, for which $d \gg \lambda$. These volume holograms are significantly more selective to the wavelength and direction of the incident field allowing for greater ability to multiplex patterns within a single hologram. In this work, we explore the generation of volume acoustic holograms using multi-material 3-D printing. First, we numerically assess the impact of sound speed contrast, absorption, and quantisation on efficiency. A greedy optimization approach is then introduced for the design of a volumetric hologram from a desired set of input/output fields. We then validate the approach experimentally using test samples printed on an Objet Connex 350. Several holograms are fabricated demonstrating both spatial and frequency multiplexing of different patterns in a single volume.

11:30

4aPA12. Impedance tube measurements of 3D printed porous materials. Noah Fitzpatrick (Phys. and Astronomy, Univ. of Central Arkansas, 201 Donaghey Ave., LSC 171, Conway, AR 72035, nfitxpatrick1@cub.uca.edu) and Carl Frederickson (Phys. and Astronomy, Univ. of Central Arkansas, Conway, AR)

The acoustics of porous media affects what materials are good for sound control in enclosed environments. The theories used in this area model porous materials as having continuous unconnected pores. The important physical factors in these theories are the porosity (ratio of pore volume to sample volume), tortuosity (ratio of pore length to sample depth), and flow resistivity (resistance to air flow through the sample). The models allow for calculation of acoustic properties but are not truly representative of most porous media. With the advent of 3D printing, it is possible to design and print 3D media with specific porosity and tortuosity characteristics. These materials can be used in an impedance tube to test the validity of the models being used. Porous plugs will be designed and 3D printed using stereolithography. The plugs' acoustic properties will be measured in an impedance tube. The results of this project will provide data that can be used to refine the acoustic models of porous media.

11:45

4aPA13. Impact of ground reflections on firearms noise exposure metrics. Jacob R. Smith (Dept. of Phys. and Astronomy, Brigham Young Univ., N281 Eyring Sci. Ctr., Provo, UT 84602, Jacobsmith1031@gmail.com), Avery K. Sorrell, Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), Reese D. Rasband (Res., U.S. Air Force, Provo, UT), Steven C. Campbell (Ball Aerosp., Beavercreek, OH), and Alan T. Wall (Res., U.S. Air Force, Wright Patterson Airforce Base, OH)

Outdoor measurements of weapons noise exposure are influenced by ground reflections. This paper analyzes the reflections' impact on A-weighted equivalent level (LAeq) for two measurement campaigns involving standing shooters and dozens of elevated microphones. Although it is only a marginally accurate model of a gunshot waveform, the modified Friedlander equation was used to model the direct and reflected muzzle blast waveforms over a perfectly reflective surface. The calculation of the reflection's effect on LAeq indicates a 2–3 dB increase relative to the free-field contribution over a relatively large spatial range. Validation of this model for most locations where the measured direct and reflected waveforms was well separated yields acceptable agreement for LAeq (<1 dB). Some challenges with the model development and comparison are discussed. This study contributes to improved weapons training safety and may eventually suggest revisions to the military noise exposure standard, MIL-STD-1474E. [Work supported by Air Force Research Laboratory through Ball Aerospace Technology Corporation.]

Session 4aPP

Psychological and Physiological Acoustics: Psychophysics, Methods, Models (Poster Session)

Monica L. Folkerts, Cochair
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Agudemu Borjigin, Cochair
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All posters will be on display from 9:00 a.m. to 12:00 noon. Authors of odd-numbered papers will be at their posters from 9:00 a.m. to 10:30 a.m. and authors of even-numbered papers will be at their posters from 10:30 a.m. to 12:00 noon.

Contributed Papers

4aPP1. Effects of ripple width on ripple density resolution. Alexander Supin (Inst. of Ecology and Evolution, 33 Leninsky Prospect, Moscow 119071, Russian Federation, alex_supin@mail.ru), Olga N. Milekhina, Dmitry I. Nechaev, and Marina Tomozova (Inst. of Ecology and Evolution, Moscow, Russian Federation)

Effects of ripple width in rippled-spectrum signals on ripple density resolution was investigated. Two measurement paradigms were tested: (i) ripple density resolution for discrimination between two rippled signals and (ii) discrimination between a rippled test signal and non-rippled reference signal. The ripple widths varied from 9% to 64% of the ripple frequency spacing. For both paradigms, the ripple density resolution increased with deceasing the ripple width. For discrimination between two rippled signals, the resolution was 8.1 ripples/oct for a ripple width of 64% and increased to 15.1 ripples/oct at the ripple width of 9%. For discrimination between a rippled test and non-rippled reference signal, the resolution was 9.3 ripples/oct at a ripple width of 64% and increased to 85 ripples/oct at a ripple width of 9%. Discrimination between two rippled signals is hypothesized to depend on ripple depth in the excitation pattern; the depth increases with narrowing the ripple width. Discrimination between a rippled test and non-rippled reference signal is hypothesized to depend on temporal processing; the effect of the ripple width appears due to increasing the ratio of the autocorrelated to uncorrelated components of the input signal with narrowing the ripples.

4aPP2. Effect of high-density additional rippled patterns on ripple density resolution thresholds. Marina Tomozova (Lab. of Vertebrate Sensory Systems, A.N. Severtsov Inst. of Ecology and Evolution of the Russian Acad. of Sci., Inst. of Ecology and Evolution, 33 Leninsky Prospect, Moscow 119071, Russian Federation, m.tomozova86@mail.ru), Alexander Supin (Inst. of Ecology and Evolution, Moscow, Russian Federation), Dmitry I. Nechaev (Inst. of Ecology and Evolution RAS, Moscow, Russian Federation), and Olga N. Milekhina (Lab. of Vertebrate Sensory Systems, A.N. Severtsov Inst. of Ecology and Evolution of the Russian Acad. of Sci., Moscow, Russian Federation)

This study aims to spot an effect of the density of an additional rippled signal on ripple density resolution thresholds when this density is greatly exceeding the density of the main signal. In such a case a spectrum is formed from two harmonic functions. Therefore, a phase-reversed spectral ripple resolution test was used in this research, which includes two experimental paradigms. In both 2 oct narrowband white noise passes through premade filters. A ripple density of the additional signal was varied from 10 to 50 ripples/oct. In a rippled test and reference stimuli the mean resolution was 9 ripples/oct. This fact supports the involvement of cochlear excitation patterns for the discrimination of such signals. In a test with the stimuli with an additional rippled signal and non-rippled reference signal the thresholds cannot be reduced to mean value. However, the ripple density resolution thresholds for 20, 30, and 50 ripple densities of additional signal gravitate towards the value of 11,25 ripples/oct. The ripple density resolution thresholds for 10 and 15 ripples/oct are higher than the other values what suggests a special combination of spectral and temporal mechanisms.

4aPP3. Adaptation effects along a voice/non-voice continuum. Zi Gao (Psych., Univ. of Minnesota, 75 E River Rd., Minneapolis, MN 55455, gao00196@umn.edu) and Andrew J. Oxenham (Psych., Univ. of Minnesota, Minneapolis, MN)

Adaptation, the selective decrease in neural responses following repeated stimulation, is likely important for perception in the face of variable sensory input. Previous researchers have observed contrastive adaptation effects involving speaker identity, gender, and vowels. The present study investigated whether such contrast effects can occur between voice and non-voice stimuli. A 10-step continuum between "voice" (/a/, /o/, or /u/ vowels) and "instrument" (bassoon, horn, or viola) was generated for each possible pair. In each trial, an adaptor, either voice or instrument, was played four times, followed by a test stimulus from along the appropriate continuum. When trials with voice and instrumental adaptors were grouped into separate blocks, strong contrastive adaptation effects were observed, with the test stimuli more likely to be identified as voice following instrumental adaptors and vice versa. Preliminary results comparing interleaved with blocked conditions and same-ear with different-ear conditions suggest longer-term build-up and persistence effects that may extend across both ears. [Work supported by NIH grant R01 DC012262.]

4aPP4. Design and manufacture of synthetic pinnæ for studying head-related transfer functions. Eric M. Sumner (Faculty of Industrial Eng., Mech. Eng., and Comput. Sci., Háskóli Íslands, Bjargargata 1, #318, Reykjavík 102, Iceland, ems36@hi.is), Morris Riedel, and Rúnar Unnþórsson (Faculty of Industrial Eng., Mech. Eng., and Comput. Sci., Háskóli Íslands, Reykjavík, Iceland)

The study of head-related transfer functions (HRTFs) is hampered by an inability to isolate the effects of individual anthropometric properties in an experimental setting. The authors propose to address this issue by studying the HRTFs of synthetic pinnæ which are designed to specification. The design, manufacture, and measurement of these synthetic pinnæ are detailed: A mold for each pinna is printed with a finite-deposition 3D printer and then cast in silicone with appropriate acoustic properties. These castings are then installed on a KEMAR mannequin whose HRTF is measured in an anechoic chamber. Additionally, comparative results are presented between

commercially produced pinnæ inserts and copies produced via the proposed method.

4aPP5. Toward an adaptive procedure for multi-frequency categorical loudness scaling: A Monte Carlo simulation study. Yi Shen (Speech and Hearing Sci., Univ. of Washington, 1417 NE 42nd St., Seattle, WA 98105-6246, shenyi@uw.edu), Yihui Zhang (Mathematics, Univ. of Washington, Seattle, WA), Winnie Shao (Northwestern Univ., Evanston, IL), and Stephen Neely (Boys Town National Res. Hospital, Omaha, NE)

To capture a listener's loudness perception profile, categorical loudness scaling (CLS) is typically repeated at various frequencies. The current study aims to develop psychophysical procedures that enable simultaneous estimation of loudness growth across frequencies. For these procedures, the listener hears a pure-tone stimulus provides a categorical rating ("Soft," "Loud," etc.) on each trial. After a response is collected, the procedures update a model of the loudness profile and leverage the model to optimize the stimulus (i.e., level and frequency) for the next trial. The modified slope-adaptive procedure selects the stimulus from a uniform distribution spanning the model-predicted dynamic range, while the modified maximum expected information (MEI) procedure optimizes the stimulus based on an entropy metric. Monte Carlo simulations were conducted to evaluate the two procedures using a database that consists of CLS data collected from 148 listeners at Boys Town National Research Hospital. For each listener, the two procedures were run based on responses simulated using their known loudness profiles (i.e., the ground truth). Both procedures were able to estimate the loudness profile close to the ground truth, with a root-mean-square error (RMSE) of about 6 dB after 100 trials. Below 100 trials, the modified MEI procedure showed a lower RMSE.

4aPP6. Multilevel probit regression for 3-alternative forced choice audibility testing. Matthew Boucher (Structural Acoust. Branch, NASA Langley Res. Ctr., Hampton, VA), Andrew Christian (Structural Acoust. Branch, NASA Langley Res. Ctr., 2 N. Dryden St., M/S 463, Hampton, VA 23681, andrew.christian@nasa.gov), Menachem Rafaelof (National Inst. of Aerosp., Hampton, VA), Kevin Shepherd (Distinguished Res. Associate, NASA Langley Res. Ctr., Hampton, VA), Stephen A. Rizzi (Aeroacoustics Branch, NASA Langley Res. Ctr., Hampton, VA), and James Stephenson (Aviation & Missile Ctr., U.S. Army Combat Capabilities Development Ctr., Hampton, VA)

A recent psychoacoustic test at NASA Langley generated a dataset of 3-alternative forced-choice responses for 40 subjects that measured the audibility of a tone complex in a shaped broadband masker. The task was completed by four subjects at a time in a small theatre-like environment using predetermined stimuli levels. These data were subject to 4 forms of probit regression: a "complete pooling" analysis in which all data from the test was fit with one curve, two forms of "no pooling" analyses in which subjects' data were treated individually (using both packaged and custom software), and a "partial pooling" analysis in which multilevel-regression software fit both individual curves as well as population-level parameters at the same time. The results of the analyses are compared in terms of both individual- and population-level parameters. Partial pooling appears to give the most consistent results at both levels, as well provide the most robustness among the packaged approaches (albeit with added complexity over single-level regression). These results mirror those contained in a recent NASA technical memorandum entitled "Comparisons of Analysis Methods Applied to Alternative Forced Choice Audibility Data."

4aPP7. The impact of temporally coherent visual and vibrotactile cues on speech perception in noise performance. Yonghee Oh (Speech, Lang., and Hearing Sci., Univ. of Florida, 1225 Ctr. Dr., Rm. 2128, Gainesville, FL 32610, yoh@phhp.ufl.edu), Nicole Kalpin, and Jessica Hunter (Speech, Lang., and Hearing Sci., Univ. of Florida, Gainesville, FL)

The inputs delivered to different sensory organs provide us with complementary information about the environment. Our recent study demonstrated that presenting abstract visual information of speech envelopes substantially improves speech perception ability in normal-hearing (NH), listeners [Yuan *et al.*, *J. Acoust. Soc. Am.* (2020)]. The purpose of this study was to expand this audiovisual speech perception to the tactile domain. Twenty adults participated in sentence recognition threshold measurements in four different sensory modalities (AO: auditory-only; AV: auditory-visual; AT: audio-tactile; and AVT: audio-visual-tactile). The target sentence [CRM speech corpus, Bolia *et al.*, *J. Acoust. Soc. Am.* (2000)] level was fixed at 60 dBA, and the masker (speech-shaped noise) levels were adaptively varied to find masked thresholds. The amplitudes of both visual and vibrotactile stimuli were temporally synchronized and non-synchronized with the target speech envelope for comparison. Results show that temporally coherent multimodal stimulation (AV, AT, and AVT) significantly improves speech perception ability when compared to audio-only (AO) stimulation. These multisensory speech perception benefits were reduced when the cross-modal temporal coherence characteristics were eliminated. These findings suggest that multisensory interactions are fundamentally important for speech perception ability in NH listeners. The outcome of this multisensory speech processing highly depends on temporal coherence characteristics between multi-modal sensory inputs.

4aPP8. Temporal binding window between three different sensory modalities: Auditory, visual, and tactile. Yonghee Oh (Speech, Lang., and Hearing Sci., Univ. of Florida, 1225 Ctr. Dr., Rm. 2128, Gainesville, FL 32610, yoh@phhp.ufl.edu), Kayla Borges, Kelli Meyers, Altemis Lopez, Shelbey Spratlin, and Elizabeth Fisch (Speech, Lang., and Hearing Sci., Univ. of Florida, Gainesville, FL)

Multisensory integration is crucial to perceptual understanding, but cues from multiple modalities do not necessarily arrive at the brain simultaneously. The window of time in which sensory inputs can be interpreted as simultaneous is termed the "temporal binding window" (TBW). Evidence suggests unimodal temporal discrimination is altered in a series of neurodevelopmental disorders but its relationship to perceptual timing of multisensory events (auditory, visual, and tactile) is not known, even in healthy populations. The purpose of this study was to characterize temporal integration of sensory modalities within a young healthy population. Twenty-five adults participated in TBW measurements with three different modalities. 100-ms stimuli (auditory: beep; visual: flash; tactile: vibration) were presented with various onset asynchrony (-500 to 500 ms) between two stimuli (AV: auditory-visual; AT: auditory-tactile; VT: visual-tactile). Participants were instructed to select "simultaneous" or "asynchronous" responses. Results show that the TBW for the AV condition ranged from 222 to 689 ms (mean $\pm$ SD = 418 $\pm$ 103), the AT condition ranged from 164 to 666 ms (376 $\pm$ 130), and the VT condition ranged from 70 to 751 ms (279 $\pm$ 131). These findings suggest that deficits in multisensory temporal function may factor into perceptual and cognitive weaknesses that characterize clinical disorders, raising the possibility that it may be a contributing pathologic factor in these conditions.

4aPP9. Examining potential sources of variability in behavioral measures of cochlear gain and gain reduction in listeners with normal hearing or minimal cochlear hearing loss. Elizabeth A. Strickland (Speech, Lang., and Hearing Sci., Purdue Univ., 715 Clinic Dr., West Lafayette, IN 47907, estrick@purdue.edu), Anna Hopkins, Andrea Rayner, William B. Salloom, Miranda Skaggs, Nicole Mielnicki, Hayley Morris, and Alexis Holt (Speech, Lang., and Hearing Sci., Purdue Univ., West Lafayette, IN)

This study examined the relationship between clinical measures of auditory function and psychoacoustic measures related to cochlear function. Listeners' audiometric thresholds for long tones ranged from well within the clinically normal range to just above this range. Where thresholds were elevated, other clinical tests were consistent with a cochlear origin. Because the medial olivocochlear reflex decreases cochlear gain in response to sound, measures were made with short stimuli. Signal frequencies were from 1 to 8 kHz. One point on the lower leg of the input/output function was measured by finding threshold masker level for a masker almost one octave below the signal frequency needed to mask a signal at 5 dB SL. Gain reduction was estimated by presenting a pink broadband noise precursor

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before the signal and masker and measuring the change in signal threshold as a function of precursor level. In a previous presentation, it was shown that the estimate of gain reduction decreased as quiet threshold increased, but was not solely determined by the amount of gain. In this presentation, the relationship between gain reduction and clinical otoacoustic emission and middle-ear muscle reflex measurements was examined to determine whether these explained some of the variability. [Work supported by NIH (NIDCD) R01 DC008327 (EAS) and NIH(NIDCD) T32 DC016853 (WBS).]

4aPP10. Discrimination of spectrotemporal ripple sweep direction near detection threshold. Christopher Conroy (Speech, Lang. & Hearing Sci., Boston Univ., 635 Commonwealth Ave., Boston, MA 02215, cwconroy@bu.edu), Andrew J. Byrne, and Gerald Kidd (Speech, Lang. & Hearing Sci., Boston Univ., Boston, MA)

There are single neurons in the auditory systems of some nonhuman animals that are sensitive to spectrotemporal ripples sweeping in one direction versus the other. We used a "2×2 forced-choice" simultaneous detection/discrimination paradigm [Nachmias and Weber, *Vision Res.* **15** 217–223 (1975)] to search for behavioral evidence of similar neurons in the human auditory system (i.e., "labeled lines" for discriminating spectrotemporal ripple sweep direction). On each two-interval trial, observers were presented two successive sounds (one in each interval). One sound was an unmodulated noise and the other sound was a noise modulated by an upward-sweeping or downward-sweeping spectrotemporal ripple (± 8 Hz/1 cycle/octave). Subsequently, observers registered two separate responses: a detection response (Which interval contained the ripple?) and a discrimination response (Was the ripple sweeping upward or downward?). Psychometric functions (proportion correct versus ripple-modulation depth) were constructed and compared for the two response categories. Labeled line theory predicts that the two functions should be identical: a correct detection response should always yield a correct discrimination response. Contrary to this prediction, observers required greater modulation depths to support discrimination than detection. This finding suggests that the detection of spectrotemporal ripples and the discrimination of spectrotemporal ripple sweep direction may depend on different mechanisms.

4aPP11. Talker adaptation or "talker" adaptation? Musical instrument variability impedes pitch perception. Anya E. Shorey (Psychol. and Brain Sci., Univ. of Louisville, 317 Life Sci. Bldg., Louisville, KY 40292, anya.shorey@louisville.edu), Caleb J. King (Psychol. and Brain Sci., Univ. of Louisville, Louisville, KY), Rachel M. Theodore (Univ. of Connecticut, Storrs, CT), and Christian E. Stilp (Psychol. and Brain Sci., Univ. of Louisville, Louisville, KY)

When perceiving speech spoken by a single talker versus multiple talkers, listeners show perceptual benefits such as increased accuracy and/or decreased response time. There are several theoretical explanations for this talker adaptation phenomenon; one way to distinguish among these is to test whether adapting to stimulus structure is speech-specific or general to auditory perception. Music, like speech, is a sound class replete with acoustic variation. Here, participants completed a musical task that mirrored talker adaptation paradigms. On each trial, participants heard a tone and reported whether it was the lower (D4, 294 Hz) or higher pitched tone ($F^{\#}4$, 370 Hz). Tones were presented in a single- or mixed-instrument block. We predicted that perceptual benefits from structure in the acoustic signal are not specific to speech but are a general auditory response. Accordingly, we hypothesized participants would respond faster in the single-instrument block, consistent with speech studies that used a similar paradigm. Pitch judgments were faster (and more accurate) in the single instrument block, parallel to results from talker adaptation studies. In agreement with general theoretical approaches to auditory perception, perceptual benefits from signal structure are not limited to speech.

4aPP12. Portable automated rapid testing in Spanish: Validation and norms of central hearing processes on a Mexican population. E. S. Lelo de Larrea-Mancera (Cognit. and Clinical Neuropsychology, National Inst. of Neurology and Neurosurgery Manuel Velasco Suárez, Av. Insurgentes Sur No. 3877 ext. 2028, Colonia La Fama, Mexico City, Cdmx 14269, Mexico, elelo001@ucr.edu), Rodolfo Solís-Vivanco (Cognit. and Clinical Neuropsychology, National Inst. of Neurology and Neurosurgery Manuel Velasco Suárez, Cdmx, Cdmx, Mexico), Yolanda Sánchez Jiménez (Otoneurology Clinic, National Inst. of Neurology and Neurosurgery Manuel Velasco Suárez, Mexico City, Cdmx, Mexico), Laura Coco, Frederick J. Gallun (Oregon Hearing Res. Ctr., Oregon Health and Sci. Univ., Portland, OR), and Aaron Seitz (Psych., Univ. of California, Riverside, Riverside, CA)

Portable automated rapid testing of central auditory processing measures implemented in the app PART (https://braingamecenter.ucr.edu/games/p-a-r-t/) were adapted for Spanish-speaking populations. This adaptation included the generation of a corpus equivalent to the CRM to test a speech-on-speech masking task, in addition to frequency modulation tasks, temporal gap, temporal, spectral and spectro-temporal modulations, and pure tone detection in noise. This battery of tests was selected for its potential clinical utility and its previous validation across several settings including remote testing with participant-owned headphones, tablets or smartphones. Here, we tested 96 young adults in Mexico City (ages 18–45) with normal hearing. As expected, distributions of scores were similar to previously reported norms for the English-language version of the battery in English-speaking populations. Scores for the newly developed Spanish-language speech-on-speech masking tests were also similar to data previously reported in English-speaking populations. Furthermore, there was no significant effects of age or subjective hearing handicap (HHIA) in any of the tests. This study can be now used as a normative dataset, and the adapted battery can be used as an accsessible tool (available freely online) for valid and reliable testing of central auditory processing measures in Spanish-language populations.

4aPP13. Predicting reflection patterns from binaural activity maps using deep neural networks. Jeramey Tyler, Mei Si (Cognit. Sci., Rensselaer Polytechnic Inst., Troy, NY), and Jonas Braasch (School of Architecture, Rensselaer Polytechnic Inst., School of Architecture, 110 8th St., Troy, NY 12180, braasj@rpi.edu)

A new model architecture is presented to predict room acoustical parameters from a running binaural signal. For this purpose, a deep neural network architecture is combined with a precedence effect model to extract the spatial and temporal locations of the direct signal and early reflections. The precedence effect model builds on the modified BICAM algorithm [Braasch, *J. Acoust. Soc. Am.* **140**, EL143], for which the 1st layer auto-/cross correlation functions are replaced with a Cepstrum method. The latter allows a better separation of features relating to the source signal's early reflections and harmonic structure. The precedence effect model is used to create binaural activity maps that are analyzed by the neural network for pattern recognition. Anechoic orchestral recordings were reverberated by adding four early reflections and late reverberation to test the model. Head-related transfer functions were used to spatialize direct sound and early reflections. The model can identify the main reflection characteristics of a room, offering applications in numerous fields, including room acoustical assessment, acoustical analysis for virtual-reality applications, and modeling of human perception. [Work supported by the National Science Foundation under Grant No. IIS-1909229.]

4aPP14. Extensions of a neural-fluctuation perspective on simultaneous notched-noise masking. Braden Maxwell (Music Theory and Biomedical Eng., Univ. of Rochester, 601 Elmwood Ave., Box 603, Rochester, NY 14642, bmaxwel2@u.rochester.edu), Mohammed Abumuaileq, and Laurel H. Carney (Biomedical Eng., Univ. of Rochester, Rochester, NY)

A previous paper [Maxwell *et al.*, *J. Acoust. Soc. Am.* **3523**, 3537 (2020)] suggested that listeners may identify the target interval in a simultaneous notched-noise task using neural fluctuations, which change with the addition of a tone to the masker. Neural fluctuations are changes in the instantaneous firing rate of auditory-nerve fibers, generally at a longer (slower) time scale than temporal fine structure, that are shaped by

peripheral nonlinearities. Differences in neural-fluctuation amplitudes across the peripheral tonotopic axis are proposed to lead to differences in rates across the population of auditory midbrain neurons, due to the sensitivity of these neurons to amplitude modulation. Here, using similar methods to Maxwell *et al*. (2020) and stimuli matching those of Baker and Rosen (*J. Ascoust. Soc. Am.* **454**, 462 (2006)], we extend previous work by demonstrating that neural fluctuations may account for additional features of simultaneous notched-noise masking results. These features include (1) increase in filter bandwidth at higher stimulus frequencies, (2) higher bandwidth with increasing masker level, and (3) filter asymmetry. These results, in conjunction with the previous conclusion that auditory-nerve excitation patterns cannot explain notched-noise thresholds in all cases, support the interpretation that psychophysical auditory filters may be based on the output of the subcortical auditory system. [Work supported by NIDCD-R01-010813.]

4aPP15. The effect of competitor rhythmic structure when recalling vowels in a complex listening environment. Dylan V. Pearson (Speech and Hearing Sci., Indiana Univ.-Bloomington, 1625 Pinestone Ct., Bloomington, IN 47401, dylpears@iu.edu), Yi Shen (Speech and Hearing Sci., Univ. of Washington, Seattle, WA), Gary R. Kidd (Speech and Hearing Sci., Indiana Univ.-Bloomington, Bloomington, IN), and J Devin McAuley (Dept. of Psych., Michigan State Univ., East Lansing, MI)

Research has shown that the rhythmic structure of speech is relevant to listening outcomes in complex environments. Several studies have shown that an intact, natural target rhythm facilitates listeners' perception of auditory signals. However, less is known about the effect of competitor signal rhythmic structure on target recognition. The current study uses a synthetic vowel identification paradigm to investigate the influence of the rhythmic structure of to-be-ignored competing signals on listeners' ability to identify speech information in a target signal. Listeners were tasked with identifying the last three vowels of target sequences of varying length presented alongside competing sequences consisting of either vowel-like harmonic tone complexes or similar vowels. The inter-onset-intervals in these sequences were manipulated to create conditions where the internal rhythm of the sequences was either isochronous or irregular. Competitor rhythm was found to be a significant factor for target recognition only with the vowel-sequence competitors. Isochronous target vowel recognition was better for vowel-sequence competitors with an irregular rhythm than with a regular (isochronous) rhythm. The results of this study highlight (1) the relevance of the rhythmic structure in selective listening tasks and (2) the importance of target and competitor similarity in determining the influence of competitor rhythm.

4aPP16. Accurately targeting an arbitrary probability of response in staircase procedures. Eric C. Hoover (Dept. of Hearing and Speech Sci., Univ. of Maryland, 0100 LeFrak Hall, 7251 Preinkert Dr., College Park, MD 20742, ehoover@umd.edu)

Staircase procedures are commonly used to identify the stimulus level corresponding to a specific probability of response. The primary benefit of staircase procedures is that they require only that the probability of response changes monotonically with the stimulus level. The probability of response targeted by the procedure is typically estimated from the mean of reversals. However, the mean of reversals can be asymptotically biased away from the targeted probability. Existing methods to address the bias add assumptions about task performance, thereby limiting their use to well understood tasks and listener groups. We evaluated the hypothesis that the bias could be mitigated without adding assumptions by accounting for the effect of invariant procedural parameters on the estimate of the target probability. The approach was successful for staircases with different ascending and descending steps, for which the bias was found to depend on step size only. For other staircases, predicting bias required adding assumptions comparable to existing solutions. Results suggest that the bias is caused by the fact that reversals are influenced by parameters that do not affect the target point. A method is proposed to accurately estimate an arbitrary probability of response without adding assumptions about the task or listeners. [R01DC015051.]

4aPP17. Psychoacoustic thresholds measured using the Portable Automated Rapid Testing (PART) iPad application in a large cohort of older listeners. Nirmal Kumar Srinivasan (Dept. of Speech-Lang. Pathol. and Audiol., Towson Univ., 8000 York Rd., Towson, MD 21252, nsrinivasan@towson.edu), Janet Kim, Kayla Coleman, Kelly Brown, Saradha Ananthakrishnan (Dept. of Speech-Lang. Pathol. and Audiol., Towson Univ., Towson, MD), Eric C. Hoover (Dept. of Hearing and Speech Sci., Univ. of Maryland, College Park, MD), and Frederick J. Gallun (Oregon Hearing Res. Ctr., Oregon Health and Sci. Univ., Portland, OR)

Traditional methods used to assess the auditory processing capabilities of individuals have limited clinical applications as they are time consuming and require specialized hardware. Portable Automated Rapid Testing (PART) is a free application that allows various clinical assessments to be conducted in an automated and rapid fashion using commercially available consumer headphones. Use of this platform in an uncontrolled listening environment has been validated in young listeners with normal hearing, yielding thresholds similar to those reported in laboratory studies. Here, we report auditory processing capabilities measured using PART in a large cohort of older listeners with varying hearing capabilities. Psychoacoustic thresholds for gap detection, spectrotemporal modulation detection, dichotic and diotic frequency modulation detection, and spatial release from masking were obtained from 70 older listeners (age range: 40—75 years; 3-frequency PTA: 8.33—48.33 dB HL). Initial analyses revealed that the thresholds obtained using PART were similar to thresholds obtained in a laboratory environment. Results demonstrate that PART can be used with older listeners with minimal instruction and validate the use of a consumer platform and rapid techniques for the assessment of auditory processing capabilities in older adults with a range of hearing ability.

4aPP18. Effect of stimulus complexity on psychometric functions. Nirmal Kumar Srinivasan (Dept. of Speech-Lang. Pathol. and Audiol., Towson Univ., 8000 York Rd., Towson, MD 21252, nsrinivasan@towson.edu), Radhika Kansangra, and Angelica Trotman (Dept. of Speech-Lang. Pathol. and Audiol., Towson Univ., Towson, MD)

Factors such as semantic and syntactic context, predictability, talker characteristics, background noise characteristics, and spatial differences between the target and maskers play major role in identifying speech in complex listening environments. Here, we present psychometric properties of two closed set speech corpus: (1) co-ordinate response measure (CRM) and (2) Boston University Corpus (BUC) obtained from a group of fifty young normal-hearing, listeners. CRM sentences are of the form "Ready CALLSIGN go to a COLOR NUMBER now" and BUC sentences are of the form "NAME VERB NUMBER ADJECTIVE NOUN." Method of constant stimuli was used to obtain psychometric functions for the two corpora in the presence of speech and speech shaped noise. One half of the participants were presented the stimuli at a constant level (20 dB SL), while it was presented at comfortable listening levels for the half. The effect of the stimulus complexity, presentation level, and spatial configuration of the target and noise on the identification thresholds and the shape of the psychometric functions will be discussed.

4aPP19. Effect of stimulus complexity on the phonemic restoration effect. Nirmal Kumar Srinivasan (Dept. of Speech-Lang. Pathol. and Audiol., Towson Univ., 8000 York Rd., Towson, MD 21252, nsrinivasan@towson.edu), Sadie O'Neill, and Chhayakanta Patro (Dept. of Speech-Lang. Pathol. and Audiol., Towson Univ., Towson, MD)

The phonemic restoration effect is the increase in speech intelligibility when the silent portions of the interrupted speech were replaced by a noise that is louder than the speech signal itself. Previous research has demonstrated that the phonemic restoration effect is maximum when the interruption rate is between 1.5 and 3 Hz with a 50% duty cycle. Here we present phonemic restoration results obtained using five different speech corpora varying in stimulus complexity (least complex – most complex): (1) Callsign Acquisition Test (CAT); (2) Co-ordinate Response Measure (CRM); (3) Boston University Corpus (BUC); (4) IEEE sentences; and (5) R-PRESTO obtained from a group of young normal-hearing, listeners. Interruption rates of 2 Hz and 3 Hz were used and speech shaped noise was used

to fill the silent interruptions. Initial analyses of the data revealed that the phonemic restoration effect was absent for the least complex stimuli (CAT) because of ceiling effects and the amount of restoration increased as the complexity of the stimulus increased. The effect of interruption rate and signal-to-noise ratio between the speech and noise on the phonemic restoration effect will also be discussed.

4aPP20. Quantifying impacts of hearing protection devices on sound localization in azimuth and elevation: Toward predictors of performance. Andrew D. Brown (Speech and Hearing Sci., Univ. of Washington, 1417 NE 42nd St., Seattle, WA 98105, andrewdb@uw.edu), Nathaniel Greene (Otolaryngol., Univ. of Colorado School of Medicine, Aurora, CO), David J. Audet (Speech and Hearing Sci., Univ. of Washington, Seattle, WA), Caylin R. McCallick (Otolaryngol., Univ. of Colorado School of Medicine, Aurora, CO), Carol A. Sammeth (Physiol. & Biophys., Univ. of Colorado School of Medicine, Aurora, CO), David A. Anderson (Appl. Res. Assoc., Inc., Denver, CO), Gregory T. Rule, and Theodore F. Argo (Appl. Res. Assoc., Inc., Littleton, CO)

Modern hearing protection devices (HPDs) effectively mitigate the risk of noise-induced hearing loss when used as intended, and some even preserve the audibility of low-to-moderate-intensity sounds. Nonetheless, negative auditory perceptual side-effects continue to limit usability in critical settings. Most notably, dozens of studies have shown that HPDs lead to significant errors in sound source localization. Several studies have specifically linked such errors to HPD-induced distortions of the spectra of transmitted signals. Here we provide an update on a multi-site study designed to capture patterns in sound localization errors (among other impacts) across different classes of HPDs, toward validation of acoustic predictors of HPD performance. At two independent study sites, human listeners localized brief broadband signals with open ears (control) and during use of passive and active earplug and earmuff-style devices. Sources spanning 360° in azimuth and −30° to +60° in elevation enabled measurement, via wireless head-tracking, of 2-D localization error across a broad range of source locations. New metrics integrating these measurements with acoustic data, obtained via standardized HPD test fixtures as well as in-ear microphones designed to capture individual variation in HPD impacts, will support application-specific selection of existing HPDs and inform the design of new HPDs.

4aPP21. Measuring harmonic benefit in musicians and non-musicians in several tasks. Daniel R. Guest (Univ. of Minnesota, Minneapolis, MN), Neha Rajappa (Univ. of Minnesota, 75 E River Pkwy, Minneapolis, MN 55455, rajap013@umn.edu), and Andrew J. Oxenham (Univ. of Minnesota, Minneapolis, MN)

Prior work has demonstrated that harmonic tones are easier to detect in noise and yield better F0 discrimination in noise than inharmonic tones. These effects, referred to as harmonic benefit, appear to be approximately the same size in musicians and non-musicians, despite musicians' overall better pitch discrimination. The present study aimed to replicate these findings and extend them to include measurements of harmonic benefit in other tasks. Non-musicians and musicians were compared in four tasks: detection in noise, F0 discrimination, FM detection, and AM detection. The stimuli in each task were either harmonic or inharmonic complex tones and were presented in threshold-equalizing noise at a range of signal-to-noise ratios. We found that harmonic benefit for F0 discrimination was large and remained large even after accounting for differences in the detectability of harmonic and inahrmonic tones. In contrast, harmonic benefit was small for FM and AM detection and could mostly be accounted for by differences in detectability. In contrast to prior studies, we found that musicians showed a larger harmonic benefit than non-musicians. These results provide insight into how musical training may specialize the auditory system for processing of harmonicity and pitch. [Work supported by NIH grants F31 DC019247 and R01 DC005216.]

4aPP22. Profile analysis and ripple discrimination at high frequencies. Daniel R. Guest (Univ. of Minnesota, 75 E River Rd., Minneapolis, MN 55455, guest121@umn.edu) and Andrew J. Oxenham (Univ. of Minnesota, Minneapolis, MN)

Profile analysis and spectrotemporal ripple discrimination are psychoacoustic tasks used to probe the auditory system's spectral and intensity resolution. In the present experiment, we compared performance at low- and high-frequencies in four related psychoacoustic tasks: level discrimination, profile analysis, spectrotemporal ripple detection, and spectrotemporal ripple direction discrimination. The level discrimination and ripple detection tasks were designed so that cues from single auditory filters were sufficient for performing the tasks. The profile analysis and ripple direction discrimination tasks were designed to render cues from single auditory filters insufficient to perform the tasks. Based on data from a group of ∼20 young normal-hearing, listeners, we found that profile analysis was markedly worse at high-frequencies than low-frequencies, even when accounting for differences in level discrimination at low- and high-frequencies. In contrast, no significant differences were observed between low- and high-frequencies for either the ripple detection or ripple direction discrimination tasks. We further analyzed our behavioral data using computational simulations of auditory-nerve and midbrain responses. This analysis suggested that differences in performance at low- and high-frequencies cannot be explained at the level of the auditory periphery, but instead emerge at more central loci.
[Work supported by NIH grants F31 DC019247 and R01 DC005216.]

4aPP23. Extended high-frequency pure-tone thresholds predict speech-in-speech recognition even when extended high-frequency speech cues are absent. Rohit Ananthanarayana (Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, 901 S Sixth St., Champaign, IL 61820, rohitma2@illinois.edu), Allison Trine, and Brian B. Monson (Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, Champaign, IL)

Recent work has demonstrated that extended high-frequency (EHF; >8 kHz) hearing is valuable for speech-in-noise recognition. These findings contradict the broadly accepted "speech bandwidth" that has historically been limited to below 8 kHz. Several studies also indicate that EHF pure-tone thresholds predict speech-in-noise performance. One question that has arisen is whether the association that has been observed between EHF pure-tone thresholds and speech-in-noise recognition is causal—that loss of audibility of EHF cues in speech degrades speech-in-noise recognition. Indeed, this effect has been demonstrated using low-pass filtering, but whether elevated EHF thresholds would produce a similar effect is not certain. Another possibility is that EHF thresholds are a marker for subclinical dysfunction at lower frequencies that degrades speech recognition. These two possibilities are not mutually exclusive (nor exhaustive) and each could contribute to the observed relationship. Here we present a reanalysis of previous data collected in our lab, the results of which suggest that 16-kHz pure-tone thresholds are consistent predictors of speech-in-speech recognition, regardless of whether EHF cues are present in the speech signal. These findings suggest elevated EHF thresholds may indicate subclinical auditory dysfunction impairing speech-in-speech recognition.

Session 4aSAa

Structural Acoustics and Vibration: Absorption, Isolation, and Noise Mitigation

Colby W. Cushing, Cochair
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Thomas Bowling, Cochair
Naval Surface Warfare Center Carderock Division, Bethesda, MD 20817

Contributed Papers

8:30

4aSAa1. Aerogel-based ultralight noise absorbers fabricated using 3D freeze printing. Guang Yang (Dept. of Industrial and Manufacturing Systems Eng., Kansas State Univ., Manhattan, KS), Bhisham Sharma (Aerosp. Eng., Wichita State Univ., 1845 Fairmount St., Wichita, KS 67260, bhisham.sharma@wichita.edu), Dong Lin, and Shuting Lei (Dept. of Industrial and Manufacturing Systems Eng., Kansas State Univ., Manhattan, KS)

Aerogels are a group of synthetic materials with a structure comprising up to 99.8% air. This ultra-sparse skeletal structure results in extremely low-density and thermal conductivity. Because of these extreme properties, aerogels are highly attractive for thermal insulation and energy absorption applications. In acoustics, their application is typically limited to their use as granular material—either as granular layers or as additives to conventional fibrous and matrix materials. Here, we explore the possibility of controlling not only their mesostructure but also their macrostructure by fabricating aerogel-based ultralight noise absorbers using a 3-D Freeze Printing (3DFP) technique. First, we introduce the novel 3DFP method, which combines the advantages of additive manufacturing and freeze casting processes. Next, we focus on the acoustic characterization of 3-D freeze printed aerogels using a normal incidence impedance tube setup. Our results show that compared with current passive noise reduction materials, the unique structures generated by the 3DFP method can provide significantly improved absorption and transmission loss properties.

8:45

4aSAa2. Sound absorption properties of spinodoid structures with spatially varying properties. Bhisham Sharma (Aerosp. Eng., Wichita State Univ., Wichita, KS) and Brittany Wojciechowski (Aerosp. Eng., Wichita State Univ., 1845 Fairmount St., Wichita, KS 67260, brwojciechowski@shockers.wichita.edu)

Creating lightweight structures with tailored noise reduction at low-frequencies remains a major engineering challenge. Open-celled porous structures with low relative densities offer a possible solution to this problem. Spinodoids—structures with microstructural architectures based on spinodal decomposition—have been recently shown to provide robust and tunable mechanical properties. Here, we study the sound absorption properties of spinodoids with spatially varying properties such as relative density, spinodoid type, and wavenumbers. We generate the sample geometries using MATLAB and print them using low-cost additive manufacturing methods. The printed samples were then tested using a normal impedance tube setup to characterize their absorption coefficients. Our results show that spatially gradient spinodoids provide an attractive solution for designing multifunctional structures with application-specific acoustic properties.

9:00

4aSAa3. Tunable performance metrics for acoustic liners. William Kresl (Oklahoma State Univ., 201 General Academic Bldg., Stillwater, OK 74078, will.kresl@okstate.edu) and James M. Manimala (Oklahoma State Univ., Stillwater, OK)

In the development of automated design tools for acoustic liners customized to mitigate specific noise spectral characteristics, it is often beneficial to have a unified quantitative metric to discriminate between candidate designs. Such metrics are especially useful when optimizing the packing of 3D folded cavities into a prescribed liner volume. We devise tunable performance metrics (TPM) that account for absorption parameters such as the peak absorption and the frequency at which it occurs, the bandwidth of appreciable absorption and its lower bound among others. Metrics are tuned by varying the weight functions for these absorption parameters in order to address the specific priorities for the liner's design. A numerical study is conducted using a procedure based on the Zwikker–Kosten Transmission Line (ZKTL) theory to demonstrate the application of TPMs for the selection of acoustic liner designs for various scenarios. When liners with complex core geometries are used to enhance acoustic performance, TPMs that also include structural parameters related to mass, volume, stiffness or strength could be utilized to provide a more comprehensive means of evaluation.

9:15

4aSAa4. Inversion of bending wave properties of viscoelastic panels from laser Doppler vibrometer measurements. Jack Taylor (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St., Greene Bldg., Troy, NY 12180, tayloj13@rpi.edu), Max Miller, and Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY)

Thin panels of viscoelastic materials have been shown to have tremendous sound transmission loss enhancement characteristics when implemented in sandwiched gypsum wall panels. However, it is challenging to obtain reliable dynamic mechanical properties of these materials across relevant building acoustics frequencies. To better understand the dynamic mechanical properties of these materials, a methodology employing bending wave characterization is discussed. Relying on inversion processes from the measured panel velocities, this methodology yields loss factors, complex elastic moduli and phase speeds. The dispersive nature of these bending wave phenomena in viscoelastic panels opens a unique set of challenges for reliable dynamic mechanical property characterization. This paper discusses these challenges as wells as an approach to mitigate these effects and improve measurement reliability.

9:30

4aSAa5. Assessment of building isolation performance against train induced vibrations by *in situ* measurement. Hamid Masoumi (CDM Stravitec, Overijse, Belgium) and Bradlay Hunt (CDM Stravitec, 342 N. Queen St., Warehouse D, Lancaster, PA 17579, b.hunt@cdm-stravitec.com)

A review of an experimental assessment of the acoustic performance of an isolated two-story building constructed adjacent to a train station. The building was isolated using reinforced rubber isolation bearings installed on top of the foundation footings at the ground level. The acoustic performance of the isolation system has been investigated by means of an *in situ* measurement campaign at different construction phases. The results of the measurement campaign have been used to help define the Building Base Isolation Indicator implemented within the frame of the "Building Insulation against Outdoor Vibrations" (BIOVIB) project.

9:45

4aSAa6. Enhancing wear prediction with spectral analysis of noise and the dynamic response of the structure. Iyabo G. Lawal (Mech. Eng., The Univ. of Texas at Austin, 204 E. Dean Keeton St., Stop C2200, Austin, TX 78712, iyabo@utexas.edu), Matthew R. Brake (Mech. Eng., Rice Univ., Houston, TX), and Michael R. Haberman (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

High frequency noise has been observed during reciprocating sliding of metal-metal in dry contact. In tribology, the study of friction and wear, this noise was historically associated with "brake squeal;" however, it has also been found to occur within structures and systems that experience high frequency, reciprocating contact of metal constituents; typically found in the connecting locations such as the joints of a variety of jointed structures ranging from automobiles, airplanes, and HVAC systems. This type of tribological behavior and fretting fatigue has been replicated in tribology experiments where researchers found increased noise followed with the increase in sliding velocity and lowered relative humidity [1]. It was also found in these experiments, that noise is only produced during the tension phase of the fretting cycle and it only occurs when specific criteria are met in the fretting system: reduction in the coefficient of friction and the self-excited vibration of the structure. In this study, 304 SS steel samples are tested within a fretting rig apparatus that produces high frequency noise. In addition to spectral analysis of the noise produced, the dynamic response of the self-excited response of the system is developed. The combination of spectral data and structural dynamics data may allow for better correlation between noise and the prediction of wear in dry metal-to-metal contacts. T. Jibiki and M. Tamura, "A basic study of friction noise caused by fretting," *Wear* **251**, 1492–1503 (2001).

10:00

4aSAa7. Simulating acoustic waves in acoustic black hole analogues. Sören Schenke (Otto-von-Guericke-Univ. Magdeburg, Universitätsplatz 2, Magdeburg 39106, Germany, soeren.schenke@ovgu.de), Fabian Sewerin, Berend van Wachem, and Fabian Denner (Otto-von-Guericke-Univ. Magdeburg, Magdeburg, Germany)

We present a physical model and a numerical method to simulate nonlinear acoustic waves emitted from moving boundaries in a moving background flow field with transition from subsonic to supersonic flow speeds. The physical model is based on a convective form of the Westervelt equation and accounts for the motion of the background medium and the progressive distortion of a finite amplitude acoustic wave. A suitable coordinate transformation in physical space allows us to accommodate the motion of the wave emitting boundary, while solving the governing equation in a fixed computational domain using standard finite difference techniques. We present the case of an oscillating spherical wave emitting boundary with an induced spherical flow as a prototypical example of an acoustic black hole analogue, where the acoustic waves cannot escape from the horizon of sonic flow speed during the contraction phase. It is demonstrated that the accelerating motion of the wave emitting boundary gives rise to frequency sidebands and amplitude modulations of the acoustic wave. The influence of the background flow speed on this nonlinear Doppler effect is investigated with the present methodology, thus contributing to an improved understanding of the acoustic wave behavior in the proximity of a sonic horizon.

Session 4aSAb

Structural Acoustics and Vibration: Acoustic Radiation from Structures

Trevor Jerome, Cochair
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Allison M. King, Cochair
Mechanical Engineering, University of Michigan, 2370 GG Brown, Ann Arbor, MI 48109

Contributed Papers

10:30

4aSAb1. Vibration response and acoustic radiation of a thin floating plate. Allison M. King (Mech. Eng., Univ. of Michigan, 2370 GG Brown, Ann Arbor, MI 48109, kingalli@umich.edu) and David R. Dowling (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

Remote sensing may be used to detect and locate acoustic sources on a thin floating plate as well as predict the geometry and material of the structure based on recordings of vibratory and acoustic waves. Remote detection and localization of impacts on a thin floating structure is desirable. For a transient excited thin floating structure, the coupling of the vibration waves in the structure and the acoustic waves radiating from it into the air above and water below can introduce complexity not encountered with traditional source identification techniques. To study this problem, repeatable axisymmetric transient impact experiments are conducted in a 1.2-m diameter cylindrical water tank by dropping a stainless-steel ball bearing onto a 6.4-mm-thick aluminum plate suspended at the air-water interface. Accelerometers and laser Doppler vibrometry are used to record the vibration of the plate. Acoustic radiation is recorded in the air above the plate and in the water below it using linear arrays of 6 microphones and hydrophones, respectively. Wave speeds in the plate ranging between 430 m/s and 1550 m/s are considered. Results are compared to a combined acoustic-structure finite element model. [Sponsored by NAVSEA through the NEEC.]

10:45

4aSAb2. Experimentally determining power flow in cylindrical shells using a scanning laser Doppler vibrometer. Tysum Ruchti (Mech. Eng., Brigham Young Univ., 635 N 100 E, Provo, UT 84606, tysumruchti@gmail.com), Jonathan D. Blotter (Mech. Eng., Brigham Young Univ., Provo, UT), and Scott D. Sommerfeldt (Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Power flow methods are used to identify energy sources and sinks in objects and to visualize transmission pathways. The most common method uses surface acceleration or velocity measurements and finite difference methods to compute power flow. In flat structures, a scanning laser Doppler vibrometer (SLDV) has been used to increase the measurement resolution. For more complicated surfaces, the general approach has been to use accelerometers at a few key points and measure transmission rather than seeking to get a full visualization of the power flow. This research uses a 3-D-SLDV to take surface velocity measurements for a simply supported cylinder. The 3-D velocity data is used to solve for a quintic B-spline representation of the velocity field. Similar representations for each of the force resultants are generated using an isotropic material model and the Love equations for cylindrical shells. The resultant and velocity fields are used to compute the direction and magnitude of the energy transfer. This method is validated by comparing experimental results to the outputs of an analytical solution and to a finite element model.

11:00

4aSAb3. Determination of radiated sound power from acoustic sources using the VBSP method and a transparent acoustic boundary. Ian C. Bacon (Dept. of Phys. & Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, ianbacon24@gmail.com), Gibson H. Campbell (Dept. of Phys. & Astronomy, Brigham Young Univ., Provo, UT), John C. Ebeling (Dept. of Mathematics, Brigham Young Univ., Provo, UT), Scott D. Sommerfeldt (Dept. of Phys. & Astronomy, Brigham Young Univ., Provo, UT), and Jonathan D. Blotter (Dept. of Mech. Eng., Brigham Young Univ., Provo, UT)

Significant progress has been made to develop a vibration-based sound power (VBSP) method as an alternative means to determine the radiated acoustic energy from structures. This method has been validated for flat plates, cylindrical-shells, simple curved plates, and arbitrarily curved panels. However, many acoustic sources, such as a blender or motor, have sound power contributions that are enclosed within the structure and therefore cannot be scanned properly using a vibrometer. A rigid enclosure with a single mylar face for one side of the enclosure was fabricated to enclose the acoustic source, such as an "acoustic tent" around the structure of interest, so that the VBSP method could be applied to and obtain the radiated sound power. Due to the nature of the sources, this application of the VBSP method will have limitations that are not present in the aforementioned work. A calibration curve was developed to account for the effects that the tent had on the sources used for testing and used to correct the experimental VBSP sound power measurements from the acoustic tent. Results will be shown comparing the sound power obtained from the VBSP method with the sound power of the source in a reverberation chamber using ISO 3741. Funding for this work was provided by the National Science Foundation (NSF). [Funding for this work was provided by the National Science Foundation (NSF).]

11:15

4aSAb4. Sound power from coupled structures using elementary radiators. Sarah A. Ostergaard (Mech. Eng., Brigham Young Univ., 2131 N 40 W Apt. 333, Provo, UT 84604, sarahaostergaard@gmail.com), Jonathan D. Blotter, Suzanna Gilbert (Mech. Eng., Brigham Young Univ., Provo, UT), Scott D. Sommerfeldt (Brigham Young Univ., Provo, UT), and Trent P. Bates (Mech. Eng., Brigham Young Univ., Provo, UT)

Sound power measurements are of interest in numerous applications, and several methods are available for obtaining the sound power of a system. A recent method that has been developed, referred to as the Vibration-Based Sound Power (VBSP) method, utilizes vibration measurements obtained from a Scanning Laser Doppler Vibrometer (SLDV). The VBSP method has previously been used to obtain the sound power radiated from flat plates, cylindrical shells and curved plates. This paper will present further developments to extend VBSP measurements to multiple coupled structures and investigate the acoustic coupling effects that may exist. Two coplanar flat plates separated by a distance d were modeled using the

4a THU. AM

boundary element method. The sound power was determined within the boundary element software and the structural velocities were also exported and processed using the VBSP method. Computational and experimental results will be shown to demonstrate abilities and limitations for the VBSP method to accurately determine the radiated sound power for coupled structures. [Funding for this work was provided by the National Science Foundation (NSF).]

11:30

4aSAb5. Improved efficiency of vibration-based sound power computation through multi-layered radiation resistance matrix symmetry. John C. Ebeling (Mathematics, Brigham Young Univ., Provo, UT, jcebeling@gmail.com), Ian C. Bacon (Dept. of Phys. & Astronomy, Brigham Young Univ., Provo, UT), Trent P. Bates (Mech. Eng., Brigham Young Univ., Provo, UT), Scott D. Sommerfeldt (Dept. of Phys. & Astronomy, Brigham Young Univ., Provo, UT), and Jonathan D. Blotter (Mech. Eng., Brigham Young Univ., Provo, UT)

Calculating sound power using complex-valued surface velocities is becoming an established practice in structural acoustics with rising optimism about this method's potential versatility compared to traditional pressure-based methods. One approach involves using a geometry-dependent acoustic radiation resistance matrix multiplied by a velocity vector to compute sound power for a given frequency range. At scale, the computational costs incurred through this calculation limits the application of this method. Given a discretized surface with constant spacing and using a well-informed average radiator area approximation, a multilayered Toeplitz symmetry exists in the radiation resistance matrix. This symmetry, its origins and necessary assumptions are explored. By exploiting the Toeplitz symmetry, computationally expensive mathematical operations that used to be performed on the entire radiation resistance matrix, can be performed on a single row of the matrix, and then expanded using the pattern that will be presented. This approach preserves accuracy and greatly accelerates the processing, as evidenced through experimental data. The approach resulted in a maximum of ~1300% computation time reduction for single radius curved plate calculations and a ~9,600% computation time reduction for cylindrical shell calculations. [Funding for this work was provided by the National Science Foundation (NSF).]

11:45

4aSAb6. A geometrical approach to estimate the modal radiation efficiency of complex structures. Carlos García A. (Roberval (Mech., Energy and Electricity), Ctr. de Recherche Royallieu, Univ. de Technol. de Compiègne, Rue du Docteur Schweitzer, Compiègne 60200, France, carlos.garcia@utc.fr), Nicolas Dauchez, and Gautier Lefebvre (Roberval (Mech., energy and electricity), Ctr. de Recherche Royallieu, Univ. de Technol. de Compiègne, Compiègne, France)

The radiation efficiency of plates has been the study of many works. The rectangular simply supported plate is the most studied case, followed by the circular one. Analytical formulas and approximations for multiple boundary conditions, baffled and unbaffled cases have been the subject of study for decades. For a structure with a complex shape, numerical methods are appropriated such as FEM and BEM approaches. Semi-analytical based on Rayleigh's integral also appear in the literature to calculate the radiation, mixed with the surface velocity calculated by FEM; they are easy to implement but they are resource consuming as large matrices are presented. In this work, we present a method for estimating the modal radiation efficiency of simply supported and clamped complex plates, based on the theory of Greenspan for circular radiators and with a geometrical approach.

THURSDAY MORNING, 26 MAY 2022 PLAZA BALLROOM E, 9:00 A.M. TO 11:15 A.M.

Session 4aSC

Speech Communication: Perspectives on Long-Distance Coarticulation

Melissa A. Redford, Cochair
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Chair's Introduction—9:00

Invited Papers

9:05

4aSC1. Beneath the surface: Interactions of subthreshold gestural systems. Sam Tilsen (Cornell Univ., 203 Morrill Hall, Ithaca, NY 14853, tilsen@cornell.edu)

Control of articulatory movements is dominated by gestural systems whose activation states are above a threshold. These highly active systems strongly drive activity in spatial fields encoding articulatory targets, thereby governing the kinematics of vocal tract systems. In the presence of these strong forces, gestural systems with below-threshold activation exert weaker, sometimes opposing forces on target fields. These subtle forces can generate consistent subcategorical patterns of phonetic variation conditioned on the co-activation of gestural systems. Depending on the relations between targets of gestural systems and their force distributions, this variation can give rise to different types of long-distance phonological patterns.

9:25

4aSC2. The acquisition of gestural timing. Phil J. Howson (Lab. Phonology, Leibniz-Zentrum Allgemeine Sprachwissenschaft, 324 E 12th Ave. Unit 4, Eugene, OR 97401, philh@uoregon.edu)

Motor plans are complex and consist not only of constriction location and degree, but also gestural timing. For children to acquire adult-like speech, they need to acquire complex timing relationships that can result in delayed acquisition. To examine this phenomenon, we examined long-distance timing patterns for English liquids, /r, l/, and short-distance timing patterns for fricatives, /θ, s, ʃ/, for 5- and 8-year-olds compared to adults. To examine the long-distance timing of liquids, I used an AV-Gated paradigm where participants saw clips of speech and predicted an upcoming /r/ or /d/ (Exp1a) or upcoming /l/ or /d/ (Exp1b). The results revealed long distance timing sequences for /r/ and /l/, with the timing for /r/ extending to at least five segments prior and /l/ 2 segments prior. However, 5-year-olds exhibited extremely long-distance timing for both /r/ and /l/, while 8-year-olds had longer distance timing for /r/ and /l/, but still not adult-like timing. Results suggest long-distance timing must be acquired. In the second experiment, we examined the dynamic COG of fricatives from 5- and 8-year-olds compared to adults. Dynamic COG results revealed significantly more vowel induced variability for /θ/ for adults than for 5- and 8-year-olds, with similar degrees of variability for /s, ʃ/. Overall, the results suggest that acquisition of timing increases the relative complexity of acquisition, and that segment specific articulation interferes with acquisition of timing in different ways.

9:45

4aSC3. Understanding anticipatory speech postures: does coarticulation extend outside the acoustic utterance? Peter A. Krause (Psych., California State Univ. Channel Islands, One University Dr., Camarillo, CA 93012, peter.krause066@csuci.edu)

Multiple studies strongly suggest that anticipatory coarticulation has a planned component. Speakers forced to begin an acoustic utterance before knowing all details of how it will unfold show reduced coarticulatory influences from those initially unknown speech segments. Therefore, anticipatory coarticulation as we currently understand it may reflect a more general process whereby speech acts that are currently planned, but whose acoustic consequences remain in the future, bias current articulator positions. This more general process could be in effect even before an utterance is acoustically active. The present talk will review the author's recent work in anticipatory speech postures. Specifically, these findings suggest that speakers bias their lips to anticipate utterances yet to be acoustically initiated, in a way that tracks currently available information. In laboratory tasks, speakers primed with just the initial consonant or nuclear vowel of an upcoming monosyllabic utterance shape their lips to reflect this priming. Similarly, speakers in natural conversation produce anticipatory lip postures while awaiting their turn, but only when the intended utterance is quite short. The conceptual links between these findings and anticipatory coarticulation will be discussed.

10:05–10:25 Break

10:25

4aSC4. What long-distance coarticulation can tell us about coarticulatory development. Margaret Cychosz (Hearing and Speech Sci., Univ. of Maryland, 7251 Preinkert Dr., College Park, MD 20742, mcychosz@umd.edu)

It is widely acknowledged that children's speech gestures overlap more than adults' in spoken language. Unlike coarticulation in adult speech, which is largely planned (Whalen, 1990), children's coarticulation appears to stem from their (1) protracted tongue-jaw coordination and (2) inexperience with language. For example, 4- to 7-year-olds with greater phonological awareness tend to coarticulate less within syllables (Noiray, *et al*., 2019). Long-distance, inter-morphemic coarticulation is another way to demonstrate that children coarticulate for different reasons than adults: by comparing across multiple segments, and contrasting coarticulation within versus across word morphemes, we can invoke speech planning, not just motor control. Since children's coarticulation is assumed to reflect inexperience and lack of planning, they should coarticulate *less* than adults in longer-distance environments. N = 51 5- to 10-year-old children and 10 female adults, all bilingual South Bolivian Quechua-Spanish speakers, completed a Quechua morphological elicitation task. Inter-syllabic coarticulation was quantified using a whole-spectrum measure within bare stems (within-morpheme) and inflected nouns (across-morpheme). Results were compared to speakers' intra-syllabic coarticulation and showed that while children, unsurprisingly, coarticulate more *within* syllables, they anticipate segments at a longer distance (inter-syllabic and across morphemes) less than adults, only approximating adult-like patterns around age 9.

Contributed Papers

10:45

4aSC5. V-to-V coarticulation may be extinct in the wild. Hannah J. Scott (Comput. Sci., Oregon State Univ., Fresno, CA) and Sean A. Fulop (Linguist., California State Univ. Fresno, 5245 N Backer Ave. Linguist PB92, Fresno, CA 93740-0001, sfulop@csufresno.edu)

Ever since the results of [S. E. G. Öhman, "Coarticulation…," *J. Ascoust. Soc. Am.* **39**, 151 (1966)] that a vowel could affect the preceding one, the phenomenon of anticipatory VCV coarticulation has been a subject of study. Most results establishing these effects have been obtained from curated laboratory speech [e.g., M. Grosvald, "Interspeaker variation…," *J. Phonetics* **37**, 173 (2009)], showing anticipatory V-to-V coarticulation across one, two and three syllables in English utterances. Here, we attempted to replicate these findings "in the wild" using the Buckeye Corpus of conversational English. F1 and F2 of [ʌ] or [ə] tokens were obtained in all [ʌ/ə]C(C…)V contexts. Two-dimensional Kolmogorov-Smirnoff tests were carried out to compare the F1/F2 distributions of these tokens in different contexts. The formant distributions of [ʌ/ə] in specific vowel contexts were also compared with the total formant distributions of [ʌ/ə] in all contexts. The replication effort was a failure. Essentially no significant differences of the [ʌ/ə] formant distributions could be found among the data, no matter how extreme the difference of the following vowel (e.g., [i] vs [ɑ] contexts). This poses a problem for theories of speech perception which rely on such effects.

11:00

4aSC6. Anticipatory vowel coarticulation in child versus adult speech and a model of developmental change. Melissa A. Redford (1290 Linguist, Dept., University of Oregon, Eugene, OR 97403-1290, redford@uoregon.edu), Maya Davis, Carissa A. Diantoro, and Jeffrey Kallay (Linguist., Univ. of Oregon, Eugene, OR)

There is by now extensive evidence to suggest that coarticulatory domains are at least as large in children's speech as in adults' speech. A series of four experiments that investigated the temporal domain of anticipatory vowel articulation in speech elicited from school-aged children and college-aged adults under different conditions (random elicitation, repetitive elicitation, spontaneous speech) confirms previous findings and suggests a new one: anticipatory cues to an upcoming vowel ramp up more quickly in adult's speech compared to children's speech. Both the established and new findings can be modeled with reference to the windows that govern path selection in perceptual-motor space. Specifically, the findings are modeled as a developmental decrease in the size of a lookback window, to be interpreted as a developmental increase in inhibitory motor control. [Work supported by NIH under grant R01HD087452.]

THURSDAY MORNING, 26 MAY 2022 GOVERNORS SQUARE 16, 8:30 A.M. TO 12:00 NOON

Session 4aSP

Signal Processing in Acoustics, Computational Acoustics, Physical Acoustics, and Underwater Acoustics: Model Based Signal Processing, Bayesian Learning, and Machine Learning I

Ananya Sen Gupta, Cochair
Department of Electrical and Computer Engineering, University of Iowa, 103 S. Capitol Street, Iowa City, IA 52242

Ning Xiang, Cochair
School of Architecture, Rensselaer Polytechnic Institute, 110 Eighth Street, Troy, NY 12180

Scott H. Hawley, Cochair
Department of Chemistry and Physics, Belmont University, 1900 Belmont Blvd., Nashville, TN 37212

Chair's Introduction—8:30

Invited Papers

8:35

4aSP1. Development tools for deep learning models of acoustical signal processing. Scott H. Hawley (Dept. of Chemistry and Phys., Belmont Univ., 1900 Belmont Blvd, Nashville, TN 37212, scott.hawley@belmont.edu)

We present a survey of available frameworks for developing acoustical signal processing models based on deep neural networks. Given that this is a dynamic space with new frameworks, libraries, and even companies appearing on timescales measured in months, we provide an up-to-date assessment of the strength, popularity, and near-future directions of several tools and platforms available for research and product deployment for deep learning models of audio signal processing. Similarly, those new to these spaces may be unaware of software systems that will allow them to obtain and interrogate results more quickly and easily, while also integrating the nearly state-of-the-art optimization methods. Included tools, packages and platforms include PyTorch, Tensorflow, Keras, JAX, fastai, PyTorch Lightning, Julia, nbdev, HuggingFace, Weights and Biases, and Gradio. Examples will be drawn from the speaker's recent research publications in musical signal processing and computer vision applied to musical acoustics, as well as recent work by others. The goal of the talk is to provide acoustics researchers, educators, students with a set of helpful possibilities for pursuing and improving their understanding, research practices, and communications.

8:55

4aSP2. Bayesian approach to passive receiver localization in a randomly scattered sound field. D. Keith Wilson (U.S. Army Engineer Res. and Development Ctr., U.S. Army ERDC-CRREL, 72 Lyme Rd., Hanover, NH 03755-1290, D.Keith.Wilson@usace.army.mil), Chris L. Pettit (Aerosp. Eng., U.S. Naval Acad., Annapolis, MD), Vladimir Ostashev, and Anthony Fragoso (U.S. Army Engineer Res. and Development Ctr., Hanover, NH)

A Bayesian approach is described for localizing sources and receivers in a randomly scattered sound field, for which the complex Wishart distribution represents the joint distribution of the signals at multiple, arbitrarily separated receivers. The signals are assumed to be narrowband and fully saturated by scattering from homogeneous turbulence (or similar scattering processes) as they propagate along line-of-sight paths to the receivers. Here, we consider a scenario in which there is an array of receivers at known locations, a source of opportunity with unknown amplitude at an unknown location, and a single receiver at an unknown location. The objective is to passively determine the unknown receiver location even though the source characteristics are not initially known. A slice-sampling strategy is found to be suitable for solving the problem. Simulations show that a small number of independent signal samples at the receivers initially enables the unknown receiver to be localized on a circle of constant amplitude around the source. With more samples, the correlation characteristics of the field enable full determination of the unknown receiver location.

9:15

4aSP3. Invertible neural networks for reconstructing acoustic fields. Xenofon Karakonstantis (Elec. Eng., Tech. Univ. of Denmark, Ørsteds Plads, Bldg. 352, Kgs. Lyngby 2800, Denmark, xenoka@elektro.dtu.dk) and Efren Fernandez-Grande (Dept. of Elec. Eng., Tech. Univ. of Denmark, Kgs Lyngby, Hovedstaden, Denmark)

Sound field reconstruction from finite measurement arrays provides a means to interpolate and extrapolate acoustic quantities that describe the field. By assuming a linear projection on a basis that follows a principled source propagation, one can recover accurate estimates of the aforementioned sound fields. However, the recovery of the basis coefficients relies on explicit models of which measurement noise and data incompleteness can profoundly affect the uncertainty of the solution. This work aims to estimate the distribution of the underlying pressure conditioned on the observations of the measured pressure in a room. A framework for approximate inference is adapted for sound field reconstruction by applying generative flow-based models and invertible neural network architectures. In particular, we use conditional normalising flows for fast conditional posterior estimation and uncertainty quantification. The model's evaluation is carried out using experimental data measured with a spherical array and compared to hierarchical Bayes with Markov Chain Monte-Carlo sampling.

9:35

4aSP4. Prediction model formulations for detection, enumeration, and localization of multiple sound sources using spherical harmonics. Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., 110 Eighth St., Troy, NY 12180, xiangn@rpi.edu) and Thomas Metzger (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY)

A spherical microphone array is used to detect and localize sound sources in terms of model-based machine learning (ML). In this application, it is crucial to establish parametric models to distinguish background sound environment from presence of sound sources. In the presence of sound sources, the parameter models are also used to localize an unknown number of potentially multiple sound sources. In this work, a model-based Bayesian learning framework is presented for localizing an unknown number of sound sources. Among them, a no-source scenario needs to be accounted for. The model-based machine learning applies the model comparison between the no-source model and the one-source model for sound source detection. After detecting sound sources, the machine learning needs to involve sound source enumeration and localization in order to correctly localize potential multiple sound sources. Specifically, sound environment is analyzed using Bayesian model comparison of two different models accounting for absence and presence of the sound sources for source detection. Upon a positive detection, potentially multiple source models are involved to analyze direction of arrivals (DoAs) for far-field and to localize sound sources for near-field including source distances, amplitudes, and DoAs using Bayesian model selection and parameter estimation.

9:55

4aSP5. Temporal and spectral analysis of melody perception with different simulated cochlear implant coding strategies. Epri W. Pratiwi (Elec. Eng., National Central Univ., No. 300, Zhongda Rd., Zhongli District, Taoyuan City 320317, Taiwan (R.O.C.), Zhongli, Toayuan 320317, Taiwan, epriwahyupratiwi@gmail.com) and Chao-Min Wu (Elec. Eng., National Central Univ., Zhongli, Taoyuan, Taiwan)

This study examined the contribution of pitch and rhythm cues in melody recognition, as well as the quality from three well-known CI coding strategies, Advanced Combinational Encoder (ACE), Fundamental Frequency Modulation (F0mod), and Envelope Enhancement (EE). Firstly, we processed the 30 popular melodies of Taiwan children's songs using NCU-CI, a cochlear implant simulation software. In the pilot subjective listening test ($n = 5$), rhythm cues perceived better ($p < 0.05$) by having a higher percent correct and faster response time than pitch cues. Foremost, rhythm cues processed by ACE strategy achieved the best score, 86.80%. Then, objective tests were conducted to measure the spectral and temporal quality of processed melody using log spectral distance (LSD), intensity mismatch pattern, and envelope difference index (EDI). LSD between original and processed by the ACE, F0mod, and EE strategy were 2.10, 2.16, and 2.19, respectively. Then, average intensity mismatch patterns between original and processed by the ACE, F0mod, and EE strategy were 5.9, 6.4, and 6.0, respectively. The lower LSD and intensity mismatch pattern, the better the amplitude melody and spectral quality was preserved. Then, EDI between original and processed by the ACE, F0mod, and EE strategy were 0.11, 0.11, and 0.15, respectively. The higher EDI, the better the temporal envelope was preserved. It reveals that rhythm cues combined with the ACE strategy performed the best feature in CI melody perception.

10:15–10:30 Break

10:30

4aSP6. Extracting useful machine learning features from acoustic resonance spectra of coupled multi-body structures. John Greenhall (Los Alamos National Lab., P.O. Box 1663, Los Alamos, NM 87545, jgreenhall@lanl.gov), Eric S. Davis (MPA-11, Los Alamos National Lab., Los Alamos, NM), Peter Kendall, Alan Graham, Dipen N. Sinha (Los Alamos National Lab., Los Alamos, NM), and Cristian Pantea (Mater. Phys. and Applications, Los Alamos National Lab., Los Alamos, NM)

The usefulness of machine learning algorithms is highly dependent on the formulation of relevant features that sufficiently represent the model. Acoustic resonance spectra consist of a series of peaks, representing resonant modes of the system and contain detailed information about the system structure, material, boundary forces, etc. We present a technique for extracting useful features from dense acoustic resonance spectra of multi-component systems. For simple geometries, the resonance spectrum is relatively sparse and it is feasible to track individual peaks to quantify properties of the system. However, for multi-component systems, the acoustic resonance spectra consist of overlapping peaks, corresponding to resonances in different components. As a result, a high density of peaks exists, and some peak positions are sensitive to changes in the contact between components. Thus, tracking specific resonance modes becomes challenging. Instead, we combine principles from wavelet transformation, nonlinear normalization, and genetic algorithm optimization to extract useful features from complicated acoustic resonance spectra. We demonstrate this technique on simulated acoustic resonance spectra for multi-layer structures. Here, we are measuring the thickness of a specific layer, which is hampered by changes in the acoustic resonance spectrum due to variation in the other layer thicknesses as well as delamination defects.

10:45

4aSP7. Exploiting spatio-temporal spectral features of the indoor acoustic field for sound source localization. Pratik Gandhi (ECE, Univ. of Massachusetts Lowell, 1 University Ave., FA 203, Lowell, MA 01854, pratik_gandhi1@student.uml.edu), Nathan Uhunsere, Adonai Paul, Kavitha Chandra, and Charles Thompson (ECE, Univ. of Massachusetts Lowell, Lowell, MA)

The estimation and classification of spectral features of the spatio-temporal acoustic field in a reverberant rectangular enclosure is undertaken with application to localizing sound sources. The field is constructed using an image source model of the impulse response that captures the acoustic properties of the enclosure boundaries. The received field is examined considering linear and randomly distributed array configurations and spectral peaks are characterized to derive a set of features for classification and source identification. The effects of noise generated by the environment and the diffuse field from multiple reflections is distinguished.

11:00

4aSP8. Pleasantness assessment of electric vehicle interior sounds using a long short-term memory model. Florian Doleschal (Dept. of Experimental Audiol., Otto von Guericke Univ. Magdeburg, Leipziger Str. 44, Magdeburg, Sachsen-Anhalt 39120, Germany, florian.doleschal@med.ovgu.de) and Jesko Verhey (Dept. of Experimental Audiol., Otto von Guericke Univ. Magdeburg, Magdeburg, Sachsen-Anhalt, Germany)

The electrification of the powertrain strongly reduces the sound pressure level in the vehicle interior. However, the absence of a broadband noise originating from the combustion engine unmasks high-frequency tonal components of the electric motor and the gearbox. As they are commonly audible in transient driving conditions, the related psychoacoustic parameters and their influence on the pleasantness have to be considered dynamically. This study presents a sequence-to-one regression model on the basis of a long short-term memory (LSTM) neural network, which models the relation between times series of psychoacoustic parameters and the overall (i.e., single-valued) pleasantness. The advantage of LSTM-based models is that they consider interdependencies between time steps. The data set to be modelled mainly consisted of pleasantness ratings of recorded sounds from the vehicle interior of pure-electric and hybrid vehicles. The data set also includes pleasantness ratings of altered sounds, where synthetic components were added, amplified or attenuated using a sound separation algorithm. The model is highly accurate in predicting the data. Thus, the LSTM model can be used as an automated pleasantness assessment for the interior sound of vehicles with electrified drives.

11:15

4aSP9. Applying matched field array processing and machine learning to computational auditory scene analysis and source separation challenges. J. K. McElveen (Wave Sci., 151 King St., Charleston, SC 29401, keith.mcelveen@wavesciencescorp.com), Leonid Krasny (Wave Sci., Cary, NC), and Scott Nordlund (Wave Sci., Charleston, SC)

Matched field processing (MFP) techniques employing physics-based models of acoustic propagation have been successfully and widely applied to underwater target detection and localization, while machine learning (ML) techniques have enabled detection and extraction of patterns in data. Fusing MFP and ML enables the estimation of Green's Function solutions to the Acoustic Wave Equation for waveguides from data captured in real, reverberant acoustic environments. These Green's Function estimates can further enable the robust separation of individual sources, even in the presence of multiple loud, interfering, interposed, and competing noise sources. We first introduce MFP and ML and then discuss their application to Computational Auditory Scene Analysis (CASA) and acoustic source separation. Results from a variety of tests using a binaural headset, as well as different wearable and free-standing microphone arrays are then presented to illustrate the effects of the number and placement of sensors on the residual noise floor after separation. Finally, speculations on the similarities between this proprietary approach and the human auditory system's use of interaural cross-correlation in formulation of acoustic spatial models will be introduced and ideas for further research proposed.

11:30

4aSP10. PLASTIC: Pressure and location-based acoustic sensing touch interface using classification. David A. Anderson (Rocky Mountain Div., Appl. Res. Assoc., Inc., 1250 S. Monaco Pkwy, Unit 42, Denver, CO 80224, danderson@ara.com)

A method is described for developing a pressure sensor panel that uses a single small electromechanical exciter, delivering a sawtooth wave, and a single sensor attached to a panel or structure. A trained classifier is able to recognize the pressure and location of a finger-sized object pressing on the panel based on the characteristics of the waveform detected at the sensor. The classification results of three experiments are reported: 55 virtual buttons on a panel, the pressure classification sensitivity of a single virtual button, and the location classification resolution of a tightly packed grid of points. All three experiments show excellent accuracy (99.7%, 85.2%, 96.0%, respectively) using a simple trained linear SVM classifier with a 70/30 training/validation data ratio, demonstrating that the PLASTIC method is an effective method for classifying the location and pressure of a touch on a panel or structure.

11:45

4aSP11. Machine learning-based room acoustics using flow maps and physics-informed neural networks. Nikolas Borrel-Jensen (Elec. Eng., Tech. Univ. of Denmark, Ørsteds Plads, Kgs. Lyngby 2800, Denmark, nibor@elektro.dtu.dk), Allan P. Engsig-Karup (Appl. Mathematics and Comput. Sci., Tech. Univ. of Denmark, Kgs. Lyngby, Denmark), and Cheol-Ho Jeong (Elec. Eng., Tech. Univ. of Denmark, Kgs. Lyngby, Denmark)

The development of efficient and accurate numerical methods for simulating realistic sound in virtual environments—such as computer games and VR/AR—has been an active research area for the last decades. However, handling dynamic scenes with many moving sources is still challenging due to intractable storage requirements and extensive computation time. A recently proposed physics-informed neural network (PINN) approach learns

a compact and efficient surrogate model with parameterized moving sources and impedance boundaries on a grid-less 1-D domain. Contrary to traditional "black-box" deep learning, PINNs minimize the residuals of the governing equations through the loss function. We will extend this work using flow maps implemented as Residual Networks (ResNets). ResNets are interpreted from a dynamic systems perspective as ordinary differential equations that can be used as building blocks to approximate the governing equations in time. We will examine the pros and cons of ResNets in acoustics and compare them with state-of-the-art numerical methods and vanilla feed-forward neural networks in terms of accuracy and efficiency.

THURSDAY MORNING, 26 MAY 2022 GOVERNORS SQUARE 14, 8:30 A.M. TO 11:30 A.M.

Session 4aUW

Underwater Acoustics: General Topics in Underwater Acoustics II

Daniel J. Brooker, Cochair
Underwater Acoustics, Navy Research Lab, 4555 Overlook Ave. SW, Code 7167, Washington, D.C. 20375

Fumin Zhang, Cochair
Electrical and Computer Engineering, Georgia Institute of Technology, 85 Fifth Street NW, Atlanta, GA 30332

Contributed Papers

8:30

4aUW1. Underwater acoustic source detection using a median based estimator. Daniel J. Brooker (Underwater Acoust., Navy Res. Lab., 4555 Overlook Ave. SW, Code 7167, Washington, DC 20375, daniel.brooker@nrl.navy.mil)

Recently a class of cross spectral density matrix (CSDM) estimators have been introduced based on the use of matrix medians to average signal dyads rather than the more conventional sample mean. Due to the medians robustness against outliers, these new estimators are intended to for use at low SNR or in highly cluttered areas. The median in this case is given a geometric interpretation, as the point which minimizes the sum of the distance to all of the samples. This geometric interpretation has been shown to be quite useful for matched field processing, and in this work it is applied to detection of unknown signals at low SNR. Several estimators are developed corresponding to different choices of non-Euclidean distance between CSDMs. These new estimators are then applied to a relative entropy based detection scheme which detects unknown signals by comparing them to *in situ* estimated noise and finding individual frequencies which are highly dissimilar from this noise estimate. Using data from the DSNCON-19 experiment the performance of these novel estimators is characterized by their Receiver Operating Characteristic (ROC) curve and compared to the maximum likelihood estimator (MLE) performance. [Work supported by the Office of Naval Research.]

8:45

4aUW2. Ice anthropogenic classification with acoustic vector sensors using transformer neural networks. Steven Whitaker (Dept. of Elec. and Comput. Eng., Michigan Technol. Univ., 1600 Townsend Dr., Houghton, MI 49931, sjwhitak@mtu.edu), Andrew Barnard (Acoust., Penn State, State College, PA), George D. Anderson (Naval Undersea Warfare Ctr., Newport, RI), and Timothy Havens (Dept. of Comput. Sci., Michigan Technol. Univ., Houghton, MI)

Acoustic classifiers are a necessary component in understanding the source. When a foreign object has been classified, physics models can be associated with the foreign object for better localization and tracking. In highly non-linear environments, like shallow ice environments, traditional classifiers cannot properly consider its compounded non-linearities: multipath, reflective surfaces, scattering fields, and the dynamic acoustic properties of first-year ice. With such significantly distorted signals, we deploy deep neural networks to better classify different acoustic sources. We collected data from 8 different acoustic sources on the Keweenaw Waterway in Houghton, Michigan: a narrow and shallow channel covered with first-year ice. Two sources were moving and the other five were stationary; the sources did not emit simultaneously. Data were recorded using two spatially separated underwater acoustic vector sensors; their time-series data were post-processed into mel-frequency cepstral coefficients (MFCC) and analyzed with different deep neural network architectures. A deep Transformer neural network and a deep residual neural network were then compared in their ability to predict which source was emitting. Preliminary results show success with the deep Transformer neural networks.

9:00

4aUW3. Modeling and learning underwater acoustic channel parameters through deep recursive neural networks. Fumin Zhang (Elec. and Comput. Eng., Georgia Inst. of Technol., 85 Fifth St. NW, Atlanta, GA 30332, fumin@gatech.edu), Ziqiao Zhang (Elec. and Comput. Eng., Georgia Inst. of Technol., Atlanta, GA), and Feng Tong (Xiamen Univ., Xiamen, Fujian, China)

We present a method that employes deep recursive neural networks (RNN) to model channel parameters of underwater acoustic channels. A deep RNN has the capacity to incorporate multiple data streams, such as temperature, wind, wave height that might affect channel property. The RNN is first trained using collected data and synthetic data. The training data is collected by sensors inboard a buoy installed at the target deployment area, and the synthetic data is generated through a simulation program based on physics principles. After the RNN is deployed, a dynamic filter is applied to update the channel parameter estimation based on real-time measurements in operations. We present several candidates for the dynamic filtering algorithms and compare their performances with other channel modeling

techniques. It is observed that RNN demonstrates improved performance while sufficient data is available.

9:15

4aUW4. Resolution of matched field processing for a single hydrophone in a rigid waveguide. Margaret Cheney (Mathematics and ECE, Colorado State Univ., Fort Collins, CO) and Ivars Kirsteins (NUWCNUWC, Newport, RI, ivars.kirsteins@navy.mil)

This paper studies resolution of matched field processing for locating, in range and depth, a broadband underwater acoustic source from data measured at a single hydrophone receiver. For the case of an ideal rigid shallow-water waveguide with a pressure-release top boundary and a rigid bottom boundary, we derive approximations for the main-lobe widths of the ambiguity surface. The two cases studied in this paper are (1) when coherent measurements of the pressure are available, with the transmitted source waveform precisely known, and (2) when only measurements of the received signal power spectral density (PSD) are available, such as occur when the transmitted signal is random and unknown. The analysis uses the normal-mode expansion for the pressure field to derive approximate expressions for the ambiguity surface main-lobe widths, as a function of the number of modes and frequency band, for both range and depth. Numerical results are presented corroborating the analytical analysis. Finally, we argue that this ambiguity analysis also gives insights into real ocean waveguide localization characteristics under appropriate conditions, and show numerical simulations of matched field localization ambiguity surfaces for some realistic shallow-water Pekeris environments.

9:30

4aUW5. Sensitivity of spatial coherence of sub-bottom scattering to sediment attenuation. Anthony P. Matriss (The Penn State Univ., 201 Appl. Sci. Bldg., State College, PA 16802, apm6227@psu.edu), Daniel C. Brown, and Thomas E. Blanford (The Penn State Univ., State College, PA)

The spatial coherence of seafloor scattering is determined by a complex interaction between parameters describing the environment, the sensor, and the measurement geometry. Forward modeling shows that near normal incidence, the spatial coherence of the field backscattered from an isotropic sub-bottom may be sensitive to the sediment attenuation coefficient. This presentation will describe a parametric study investigating the sensitivity of spatial coherence to specific combinations of environment, sensor, and measurement geometry. Particular emphasis will be placed on identifying configurations where the spatial coherence is sensitive to the sediment attenuation coefficient. To support this analysis, a numerical model using the van Cittert–Zernike Theorem and a point-based scattering model will be used to provide forward modeling of the spatial coherence. These models are compared for several simulated environments of increasing complexity to explore the sensitivity of the spatial coherence to sediment attenuation. Agreement between these models will provide a path to developing the collection geometry and algorithms necessary for inversion using simulated data.

9:45

4aUW6. Resolution dependence of acoustic scattering statistics for complex seafloors. Alexander Lehman (Oceanogr., Naval Postgrad. School, 798 Belden St., Monterey, CA 93940, alexander.lehman@nps.edu) and Derek R. Olson (Oceanogr., Naval Postgrad. School, Monterey, CA)

Underwater Unmanned Vehicles (UUVs) utilize sonar perception to conduct sea floor mapping and target detection operations. However, cooperative surveys with multiple types of UUVs are difficult because platforms with different resolutions may generate different probability density functions (PDFs) of the magnitude of the complex pressure. An area of research that has not been adequately studied is the effects of resolution manipulation during the post-processing of high-resolution data from complex seafloor environments. This work analyzes synthetic aperture sonar (SAS) data collected from multiple seafloor geomorphologies surrounding Bergen, Norway to study the resolution dependence of scattering statistics for complex seafloors. Multi-look methods will be applied to reduce the resolution. The original data and reduced resolution data will be compared in terms of PDF amplitude and evaluated by standard goodness of fit tests with heavily tailed statistical models that are commonly used in the radar and sonar community, including mixture models. The goal of the paper is to provide a bridge to combining high-resolution and low-resolution sonar data together to enhance sonar perception. Data provided by the Norwegian Defense Research Establishment. [Funding provided by the Office of Naval Research.]

10:00–10:15 Break

10:15

4aUW7. Underwater acoustic localization using a modular differentiable model for acoustic wave propagation. Dariush Kari (Elec. and Comput. Eng., Univ. of Illinois Urbana-Champaign, 1308 W Main St., Urbana, IL 61801, dkari2@illinois.edu) and Andrew C. Singer (Elec. and Comput. Eng., Univ. of Illinois Urbana-Champaign, Urbana, IL)

Model-based algorithms for underwater acoustic localization are highly dependent upon environmental knowledge that is rarely available. Moreover, model-based localization usually involves optimization in high-dimensional spaces to find the best candidate for source localization. On the other hand, data-driven methods deliver poor generalization performance in data starved scenarios. Therefore, in order to improve generalization, we propose a modular, deep learning-based architecture that inherently learns the multipath structure, while learning the environmental properties from the training data. Furthermore, since ReLU-based fully connected networks cannot capture the high-frequency contents of the signals of interest properly, we use sinusoidal activations. Although the model needs an estimate of the carrier frequency as a hyper-parameter, it is observed that it can tolerate deviations about the true frequency. This model maps the source and receiver locations to received signals and is then used in a gradient descent manner exploiting the automatic differentiation toolbox in PyTorch, to find the source location. To evaluate the performance of the algorithm, we use Bellhop to generate data for a given set of environmental parameters. We show that our method outperforms matched-field processing and a deep fully connected network without a modular structure.

10:30

4aUW8. Centralized data repositories: NOAA's National Archives for Marine Acoustic Data. Veronica Martinez (CIRES, Univ. of Colorado, Boulder, 325 Broadway, 1B117, Boulder, CO 80305, veronica.martinez@noaa.gov), Charles Anderson, Carrie Wall, and Elizabeth Jimenez (CIRES, Univ. of Colorado, Boulder, Boulder, CO)

The NOAA National Centers for Environmental Information maintains two marine acoustic archives for the long-term stewardship of globally collected water-column sonar and passive acoustic data. These vast datasets are collected across NOAA and academia for a wide range of scientific objectives. The archives document the datasets using standards-driven metadata and preserve them on long-term storage systems. Users can discover, query, and access archived data using the archives' web-based map viewers. Further, cloud-based access to 200 + TB through the NOAA Big Data Program enables free and immediate download of desired data and allows users to bring processing routines to large volumes of data—from simple statistical analyses to artificial intelligence. Cloud-based tools are being developed in collaboration with our partners in the ocean acoustic community to visualize and analyze data in the archives. This allows researchers of varying backgrounds to easily understand the quality and content of these complex data. Providing free access to data archives and facilitating the utility of these data increases the potential for researchers to address new questions that will advance the field of marine ecosystem acoustics.

10:45

4aUW9. Lake experimentation of in-band full-duplex underwater acoustic communications. Zheng Guo (Elec. and Comput. Eng., Univ. of Alabama, Tuscaloosa, AL), Aijun Song (Elec. and Comput. Eng., Univ. of Alabama, 245 7th Ave. Tuscaloosa, AL 35487, song@eng.ua.edu), Mohammad Towliat, Leonard Cimini, and Xiang-Gen Xia (Elec. and Comput. Eng., Univ. of Delaware, Newark, DE)

In-band full-duplex (IBFD) communications is a promising technique to achieve spectral-efficient communication in the underwater environment. Characteristics of self-interference and methods to suppress the self-interference are crucial to attaining IBFD acoustic communications. In this effort, we analyze the characteristic of the self-interference for the underwater acoustic channel through experimental measurements. Self-interference reduction using physical separation and digital cancellation is investigated under different transmitter-receiver geometries. An iterative IBFD receiver combining joint channel estimation, SI cancellation, and multichannel decision feedback equalizer is developed. The feasibility of coherent IBFD underwater acoustic communications was tested in a lake environment. The self-interference suppression methods and the receiver performance were evaluated using the experimental measurements. IBFD communications was demonstrated at 80 m. With 10 dB local self-interference source level reduction, IBFD communications was extended to the 170 m range. [Work supported by the National Science Foundation.]

11:00

4aUW10. Non-coherent turbo coded frequency shift keying for reliable covert underwater acoustic communications. Hyun-Woo Jeong (Radio Commun. Eng., Korea Maritime and Ocean Univ., PUSAN, South Korea, gusdn0930@g.kmou.ac.kr), Ji-Won Jung (Radio Commun. Eng., Korea Maritime and Ocean Univ., Busan, South Korea), and In-Soo Kim (Agency for Defense Development, Changwon, South Korea)

This paper presented efficient receiver structure of non-coherent turbo coded FSK signals for maintaining covertness and performance in underwater communication. In aspect to covertness, direct sequence spread spectrum method, which used to hide the transmitted signal by transmitting it at low power and, thus, making it difficult for an unintended listener to detect the signal in the presence of background noise. In aspect to performance improvement in the underwater acoustic environment with multipath and Doppler spreading, non-coherent FSK which is not required phase information, weighted multiband technology, and turbo equalization model with rate of 1/3 were suggested. In order to analyze the performance by applying four bands and spreading chips with 8 and 32, the experiment was conducted on a lake by moving in the range of 300m to 500m. Through the experimental results, we confirmed the performance was improved as increasing number of bands, turbo iterations, and spreading chips. Furthermore, weighted multiband technology which allocates lower weight to inferior bands based on preamble error rates were applied for some failed packets, all packets were decoded successfully. [This research was funded by Agency for Defense Development, South Korea, Grant No. UD200010DD.]

11:15

4aUW11. Improvement of ray-based blind deconvolution through kurtosis-based phase correction. Seunghyun Yoon (Seoul National Univ., 1, Gwanak-ro, Gwanak-gu, Seoul 08826, South Korea, justin1128@snu.ac.kr), Haesang Yang, and Woojae Seong (Seoul National Univ., Seoul, South Korea)

Ray-based blind deconvolution (RBD) has demonstrated remarkable performance on several *in situ* underwater acoustic data. RBD estimates the channel impulse response by compensating the source phase estimated by beamforming. However, when beamforming fails to achieve the desired signal-to-noise ratio, RBD fails to achieve the expected performance. To address this issue, we propose an improved RBD framework that uses the kurtosis of the channel impulse response to provide additional phase correction after beamforming. The proposed method is validated with the measurement data from the shallow-water acoustic variability experiment 2015 using a high-frequency (11–31 kHz) broadband source and a vertical line array of 16 sensors located approximately 3 km away. The results are compared with conventional RBD and discussed from an optimization point of view.

4a THU. AM

Session 4pAA

Architectural Acoustics, Musical Acoustics, Noise, and Education in Acoustics: Music Education Facilities II

Shane J. Kanter, Cochair
Threshold Acoustics, 141 W Jackson Blvd, Suite 2080, Chicago, IL 60604

Lauren Ronsse, Cochair
University of Nebraska-Lincoln, 1110 S. 67th St., Omaha, NE 68182-0816

David T. Carreon Bradley, Cochair
Occidental College, Los Angeles, CA 90041

Martin S. Lawless, Cochair
Mechanical Engineering, The Cooper Union for the Advancement of Science and Art, 41 Cooper Sq., Rm. 720, New York, NY 10003

Invited Papers

1:30

4pAA1. Recital halls for K-12, higher education, and community music education: A trio of case studies. Benjamin E. Markham (Acentech, 33 Moulton St., Cambridge, MA 02138, bmarkham@acentech.com) and Jonah Sacks (Acentech, Cambridge, MA)

The recital halls at Deerfield Academy, UMass Boston, and the Brattleboro Music Center each play a critical role at the center of the music education programs at these institutions. Deerfield's 160-seat Elizabeth Wachsman Concert Hall, at the independent high school's Hess Center for the Arts, is both intimate and warmly resonant, and features adjustable curtains at the stage to suit varying repertoire for rehearsal and performance, particularly of Deerfield's student music ensembles. The 150-seat recital hall at UMass Boston's University Hall serves an even wider array of ensembles, from classical choral to jazz, in a building that not only serves the Performing Arts Department but also Chemistry and other general-purpose classrooms; the adjustable curtains there surround the audience on all sides. In Brattleboro, a 331-seat recital hall is the crown jewel of a new community music center that also serves as the home to a Chamber Music Series and a range of ensembles including Juno Orchestra, a concert choir, chorale, and camerata, and the music center's "Educate. Open. Strengthen." Program. These three rooms all serve critical pedagogical missions, but each with a different focus and a different set of needs and constraints that impact budget, architecture, and acoustical design considerations.

1:50

4pAA2. Modifying a recital hall to extend its range of use: A case study. Jonah Sacks (Acentech, 33 Moulton St., Cambridge, MA 02138, jsacks@acentech.com) and Benjamin E. Markham (Acentech, Cambridge, MA)

Rosen Concert Hall, in the Broyhill Music Center at the Appalachian State University Hayes School of Music, is a beloved 400-seat recital hall designed in the 1980s by architect Dennis Yates and acoustician Rein Pirn (Acentech/BBN) and was featured in the 1990 ASA publication "Acoustical Design of Music Education Facilities." The hall was designed for pipe organ, vocal chorus, and smaller classical instrumental ensembles, with significant variable absorption in the form of curtains. Since then, the school's program has broadened, and the hall is now used also for jazz band, large wind ensemble, and some amplified forms. A 2017 study included acoustical measurements in the hall, listening sessions with eight different ensembles in the hall, and discussions with faculty and staff. These resulted in detailed recommendations for acoustical improvements and new audiovisual equipment. The school has completed two recommended acoustical improvements, with positive results: hinged absorptive wall panels surrounding the platform, and extended reflective canopy. Users report improved on-platform clarity for louder ensembles, and better self-hearing and presence of sound for downstage performance locations.

2:10

4pAA3. My experience at threshold and the consultation process. Angel Castañon (Ventura College, 4667 Telegraph Rd., Ventura, CA 93015, angelc0428@gmail.com)

I had the great opportunity to take part in the Acoustical Society of America's (ASA) 2021Summer Undergraduate Research or Internship Experience in Acoustics (SURIEA) Program. The program was designed for underrepresented minority groups in order to introduce them to the field of acoustics. After a two-week crash course I was able to experience what it was like working in an

architectural acoustic consultation firm and learn the extensive consultation process at Threshold. Not only that, but learning some of the programs, the methodology, skills, and purpose of Threshold as well. The impact of the program has certainly opened my eyes to the field of architectural acoustics, and how much work has to go into a project for the best possible outcome. In essence, I found the experience to be one of the most impactful I had in my life.

2:30

4pAA4. Too loud? Sound absorption and volume as related to music educators' subjective response to rehearsal room loudness. Arjun K. Shankar (McKay Conant Hoover, Inc., 4950 E Van Buren St., Apt. 179, Phoenix, AZ 85008, arjunkshankar@gmail.com), David A. Conant, and K. Anthony Hoover (McKay Conant Hoover, Westlake Village, CA)

Controlling loudness is important for successful music rehearsal rooms. Volume, sound absorption, and often shaping strongly influence music faculty acceptance regarding room loudness. We assessed various rehearsal rooms, as well as data collected by Ron McKay in his BBN days, regarding the sound absorption, volume, and educators' subjective responses to loudness within the rooms, in an attempt to better understand correlations among these parameters.

Contributed Papers

2:50

4pAA5. Auralization of a flexible recital hall. Efrain Avendano-Gutierrez (Music, UCSD, 382 Vance St., Apt. B, Chula Vista, CA 91910, efrainag0171@gmail.com), Kelsey Rogers, and Benjamin E. Markham (Acentech, Cambridge, MA)

Auralizations have proven to be a powerful tool when it comes to the design phase of a room. They allow consultants to analyze and convey the acoustical challenges to make decisions accordingly. For this presentation, we look at the process of creating auralizations for a multipurpose recital hall, and how the different configurations of the room affect it acoustically. The recital hall will need to accommodate a large variety of rehearsals and performances, ranging from solo musicians to large combined ensembles. Therefore the balance between stage, audience, and ensemble must be kept in mind, spatially and acoustically. This project and presentation is thanks to the SURIEA program; an intensive summer program that emphasizes training, mentoring, research, and preparing students for graduate studies and careers in acoustics. As part of the program, students were paired with mentors to work on a project, and present their results in the next ASA conference.

3:05

4pAA6. Toward a more versatile venue: The Hahn Recital Hall renovation at the Music Academy of the West, Montecito, CA. David A. Conant (McKay Conant Hoover, Inc., 5655 Lindero Canyon Rd., McKay Conant Hoover, Ste. 325, Westlake Village, CA 91362, dconant@mchinc.com)

The Music Academy of the West was founded in 1947 by German operatic soprano and lieder singer, Lotte Lehmann and Otto Klemperer and has evolved to be regularly lauded as the "Julliard of the West." Because their 1970s, 350-seat Recital Hall could no longer properly serve as a suitable recital hall or rehearsal hall, MCH was retained in 2005 to design a full overhaul to ensure the hall could wonderfully serve ensembles ranging from 2 or 3 musicians to the 70-piece Santa Barbara Symphony in rehearsal. It serves now as the home of Marilyn Horne's Vocal Institute. We present our design process that involved major reshaping, mitigating rail traffic 300 ft away, a new HVAC system, a sensitive re-do of the copious stage (including mitigating practice rooms below), designing novel recital screens, providing variable acoustics and an 11th hour effort to accommodate *"Met Live!"*.

Session 4pAB

Animal Bioacoustics, Noise, Underwater Acoustics, and Acoustical Oceanography: Acoustical Impacts and Monitoring Protocols Associated with Offshore Windfarms

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

Chair's Introduction—1:00

Invited Papers

1:05

4pAB1. Offshore wind turbines as an anthropogenic noise source. Michael Stocker (Ocean Conservation Res., P.O. Box 559, Lagunitas, CA 94938, mstocker@ocr.org)

The pivot from offshore oil to offshore wind has ushered in the advancement and development of offshore windfarms. While wind energy generation is an environmental improvement with regards to decreasing atmospheric carbon associated with fossil fuel energy generation, it does not come without environmental costs. Siting, construction, installation, operations, and maintenance are all accompanied by noise—from airborne and underwater sound transmission, to benthic substrate vibration and mechanical impulse propagation. These noises will be transforming the soundscapes of large areas of our Outer Continental Shelf habitats, with known and unknown impacts on marine and avian biota. This presentation will provide an overview of turbine and windfarm noise sources with a focus on the known and potential biological impacts.

1:25

4pAB2. National Oceanic and Atmospheric Administration (NOAA) and Bureau of Ocean Energy Management (BOEM) minimum recommendations for use of passive acoustic listening systems in offshore wind energy development monitoring and mitigation programs. Nicholas B. Sisson (Greater Atlantic Regional Fisheries Office, National Oceanic and Atmospheric Administration, 55 Great Republic Dr., Gloucester, MA 01930, nick.sisson@noaa.gov), Kyle Baker (Office of Renewable Energy Programs, Bureau of Ocean Energy Management, Sterling, VA), Jaclyn Daly (Office of Protected Resources, National Oceanic and Atmospheric Administration, Silver Spring, MD), Genevieve Davis (Northeast Fisheries Sci. Ctr., National Oceanic and Atmospheric Administration, Woods Hole, MA), Carter Esch (Office of Protected Resources, National Oceanic and Atmospheric Administration, Silver Spring, MD), Shane Guan (Div. of Environ. Sci., Bureau of Ocean Energy Management, Sterling, VA), Amy Scholik-Schlomer (Office of Protected Resources, National Oceanic and Atmospheric Administration, Silver Spring, MD), Erica Staaterman (Ctr. for Marine Acoust., Bureau of Ocean Energy Management, Sterling, VA), and Sofie Van Parijs (Northeast Fisheries Sci. Ctr., National Oceanic and Atmospheric Administration, Woods Hole, MA)

Offshore wind energy development is rapidly advancing in United States waters to meet state and federal renewable energy goals. With a diverse suite of endangered large whale species and a multitude of other protected marine species inhabiting these same waters, understanding the potential consequences of construction and operation activities is essential to advancing responsible offshore wind development. Passive acoustic monitoring (PAM) represents a newer technology that has become one of several methods of choice for monitoring trends in the soundscape, presence of species, mitigating risk, and monitoring potential behavioral and distributional changes resulting from offshore wind activities. Federal and State regulators, the offshore wind industry, and environmental advocates require detailed information on PAM capabilities and techniques needed to promote efficient, consistent, and meaningful data collection efforts on local and regional scales. We provide capabilities and suggested applications of archival and real-time PAM systems, PAM study design considerations, and data management needs. We also provide key considerations for long-term baseline monitoring and vessel strike risk reduction using PAM. These recommendations provide an initial guide for stakeholders seeking to use PAM systems associated with the rapid expansion of offshore wind development in the United States.

1:45

4pAB3. Acoustic impact studies and assessments by Bureau of Ocean Energy Management on offshore wind development. Shane Guan (Div. of Environ. Sci., Bureau of Ocean Energy Management, 1315 East-West Hwy., Silver Spring, MD 20910, guan@cua.edu) and Samuel L. Denes (Ctr. for Marine Acoust., Bureau of Ocean Energy Management, Sterling, VA)

The U.S. Bureau of Ocean Energy Management (BOEM) manages the exploration and development of offshore energy resources, including offshore wind development, on the U.S. Outer Continental Shelf. Construction, operation, and decommissioning of offshore wind facilities generate intense or long-lasting underwater sounds and vibrations that may be detrimental to marine life. This presentation will provide an overview of the activities that BOEM is in engaging in to address these impacts. BOEM's Environmental Studies Program (ESP) has been developing funding and managing numerous studies to understand sound field characteristics and sound

propagation from pile driving and geophysical surveys related to offshore wind development, as well as marine species responses to these activities. In addition, BOEM established the Center for Marine Acoustics (CMA) to strengthen its role as a driving force within the regulatory community on sound in the marine environment and provides acoustical expertise within the environmental permitting process. The CMA continues to provide guidance on the development of best practices for the use of acoustics in the evaluating the effects of offshore wind on marine fauna. This presentation will highlight the relevant acoustic studies and explain how BOEM is using the science to better manage noise impacts on the ocean environment.

2:05

4pAB4. Stevens Passive Acoustic Detection System (SPADES -2) and its prospective application for windfarm underwater noise assessment. Alexander Sedunov (STAR Ctr., Stevens Inst. of Technol., One Castle Point on Hudson, Hoboken, NJ 07030, asedunov@research.stevens.edu), Hady Salloum, Nikolay Sedunov, Christopher Francis, Sergey Tsyuryupa, Aleksandr Merzhevskiy, Daniel Kadyrov, and Alexander Sutin (STAR Ctr., Stevens Inst. of Technol., Hoboken, NJ)

Stevens Institute of Technology has been conducting development and field tests of various underwater passive acoustic systems for several years. Several such systems provided localization of boats and divers triangulation. The new version of SPADES has a tethered low-cost bottom-mounted circular 2.2-m underwater acoustic array with eight custom-built hydrophones. The cost of the array was significantly reduced by manufacturing the hydrophones in-house and utilizing a lightweight and low-cost tether. The tether can provide power and communication up to 1 km away and power to the data acquisition. The software has been developed for real-time direction-finding using Steered Power Response Phase Transform (SRP-PHAT) method, combined with region-zeroing (RZ) approach to multi-source separation and custom noise background estimation subtraction. The array was tested for seven months in the shallow and busy waters of the Hudson River tracking small boat activity. The system's reliability and long tether make it attractive for long-term observation of underwater noise such as monitoring wind farm noise marine mammals and shipping traffic. Direction-finding can help identify noise unrelated to wind farms.

2:25

4pAB5. Operational underwater sound from future offshore wind turbines can affect the behavior of marine mammals. Frank Thomsen (DHI, Agern Allee 5, Hørsholm 2970, Denmark, frth@dhigroup.com) and Uwe Stöber (DHI, Bremen, Germany)

Offshore wind farms are part of the transition to renewable sources of energy and both the total numbers and size of wind turbines are rapidly increasing. While the impact of underwater sound related to construction work has been in the focus of research and regulation, few data exist on the potential impact of underwater sound from operational wind farms. Here, we reviewed published sound levels of underwater sound from operational wind farms and found an increase with size of wind turbines expressed in terms of their nominal power. This trend was identified in both broadband and turbine-specific spectral band sound pressure levels (SPLs). For a nominal power of 10 MW, the trends in broadband SPLs and turbine-specific spectral band SPLs yielded source levels of 170 and 177 dB *re* 1 μPa m, respectively. The shift from using gear boxes to direct drive technology is expected to reduce the sound level by 10 dB. Using the U.S. regulatory criterion for behavioral disruption for continuous noise (i.e., level B), a single 10MW direct drive turbine is expected to cause behavioral response in marine mammals up to 1.4 km distance from the turbine, compared to 6.3 km for a turbine with gear box.

2:45–3:00 Break

3:00

4pAB6. Passive acoustic monitoring during the construction of the Coastal Virginia Offshore Wind project. Ying-Tsong Lin (Woods Hole Oceanographic Inst., 266 Woods Hole Rd., Woods Hole, MA 02543, ytlin@whoi.edu), Arthur E. Newhall (Woods Hole Oceanographic Inst., Woods Hole, MA), James H. Miller, Gopu R. Potty (Ocean Eng., Univ. of Rhode Island, Narragansett, RI), Jennifer L. Amaral, Adam S. Frankel (Marine Acoust., Inc., Middletown, RI), Tim Mason (Subacoustech Environ. Ltd., Southampton, United Kingdom), Kristen Ampela (HDR, Inc., San Diego, CA), and Anwar A. Khan (HDR, Inc., Fort Lauderdale, FL)

A suite of hydrophone arrays was deployed to monitor pile driving sound and seafloor particle motion during construction of two wind turbine towers in the Coastal Virginia Offshore Wind (CVOW) project in May 2020. The primary objective of this passive acoustic monitoring (PAM) work was to assess the effectiveness of bubble curtains for underwater noise mitigation. The arrangement of the PAM tracks also enabled studies of azimuthal and range dependencies of pile driving sound propagation. Measurements of water temperature, salinity and surface heights were made during the monitoring period to assess underwater sound propagation conditions. PAM data showed that the bubble curtain effectively reduced the pile driving noise above 200 Hz, and a significant azimuthal dependency was observed. Statistical analyses of pile driving noise will be presented, along with recommendations for future wind farm construction monitoring, especially on marine mammal acoustic monitoring during construction. [Work supported by the Bureau of Ocean Energy Management]

3:20

4pAB7. Analysis of underwater sounds from impact pile driving at the Block Island Wind Farm. Jennifer L. Amaral (Marine Acoust., Inc., 2 Corporate Pl., Ste. 105, Middletown, RI 02842, jennifer.amaral@marineacoustics.com), James H. Miller, Gopu R. Potty (Ocean Eng., Univ. of Rhode Island, Narragansett, RI), Kathleen J. Vigness-Raposa (INSPIRE Environ., Middletown, RI), Ying-Tsong Lin (Woods Hole Oceanographic Inst., Woods Hole, MA), Adam S. Frankel (Marine Acoust., Inc., Middletown, RI), Arthur E. Newhall (Woods Hole Oceanographic Inst., Woods Hole, MA), Daniel R. Wilkes, and Alexander N. Gavrilov (Ctr. for Marine Sci. and Technol., School of Earth and Planetary Sci., Curtin Univ., Perth, Western Australia, Australia)

Impact pile driving creates intense, impulsive sound that radiates into the surrounding environment. Piles driven vertically into the seabed generate an azimuthally symmetric underwater sound field whereas piles driven on an angle will generate an azimuthally dependent sound field. Measurements were made during impact pile driving of raked piles to secure jacket foundation structures to the seabed at the Block Island Wind Farm at ranges between 500 m and 15 km. These measurements were analyzed to investigate variations in rise

time, decay time, pulse duration, kurtosis, and sound received levels as a function of range and azimuth. Variations in the radiated sound field along opposing azimuths resulted in differences in measured sound exposure levels of up to 10 dB and greater due to the pile rake as the sound propagated in range. The raked pile configuration was modeled using an equivalent axisymmetric FEM model to describe the azimuthally dependent measured sound fields and compared to the measured data. These measurements made during wind farm construction will be presented and discussed. [Work supported by the Bureau of Ocean Energy Management.]

3:40

4pAB8. Research priorities for sound and vibration effects on fishes and aquatic invertebrates from offshore wind energy development. Frank Thomsen (DHI, Agern Allee 5, Hørsholm 2970, Denmark, frth@dhigroup.com), Arthur N. Popper (Univ. of Maryland, College Park, MD), Kathryn Williams (Biodiversity Res. Inst., Portland, ME), Lyndie Hice-Dunton (Responsible Offshore Sci. Alliance, Washington, DC), Edward Jenkins (Biodiversity Res. Inst., Portland, ME), Dennis M. Higgs (Univ. of Windsor, Windsor, ON, Canada), Justin Krebs (AKRF, Hanover, MD), Aran Mooney (Woods Hole Oceanographic Inst., Woods Hole, MA), Aaron Rice, Louise Roberts (Cornell Univ., Ithaca, NY), Kathy Vigness-Raposa (INSPIRE, Newport, RI), and David Zeddies (JASCO, Silver Spring, MD)

There are substantial knowledge gaps regarding both the bioacoustics and the responses of animals to sounds associated with preconstruction, construction, and operations of offshore wind (OSW) energy development. A workgroup of the 2020 State of the Science Workshop on Wildlife and Offshore Wind Energy recommended priority studies for the next five years to help stakeholders better understand potential cumulative biological impacts of sound and vibration to fishes and aquatic invertebrates as the OSW industry develops. The workgroup identified seven short-term priorities that include a mix of primary research and coordination efforts. Key research needs include the examination of animal displacement and other behavioral responses to sound, as well as hearing sensitivity studies related to particle motion, substrate vibration, and sound pressure. Other needs include: identification of priority taxa on which to focus research; standardization of methods; development of a long-term highly instrumented field site; and examination of sound mitigation options for fishes and aquatic invertebrates. Effective assessment of potential cumulative impacts of sound and vibration on fishes and aquatic invertebrates is currently precluded by these and other knowledge gaps. Filling critical gaps in knowledge will improve our understanding of possible sound-related impacts of OSW energy development to populations and ecosystems.

4:00–4:30
Panel Discussion

THURSDAY AFTERNOON, 26 MAY 2022 GOVERNORS SQUARE 14, 1:20 P.M. TO 3:20 P.M.

Session 4pAO

Acoustical Oceanography: Topics in Acoustical Oceanography

Christopher Bassett, Cochair
Applied Physics Laboratory, University of Washington, 1013 NE 40th St., Seattle, NJ 98105

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

Contributed Papers

1:20

4pAO1. Weathering the storm: Impacts of hurricane-induced noise on the probability of detecting cetaceans. Aditi Tripathy (School of Marine Sci. and Ocean Eng., Univ. of New Hampshire, Durham, NH 03824, atripathy@ccom.unh.edu), Jennifer Miksis-Olds (Ctr. for Acoust. Res. and Education, Univ. of New Hampshire, Durham, NH), and Anthony P. Lyons (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, Durham, NH)

Hurricanes Dorian (Category 5; 2019), Florence (Category 4; 2018), and Humberto (Category 3; 2019) impacted the soundscape as observed at the Atlantic Deepwater Ecosystem Observatory Network (ADEON) locations in the US Mid- and South Atlantic Outer Continental Shelf. Passive acoustic data obtained from bottom-mounted hydrophones at the ADEON locations were examined to assess changes in ambient sound before, during, and after hurricane presence. Ambient sound level increased up to 25 dB in the 0.1–7 kHz band which could impact the detectability of cetaceans vocalizing at those frequencies. The probability of detection of fin whales, minke whales, and pilot whales was estimated using empirical ambient sound levels, modelled propagation loss, and pre-defined detection parameters. Detection area, as defined by range of minimum detectability of the cetacean from the receiver, was estimated before, during, and after hurricane presence at each ADEON location. Detection area changed considerably during hurricane presence with site-specific impacts for each of the cetaceans, which

may affect estimates of their abundance from passive acoustic recordings. [Study concept, oversight, and funding were provided by ONR Award N00014-16-1-2594 and BOEM under Contract No. M16PC00003, in partnership with ONR and NOAA. Funding for ship time was provided under separate contracts by ONR, Code 32.]

1:35

4pAO2. Sound field variability and mode coupling in the presence of internal Kelvin waves in Lake Kinneret, Israel. Ernst Uzhansky (Marine Geosciences, Univ. of Haifa, Haifa, Israel, Israel, Israel), Andrey Lun'kov (Wave Res. Ctr., General Phys. Inst., Moscow, Russian Federation), and Boris Katsnelson (Marine Geosciences, Univ. of Haifa, 199 Adda Khouchy Ave. Haifa 3498838, Israel, bkatsnels@univ.haifa.ac.il)

Spatiotemporal variability of the low- and mid-frequency sound field in the presence of internal Kelvin waves (IKWs) was studied theoretically and in experiment in Lake Kinneret, Israel. The experiments were done in 2021, where sound field was being recorded during two days (two periods of IKW) using two synchronized vertical line arrays (VLAs) of ten hydrophones each and 10 m between them, deployed in the center of the lake (depth ~41 m). Wideband linearly frequency modulated sound signals (300 Hz–7 kHz) were transmitted from the source deployed near the shore (depth ~11 m, distance 5.5 km from VLA). IKWs were registered with thermistor strings together with CTD. Sound speed profile is characterized by the thermocline located between 15 and 20 m of the depth with sound speeds 1510 and 1480 m/s above and below the thermocline, respectively. The sound propagation modeling was done using a Parabolic Equation and Normal Modes approaches, considering the real parameters of the gas-saturated bottom and the bathymetry. Results show strong mode coupling along the acoustic track, initiating ~12 dB variability of the sound intensity. Results of modeling are verified in the experiment. [This work was supported by RFBR, grant 20-55-S52005.]

1:50

4pAO3. Quantifying ship noise bias in fish abundance estimates in the Great Lakes. Andrew Barnard (Acoust., Penn State, Penn State University, State College, PA 16801, barnard@psu.edu), Miles Penhale (Keweenaw Res. Ctr., Michigan Technol. Univ., Houghton, MI), Steven Senczyszyn, Erik Kocher, Jason Swain (Great Lakes Res. Ctr., Michigan Technol. Univ., Houghton, MI), and Peter Esselman (Great Lakes Sci. Ctr., U.S. Geologic Survey, Ann Arbor, MI)

Prey fish abundances in the Great Lakes are a driver for several agencies commitments to the Council of Lake Committees to support fisheries management. These management decisions have profound economic and social impacts within the Great Lakes region. Fisheries estimates done by echosounders or trawling may be biased due to the propagated noise from large fisheries vessels. In the first year of a four year collaborative study, crewed fisheries vessels and uncrewed Saildrone vessels were used to compare abundance estimates between "loud" and "quiet" vessels. In order to quantify the effects of ship noise on prey fish abundance estimates, a mobile ship noise measurement system was designed and deployed to measure radiated acoustic signatures of several ships in the Great Lakes. This talk will discuss the deployment of a mobile underwater acoustic test range, and show initial results of ship noise measurements from the first year of the program. In addition, an overview of the echosounder abundance results will be given along with plans for future data generation, analysis, and comparison between crewed and autonomous systems.

2:05

4pAO4. Spatio-temporal contrastive learning for acoustic data. Yawen Zhang (Dept. of Comput. Sci., Univ. of Colorado Boulder, 1111 Eng. Dr., Boulder, CO, Boulder, CO 80309, yawen.zhang@colorado.edu), Carrie Wall (Cooperative Inst. for Res. in Environ. Sci., Univ. of Colorado Boulder, Boulder, CO), J Michael Jech (NEFSC, Woods Hole, MA), and Qin Lv (Dept. of Comput. Sci., Univ. of Colorado Boulder, Boulder, CO)

Large volumes of water column sonar data have been generated from acoustic surveys of living marine resources. They provide valuable information about marine ecosystems. To leverage them for acoustic target identification, scarcity of annotations is usually an issue. While some annotations have been generated by scientists via manual scrutiny or limited automation, they are limited to certain species. To fill in this gap, we propose a spatio-temporal contrastive learning approach for acoustic data. This unsupervised deep learning technique leverages both acoustic and non-acoustic (spatial and temporal) information for acoustic target classification. We firstly employ the Simple Linear Iterative Clustering (SLIC) algorithm to extract superpixels from acoustic data. Then the spatial and temporal similarity between two superpixels is computed, and the proposed spatio-temporal contrastive learning approach is applied to learn a semantically meaningful representation for superpixels. Finally, the learned representations of superpixels are used in acoustic target classification. We demonstrate that this approach outperforms previous methods with acoustic data collected by the NOAA National Marine Fisheries Service in the Gulf of Maine and Georges Bank region over 10 years. This dataset is archived at NOAA National Centers for Environmental Information and accessed for free through AWS and NOAA Big Data Program.

2:20

4pAO5. The effects of array design on acoustic data collected during marine seismic reflection surveys. Alexander S. Douglass (Oceanogr., Univ. of Washington, Marine Sci. Bldg., Rm 206, Seattle, WA 98133, asd21@uw.edu), Warren T. Wood, Benjamin J. Phrampus (U.S. Naval Res. Lab., Hancock County, MS), and Shima Abadi (Univ. of Washington, Seattle, WA)

Marine seismic reflection surveys provide an abundance of acoustic data that are potentially useful for an array of analyses within and in addition to geoacoustic studies. A single cruise typically produces thousands to tens of thousands of acoustic events, with hundreds of hydrophones recording each event over towed arrays that can span up to 15 km or more. However, the structure of the airgun source arrays and the receiver arrays is typically not obvious from the data alone and the pre-processing of the data may yield misleading results if not properly accounted for. Generally, the acoustic source consists of an array of airguns configured such that the acoustic energy is focused towards the ocean floor, significantly impacting the relative intensity of the direct and reflected paths. Additionally, each array channel output typically consists of multiple hydrophone outputs, which are not individually available, averaged to generate a single channel output. Understanding the impact of these constructions is crucial for many analyses, such as mitigation of airgun pulse impacts on marine mammals. Here, we quantify the impact that these factors have on the output data and provide an analysis of this influence for an experimental case. [Work supported by ONR.]

2:35

4pAO6. Modal dispersion curve extraction from shipping noise at two synchronized vertical arrays. Marina Yarina (Marine Geoscience, Univ. of Haifa, Haifa, Israel), Andrey Lun'kov (Wave Res. Ctr., General Phys. Inst., Moscow, Russian Federation), Oleg A. Godin (Phys., Naval Postgrad. School, Monterey, CA), and Boris Katsnelson (Marine Geosciences, Univ. of Haifa, 199 Adda Khouchy Ave., Haifa 3498838, Israel, bkatsnels@univ.haifa.ac.il)

The results of modal dispersion analysis, using low-frequency (20 to 250 Hz) shipping noise in shallow water with a soft bottom is presented. Experiments were carried out in the Sea of Galilee (Lake Kinneret, Israel) having a maximum depth of ~40 m and a gassy upper sedimentary layer. A receiving system combined two synchronized, 10-hydrophone vertical line arrays (VLAs) located at the center of the lake with distance Dr = 40 m between them. The noise source, R/V "Hermona," was moving along the straight line joining VLAs, up to 1 km from VLAs. A method of obtaining modal dispersion curves from shipping noise is proposed. For each frequency ω and variable horizontal wavenumber $q=\omega/c_{ph}$ the noise is spatially filtered at each VLA using calculated solution of the wave equation with the measured sound speed profile in water and the pressure-release condition imposed at the water surface. Then, the ratio of complex modal amplitudes at VLAs is calculated and multiplied by a factor of . The real part of the resulting 2-D structure exhibits the modal dispersion curves, which are used for acoustic characterization of the seabed. [Work supported by ISF, grant 946/20, and RFBR, grant 20-05-00119.]

4p THU. PM

2:50

4pAO7. Geoacoustic inversion for gassy sediment parameters using reflection and scattering of acoustic signals. Boris Katsnelson (Marine Geosciences, Univ. of Haifa, 199 Adda Khouchy Ave., Haifa 3498838, Israel, bkatsnels@univ.haifa.ac.il), Ernst Uzhansky, Regina Katsman (Marine Geosciences, Univ. of Haifa, Haifa, Israel), Andrey Lun'kov (Wave Res. Ctr., General Phys. Inst., Moscow, Russian Federation), and Anatoliy N. Ivakin (Univ. of Washington, Seattle, WA)

Results of experimental study and theoretical modeling of acoustic propagation and reverberation in Lake Kinneret (Israel) with gassy sediments are presented. The presence of methane bubbles in sediments significantly influences reflection and scattering of sound signals from the bottom, which in turn allows estimating properties of sediment using acoustic sensing. Experiments were carried out using R/V Hermona with sound source located at a 7 m depth radiating 1 s-long 0.3–7 kHz LFM sweeps with intervals from 1 to 20 s. Acoustic pressure time series were received on a single hydrophone at $\sim$1 m from the source and two vertical arrays fixed in the lake center at either 40 m or 10 m from each other. Analysis of both monostatic and bistatic experimental data (including long range propagation) was aimed to estimate acoustic characteristics of the bottom and then to infer the related gas content and its spatial and temporal variability. The sound speed in sediments at different locations (with different depth of water layer, maximal depth 40 m) was estimated to be $\sim$170–250 m/s that corresponds to gas volume concentrations $\sim$1%, which is in accordance with direct measurements made using non-acoustic probes. [Work was supported by BSF grant 2018150.]

3:05

4pAO8. Tools for climate solutions: Developing techniques for marine carbon dioxide removal measurement, reporting and validation. Simon Freeman (Dept. of Energy, ARPA-E, 50 Lambie Circle, Portsmouth, RI 02871, simon.freeman@hq.doe.gov)

Climate change impacts and mitigation strategies will define our interaction with the oceans this century. Marine carbon sequestration could facilitate the enormous scaling necessary for gigaton-level carbon dioxide (CO_2) removal: at least 10 GT/y by 2050 and 20 GT/y by 2100, required just to limit anthropogenic warming to below 2°C. Many proposed marine CO2 removal techniques involve the distributed capture of carbon, i.e., accelerating the biological carbon pump (e.g., iron ocean fertilization or artificial upwelling) or shifting the dissolution equilibrium of CO_2 (e.g., ocean alkalinity enhancement). However, technology that enables rapid, inexpensive, persistent and accurate measurement and validation of drawn-down CO_2 at the sequestration time- and regional ocean spatial-scales necessary to quantify carbon capture does not exist today. The accuracy and wholeness of these future techniques will be important for assigning financial value to marine CO_2 removal processes in a carbon market, as well as enabling thorough evaluation of environmental impacts and a comprehensive understanding of ocean carbon dynamics. I will discuss ARPA-E's interest in carbon sensing approaches, including passive and active acoustic techniques, which could rapidly quantify ocean carbon flux at scale and introduce powerful tools to address the challenges of mitigating climate impacts at sea.

Session 4pBA

Biomedical Acoustics, Education in Acoustics and Physical Acoustics: Ultrasound and Its Role in Powering, Sensing, and Communicating for Medical and Non-Medical Applications

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Inder Raj S. Makin, Cochair
SOMA, A.T. Still University, 5850 E. Still Circle, Mesa, AZ 85206

Chair's Introduction—1:30

Invited Papers

1:35

4pBA1. Ultrasound feasibility study for in body communication using OFDM modulation and small scale transducers. Thomas Bos (Elec. Eng., KU Leuven, Kasteelpark Arenberg 10, Box 2443, Leuven B-3001, Belgium, thomas.bos@esat.kuleuven.be), Jan D'hooge (Cardiovascular Sci., KU Leuven, Leuven, Belgium), Marian Verhelst, and Wim Dehaene (Elec. Eng., KU Leuven, Leuven, Belgium)

Ultrasound waves pose a promising alternative to the commonly used electromagnetic waves for in-body communication. This due to the lower ultrasound wave attenuation, the reduced health risks and the reduced external interference. These properties might realize a more reliable, low-energy, secure and high throughput in-body communication link. Current state-of-the-art ultrasound designs, however, are limited in their practical in-body deployment and reliability. This stems from their use of bulky, focused transducers, the use of simple modulation schemes or the absence of a realistic test environment and corresponding realistic channel models. This talk focuses on enabling communication with small-scale (2mm size) and omni-directional transducers (1.2 MHz center frequency). First, we will address the specific channel characteristics of such communication links. Second, to cope with these channel characteristics, we propose an OFDM modulation scheme, optimized for ultrasound in-body communication scenarios. Third, we will discuss the performance of a device optimized to assess this communication link.

1:55

4pBA2. A review of industrial ultrasonic power and communication systems and lessons towards medical applications. Kyle Wilt (Elec., Comput., and Systems Eng., Rensselaer Polytechnic Inst., JEC 6027, 110 8th St., Troy, NY 12180, wiltk2@rpi.edu)

The use of ultrasound in producing wireless powering and communication links, primarily used to transmit through metallic walls as a form of penetration free feedthrough, has been of significant interest recently. These ultrasonic feedthroughs enable sensing in environments which are otherwise inaccessible or dangerous to instrument, such as the interiors of nuclear containment vessels or the external hulls of submersibles. A review of these types of systems will be presented which summarizes the technologies and capabilities developed both at Rensselaer, with support of industry sponsors, and by others; considering both low-rate and high-rate sensing with power delivery support. Parallels and differences will be explored between the systems employed to provide such connections across or along rigid structures and applications involving communication with biologically implanted devices, generally focusing on the challenges present in producing effective ultrasonic channels with respect to both power transmission efficiency and data link bandwidth characteristics.

2:15

4pBA3. Ultrasonic implantable wireless system for deep-tissue oxygenation monitoring. Soner Sonmezoglu (Elec. Eng. and Comput. Sci., Univ. of California, Berkeley, Cory Hall, 468 Swarm Lab., Desk 12, UC Berkeley, Berkeley, CA 94720, ssonmezoglu@berkeley.edu)

Continuous real-time monitoring of physiological parameters can yield insights into the underlying aspects of many diseases and guide diagnostic and therapeutic decisions in surgeries and for critical care patients. Tissue oxygenation is one of the key physiological parameters and a critical determinant of organ function. Existing systems for monitoring deep-tissue oxygenation are limited by a few factors, including the need for wired connections, the inability to provide real-time data or operation restricted to surface tissues. We demonstrate the first minimally invasive ultrasonic wireless system to monitor deep-tissue O_2 that avoids all these drawbacks. The system is composed of a millimeter-sized O_2 sensor implant that communicates bi-directionally and digitally with an external transceiver outside the body, enabling deep-tissue O_2 monitoring for surgical or critical care indications. The implant is based on integrated circuit and microsystems technologies that enable extreme miniaturization with advanced performance. Furthermore, the system includes a

novel wireless power/data transmission strategy that enables the operation of the implant at centimeter-scale depths (up to 10 cm) in tissue. Overall, this technology represents a new class of monitoring and diagnostic system particularly suitable for organ monitoring, as well as other surgical or critical care situations.

2:35

4pBA4. High data rate ultrasonic communications. Andrew C. Singer (Elec. and Comput. Eng., Univ. of Illinois, 110 Coordinated Sci. Lab., 1308 West Main St., Urbana, IL 61801, acsinger@illinois.edu), Gizem Tabak (Univ. of Illinois at Urbana-Champaign, Urbana, IL), and Michael L. Oelze (ECE, Univ. of Illinois at Urbana-Champaign, Urbana, IL)

Ultrasonic signals have been used in biomedical applications since at least the 1940s. Underwater acoustic signals have been used in the subsea industry for ranging since the early 1900s and for SONAR imaging and communication in the decades that followed. Since the development of subsea communication systems in the 1940s, increasingly sophisticated means have be developed for transmitting information acoustically through the highly variable ocean environment. State-of-the-art acoustic communication systems leverage ultrasonic transducers to achieve data transmission rates that mirror radio-frequency (RF)-based WiFi systems on land. The increasing use of wireless implanted medical devices (IMDs) have increased interest in through-tissue communications. Restrictions on the available bandwidth and transmit signal power limit the data rates of RF-based IMDs. In this talk, we give an overview of video-capable ultrasonic wireless communication between IMDs and external devices. We emphasize the potential of such a system for improving video capsule endoscopy and describe the challenges of through-tissue acoustic communication. Building on experience with subsea-communication systems, we discuss methods to cope with attenuation, dispersion, reflection, and nonlinear propagation. Some of our recent experiments performed *ex vivo* with soft tissue samples and *in vivo* with anesthetized animal subjects will be discussed.

2:55

4pBA5. Toward an ultrasonic Internet of medical things. Tommaso Melodia (Northeastern Univ., 360 Huntington Ave., Boston, MA 02115, melodia@northeastern.edu), Raffaele Guida (Bionet Sonar, Boston, MA), Enrico Santagati (Northeastern Univ., Boston, MA, MA), Emrecan Demirors (Bionet Sonar, Boston, MA), Daniel Uvaydov, Pedram Johari (Northeastern Univ., Boston, MA), and Jorge Jimenez (Bionet Sonar, Atlanta, GA)

Wireless networked systems of “smart” miniaturized and electronically controlled implantable or wearable medical sensors and actuators will be the basis of many innovative and potentially revolutionary therapies and applications. The main obstacle in realizing this vision of smart networked implants is posed by the dielectric nature of the human body, which strongly attenuates radio-frequency electromagnetic waves used in traditional wireless technologies such as Bluetooth or WiFi. This talk will give an overview of our work exploring a radically different approach, i.e., establishing wireless networked systems in human tissues that transfer data and energy through acoustic waves at ultrasonic frequencies. We will start off by discussing applications of networked implantable medical systems. We will then analyze fundamental aspects of ultrasonic propagation in human tissues and their impact on the design of wireless networking protocols at different layers of the networking protocol stack. We will then review our work on designing and prototyping ultrasonically rechargeable and connected Internet-of-Things platforms through a closed-loop combination of mathematical modeling, simulation, and experimental evaluation.

3:15

4pBA6. Ultrasound water-marking for simultaneous B-scan imaging and bio-telemetry. Chaohui Li, Peter Yang (Washington Univ. in St. Louis, Saint Louis, MO), Sri Harsha Kondapalli (Mathworks, Inc., Santa Clara, CA), and Shantanu Chakrabartty (Washington Univ. in St. Louis, Elec. and Systems Eng., One Brookings Dr., Campus Box: 1042, Saint Louis, MO 63130, shantanu@wustl.edu)

Ultrasound imaging technology has undergone a revolution during the last decade due to the availability of transducers that can operate over a large range of frequencies, and also due to the availability of high-speed, high-resolution analog-to-digital converters and signal processors. The large data acquisition and computational bandwidth afforded on these portable and bench-top ultrasound imaging systems could potentially be leveraged for designing energy-efficient bio-telemetry links that can be used for communicating with devices implanted *in vivo*. In this paper, we discuss an ultrasound water-marking approach that can be used for simultaneous imaging and bio-telemetry. The trade-off involves supporting a reasonable data-rate bio-telemetry link and at the same time minimizing artifacts due to bio-telemetry data in the B-scan images. Specifically, we exploit a variance-based signal representation where the image background noise is modulated to encode the data to be transmitted or sensed. Using a B-scan ultrasound imaging system and a phantom setup we show that the approach can support biotelemetry links at sub-nanowatt transmission power and at depths greater than 10 cm.

3:35–3:50 Break

3:50

4pBA7. Coupling and maintaining link efficiency ultrasonics in transcutaneous energy transfer systems. Jeffrey Leadbetter, Hugo Vihvelin (Daxsonics Ultrasound, Halifax, NS, Canada), Akhilesh Kotiya (School of Biomedical Eng., Dalhousie Univ., Halifax, NS, Canada), Daniel Zaenker (Daxsonics Ultrasound, Halifax, NS, Canada), Jeremy Norman, Jeremy Brown (School of Biomedical Eng., Dalhousie Univiersity, Halifax, NS, Canada), and Robert B. Adamson (School of Biomedical Eng., Dalhousie Univiersity, PO Box 15000, Halifax, NS B3H4R2, Canada, rob.adamson@dal.ca)

We present on the development of a 12.8 mm diameter ultrasonic transcutaneous energy transfer system for powering implanted hearing devices. The system was based on two custom 8mm diameter PMN-PT 1-3 composite transducers operating at 1.25MHz and featured three significant innovations. First, to ensure long-term biocompatibility and reliability of the implanted portion of the link and to ensure robust attachment to bone, the implanted transducer was designed into a 12.8 mm diameter, 3.5 mm thick hermetically sealed titanium package in an easy-to-implant form factor. The transducer was designed to deliver sound efficiently through the casing walls using a mass-spring acoustic matching technique. Second, for the external unit a “dry” acoustic coupling system based on a silicone pressure sensitive adhesive was designed that combined efficient acoustic coupling, robust adhesion to skin and that was comfortable to

wear. The external unit was aligned with the internal unit using a magnetic alignment system. Third, we developed and demonstrated an efficient Class E transmit amplifier and receive electronics that incorporated compensation for changes in acoustic separation between the transducer to maintain robust, high efficiency power transmission. The system achieved 33% DC-to-DC electrical conversion efficiency through 5mm of water and 19% DC-to-DC efficiency through 5mm of porcine tissue.

4:10

4pBA8. Charging of devices for healthcare applications, using ultrasound wireless power. Inder Raj S. Makin (A.T. Still Univ., 5850 E Still Circle, Mesa, AZ 85206, imakin@atsu.edu), Harry Jabs (Piezo Energy Technol., LLC, Mesa, AZ), T. Douglas Mast (Univ. of Cincinnati, Cincinnati, OH), and Leon Radziemski (Piezo Energy Technol., LLC, Mesa, AZ)

Current wireless powering is frequently accomplished through inductive fields. An alternative for wireless powering systems is via ultrasound. Potential advantages are deeper penetration, delivery through metals, no electronic interference, and beam steerability to correct for misalignment. These characteristics enable ultrasound-based recharging of batteries located within a hermetically sealed enclosure, e.g., an implantable pulse generator (IPG). Another growth area for wireless power delivery is to support healthcare operations, when digital devices used for healthcare delivery need regular charging. Ultrasound Wireless Powering (UWP) is convenient, avoids cable clutter, and is safe. Planar ultrasound transducers operating between 1 and 3 MHz provide a practical compromise between useful directivity and effective propagation range. This talk will describe the development and evaluation of a UWP recharging system operating at 1 MHz transferring up to 100 mA at nominally 4 V, to charge a Li-Ion battery 10–20 mm through tissue. For potential neurostimulator applications, the receiver module was packaged within 1.2 cc inside a 17-cc titanium-IPG. For digital device charging with UWP, benchtop experiments will demonstrate characteristics to power smartphones comparable to the Qi-standard, version 1.0, which is an industry standard for characterizing charging with inductive fields. [Work partially supported by the NIH/NIBIB R43EB019225.]

4:30

4pBA9. Acoustic vibrations for versatile wireless power transfer. Victor Farm-Guoo Tseng (US Army Combat Capabilities Command-Army Res. Lab., 2800 Powder Mill Rd., Adelphi, MD 20783, vfgtseng@gmail.com), Joshua J. Radice (Mech. Eng., U.S. Naval Acad., Annapolis, MD), Nathan Lazarus, and Sarah Bedair (U.S. Army Combat Capabilities Command-Army Res. Lab., Adelphi, MD)

We present an overview of prior/recent work done at the US Army Combat Capabilities Command—Army Research Laboratory on using acoustic waves for wireless power transfer (WPT). In the past we have demonstrated through-air ultrasonic phased array power beaming, through-metal longer distance transfer using Lamb waves, and power/data transfer to embedded sensor nodes. Acoustic WPT has the advantage of not being shielded by metal, and can achieve higher efficiencies than inductive WPT at larger distances. Part of our recent work is on developing a through-metal acoustic WPT based wireless UAV recharging platform. For the first time, we have demonstrated a self-detachable acoustic WPT receiver, using the switchable magnetic force from electro-permanent-magnets to attach a receiver transducer onto a steel platform to receive power (a transmitter is bonded on the other side of the platform), without the need of any liquid couplant. The magnetic force enhances the mechanical mode coupling, while allowing the receiver to attach/detach freely from the platform. AC-AC power efficiency up to 63% was achieved, and with the appropriate power electronics and matching network, an overall DC-DC efficiency of 44% was achieved when charging a 3-cell LiPo UAV battery with 4.6 W, with higher power levels achievable.

Contributed Papers

4:50

4pBA10. Design of an electronic radiological clip for improved breast cancer imaging with ultrasound. Jenna Cario (ECE, Univ. of Illinois at Urbana-Champaign, 405 N. Mathews Ave., Urbana, IL 61801, jcario2@illinois.edu) and Michael L. Oelze (ECE, Univ. of Illinois at Urbana-Champaign, Urbana, IL)

Radiological clips are commonly used to track and identify lesions in breast cancer patients receiving neoadjuvant chemotherapy. Available radiological clips often suffer degradation of visibility in ultrasound during the course of treatment, thus requiring more involved localization procedures to identify regions prior to resection. Furthermore, upon localization, additional markers are placed to aid surgeons during resection. We describe a biocompatible electronic device which transmits a unique identification signal when imaged with ultrasound pulse inversion,*i.e.*, ultrasound identification (USID). The USID clip provides better localization and visualization of the clip compared to passive clips. Furthermore, the ultrasound systems can be programmed to produce an audible indicator of a specific clip's presence and proximity to the probe, thereby eliminating the need for insertion of additional markers. This concept is tested using a microcontroller and a direct digital synthesis board connected to two sonomicrocrystals, a Verasonics research system, and tissue mimicking phantoms. Clips with six different USID tags were successfully visualized at 10 dB above the background signal. Localization and identification are demonstrated using pulse inversion and image processing techniques.

5:05

4pBA11. Implementation of real-time high-speed ultrasound communications through tissue. Zhengchang Kou (Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, B420 Beckman Inst., 405 North Mathews Ave., Urbana, IL 61801, zkou2@illinois.edu) and Michael L. Oelze (ECE, Univ. of Illinois at Urbana-Champaign, Urbana, IL)

In this work, we propose a novel implementation of both a transmitter and receiver with field programmable gate arrays (FPGAs) to achieve real-time continuous high-definition (HD) video transmission through tissue, which can enable HD and higher frame rate wireless capsule endoscopy. We used a Texas Instruments AFE58JD48EVM 16 channel analog front end (AFE) evaluation board as the receiver connected to a Xilinx ZCU106 Zynq Ultrascale MPSoC development board in which we implemented a digital down converter (DDC), OFDM demodulator, maximum ratio combiner and low-density parity-check (LDPC) decoder. For the transmitter, we used an Analog Devices EVAL-AD9166 vector signal generator evaluation board, which has a built-in 4.3 dBm output power amplifier as a transmitter connected to another ZCU106 development board in which we implemented a LDPC encoder, OFDM modulator and digital up converter (DUC). The modulated signal was transmitted through a tissue-mimicking abdominal phantom using a 2-mm microcrystal transducer and received with a Sonic Concepts IP103 64 channel phased array at a center frequency of 3.2 MHz. We achieved the continuous transmission of up to over[OML1] 6 Mbps error free payload data rate after LDPC decoder which is used to carry HD video streams through ultrasound.

Session 4pEA

Engineering Acoustics, Education in Acoustics, Physical Acoustics, Noise, and Architectural Acoustics: Low Cost Acoustical Measurement Systems

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David A. Brown, Cochair
ATMC/ECE, University of Massachusetts Dartmouth, 151 Martine St., UMass Dartmouth, Fall River, MA 02723

Chair's Introduction—1:00

Invited Papers

1:05

4pEA1. Design of a scanning acoustic and photoacoustic microscopy system using open-source hardware and software components. Ronald E. Kumon (Kettering Univ., 23700 Maude Lea St., Novi, MI 48375, rkumon@kettering.edu), John T. Bonhomme, Corneliu I. Rablau, and Timothy A. Stiles (Kettering Univ., Flint, MI)

We have designed and started construction of an instrument that will be able to operate as both a scanning acoustic microscope and photoacoustic microscope. To keep costs down, we are using open-source hardware and software components wherever possible. The system is designed to scan specimens that are approximately 2 cm $\times$ 2 cm in lateral dimensions with lateral steps of 1 micron or less. When operating as a scanning acoustic microscope, the specimen will be water-coupled to a high-frequency ultrasound transducer operating in pulse-echo mode. When operating as a photoacoustic microscope, short light pulses infrared laser diode located under the specimen will generate ultrasound pulses thermoelastically, which will then be received by a confocal high-frequency transducer. In both cases, the specimen will be raster-scanned under the transducer by a moving stage. The mechanical scanning system was designed and built using a spring-loaded microscope stage, micrometers, stepper motors, a shield board used for 3D printers, an Arduino Mega microcontroller, and a Raspberry Pi 4 microcomputer. A graphical user interface has been written in Python using Tkinter to send the motion control commands to the stage. Future work will include incorporation of the laser and transducer control systems.

1:25

4pEA2. Low-cost engineering experimentation laboratory kit designed for at-home project-based learning. Martin S. Lawless (Mech. Eng., The Cooper Union for the Advancement of Sci. and Art, 6 Pennyfield Ave., S&E 2-38, Bronx, NY 10465, mlawless@sunymaritime.edu), Michael Giglia, and David Wootton (Mech. Eng., The Cooper Union for the Advancement of Sci. and Art, New York, NY)

During the Spring 2021 semester, the third-year Engineering Experimentation course at the Cooper Union was administered virtually due to the COVID-19 pandemic. The online format presented many challenges to teaching a hands-on, project-based laboratory class. To provide students with the experience of performing real engineering experiments, a low-cost laboratory kit was assembled and sent to each student. The kit included a microcontroller to serve as a data acquisition device, two electret microphones, and electronic components necessary to collect acoustic data. All of the students used this equipment to run a speed of sound laboratory by determining the time an impulsive signal took to travel between the microphones placed a known distance apart. For the final project, two teams of students conducted independent research of acoustic phenomena and further developed low-cost solutions. One team constructed an impedance tube from PVC pipe to measure the absorption characteristics of unknown materials. Normal incidence sound absorption coefficients were calculated for four frequencies by measuring the standing-wave ratio in accordance with ASTM standard C384. The second team built two experimental apparatuses to demonstrate the Doppler Effect. Both at-home Doppler experiments were able to clearly show the shift of the observed frequency of a moving source.

1:45

4pEA3. A remote lab exercise to introduce students to the fundamentals of bioacoustics using smartphones and R. Dena J. Clink (K. Lisa Yang Ctr. for Conservation Bioacoustics, Cornell Univ., 159 Sapsucker Woods Rd., Ithaca, NY 14850, djc426@cornell.edu), Isabel Comella, and Maryam Zafar (K. Lisa Yang Ctr. for Conservation Bioacoustics, Cornell Univ., Ithaca, NY)

Animals communicate through a variety of modalities including visual, auditory and chemical. Acoustic signals are particularly well suited for studying the evolution of animal communication due to the relative ease in which sounds can be recorded and analyzed. Therefore, a thorough overview of acoustic communication is a fundamental aspect of any animal behavior course. The purpose of this lab exercise is to introduce students to bioacoustics, or the study of animal sounds and their habitats. For the field component, students use their smartphones to collect focal recordings of target animals as well as collect acoustic data that will be used to investigate variation in soundscapes at different times (e.g. dawn and dusk) and/or different locations (e.g., urban versus rural). The computer lab component utilizes the R programming environment to import sound files and visualize differences in acoustic data using both spectrograms and principal component analysis biplots. The computer lab exercises are meant for undergraduate students with relatively little coding background but could be easily adapted for students of different skill levels. All materials needed to use and modify the course materials are openly available on GitHub, which should remove any financial barriers for students or instructors.

2:05

4pEA4. Comparing a low-cost bioacoustics-focused recording system with traditional acoustical measurement hardware. Levi T. Moats (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, lmoats359@gmail.com), Kelsey B. Moore, Mylan R. Cook (Brigham Young Univ., Provo, UT), Lucas K. Hall (Biology, California State Univ. Bakersfield, Bakersfield, CA), Kent L. Gee (Brigham Young Univ., Provo, UT), and Steven C. Campbell (Ball Aerosp., Beavercreek, OH)

Various types of low-cost acoustic recording equipment are available to the scientific community and the general public. While financial budgets can restrict what equipment can be purchased for a particular application, it is necessary to quantify the limitations of particular systems to determine their usefulness. Direct comparisons between recordings using low-cost equipment and using high fidelity equipment can highlight how the quality of low-cost recording is affected, which can help to determine whether or not budget recording systems are sufficient for a particular application. The Cornell Lab of Ornithology developed a low-cost terrestrial recording device for bioacoustic data collection known as the SwiftOne. This device can be set with various sampling rates and configurable recording schedules, making them suitable for long term data collection. Multiple SwiftOne units were used alongside a high-fidelity data acquisition system in both laboratory and outdoor data collection, including loudspeaker measurements in an anechoic chamber, ambient recordings at a bird refuge, and at a coastal rocket launch. The overall quality of the low-budget data recordings is investigated to determine the usefulness of the SwiftOne in these acoustic environments. Analyses include comparisons of measured waveforms, spectral content, and sound pressure levels.

2:25

4pEA5. Characterizing the complex shear modulus of materials with an iPhone. Gabriel R. Venegas (Ctr. for Acoust. Res. and Education, Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824, g.venegas@unh.edu), Aytahn Ben-avi, Christina Naify, Kevin M. Lee (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)

Dynamic mechanical analysis (DMA) is a technique that accurately measures the mechanical properties, such as stiffness and damping, of a material as a function of frequency, temperature, and time. However, DMA is expensive, requires significant training to operate, and must be performed on-site, lending itself impractical for hands-on classroom activities or research during the early stages of a global pandemic. To characterize the complex shear modulus of seagrass during the COVID-19 shelter-in-place order in May 2020, an iPhone XS Max was used as the tail mass of a freely oscillating torsional pendulum. The Vibrations application (Diffraction Limited Design, LLC) acquired the angular velocity of the underdamped system during oscillation, and the damped natural frequency and decay rate were later extracted from the time series. By measuring the cross-section and length of the material under test, the natural frequency and decay rate were used to determine the complex shear modulus. A strip of 1-cm-wide polyethylene film (McMaster Carr #85655K13) was first used to test the measurement technique. iPhone-obtained shear modulus of the strip was within the range of tabulated values, and was in agreement with DMA-obtained shear modulus measured after COVID-19 restrictions were relaxed. Uncertainty analysis and instructional applications will be discussed.

Contributed Papers

2:45

4pEA6. Affordable and customizable high-resolution scanning system for measurement of acoustic fields. Colby W. Cushing (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, colby.cushing@utexas.edu), Samuel D. Parker, Janghoon Kang (Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX), Gabriel R. Venegas (Ctr. for Acoust. Res. and Education, Univ. of New Hampshire, Durham, NH), Preston S. Wilson, and Michael R. Haberman (Appl. Res. Labs. and the Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX)

Spatial scans of acoustic fields are often required to characterize acoustic system behavior. Automated positioning systems allow for higher spatial resolution, more efficient use of time, and increased repeatability and precision during data collection than is possible manually. Unfortunately, commercial systems can be expensive, difficult to customize after purchase, and can be very cumbersome to manipulate. The present work describes an affordable and customizable scanning system that can be operated through MATLAB using an Arduino microcontroller and constructed with OpenBuilds hardware. We will outline the integration of an oscilloscope and function generator with the motion control system to enable a completely automated measurement system. Further, we will discuss potential pitfalls one may encounter when creating a similar system and highlight best practices that allow you to avoid those problems at the system design stage. This work will discuss similar systems created to characterize underwater acoustic metamaterials (doi:10.1121/10.0009161), quantify performance of an additively manufactured acoustic diffuser (doi:10.1121/1.5147148), define pressure field responses due to inclusion of an acoustic metasurface used for

acoustic holography (doi:10.26153/tsw/13999), and analyze acoustic properties of cylindrical sediment samples containing infauna (doi:10.1121/2.0000744). [Work sponsored by ARLUT, ONR, Sandia, and Samsung.]

3:00

4pEA7. Macriaphone: A Stargazer's microphone design. Alexey Titovich (Carderock Div., Naval Undersea Warfare Ctr., 9500 MacArthur Blvd., West Bethesda, MD 20817-5700, alexey.titovich@navy.mil) and Alex Titovich (Carderock Div., Naval Undersea Warfare Ctr., Poolesville, MD)

A highly directional, single channel microphone design is presented. The design utilizes a line of sight waveguide similar to a Keplerian reflector-type telescope. The key feature is that the waveguide is acoustically reflection-less thereby preventing off-axis reflected and diffracted fields from contaminating the measurement. The signal in the desired direction is amplified by a parabolic reflector, which terminates the waveguide and supports a single microphone at its focal point. The resulting directivity index is superior to that of a parabolic reflector microphone. Due to the line-of-sight cone provided by the waveguide, this device is particularly well suited for measuring distant sound, which is, why it is called the macriaphone (meaning far away sound in Greek). Theoretical background, numerical predictions, and experimental results from the prototype are presented.

3:15–3:30 Break

3:30

4pEA8. Measurements of electromechanical impedance to determine effects of pressure and temperature on underwater acoustic transducers. Eric K. Aikins (Elec. AND Comput., Univ. Massachusetts Dartmouth and Raytheon Technol., 61 Stephanie Pl., New Bedford, MA 02745, eaikins@umassd.edu) and David A. Brown (ECE, Univ. Massachusetts Dartmouth, Fall River, MA)

Underwater electroacoustic transducers can be subjected to large static pressures, wide range of temperature, and high operational electric fields causing internal heating. These conditions can adversely impact the performance of transducers including transmit output source level, reliability, survivability, useable bandwidth, tuning and matching to the amplifier, and efficiency. A low cost electroacoustic impedance measurement system was implemented with an air-pressurized mechanical transformer using SCUBA tanks to investigate transducer properties to 1,500 PSI. Subsequently, a water pressurized tank was used to 10 000 psi (or 7000 m of equivalent depth). A variety of underwater cylindrical transducers were tested and the changes in electromechanical properties (dielectric constant, capacitance, loss-tangent, mechanical compliance, piezoelectric d-constant, and effective coupling constant) were measured dynamically using the impedance methods. An analytical model was developed with a GUI to predict acoustic performance and tuning for different operational (depth, temperature, and drive level) conditions. Cylindrical transducers of oil-filled and polyurethane filled designs were shown to have moderate changes in electromechanical properties to pressures of 10 000 psi.

3:45

4pEA9. Low cost acoustic transducer calibration system. David A. Brown (Elec. Eng., Univ. of Massachusetts, Dartmouth, Fall River, MA), Corey Bachand, Austin Souza (BTech Acoust. LLC, Fall River, MA), Isabella Di Bona (Elec. Eng., Univ. of Massachusetts, Dartmouth, 151 Martine St., Ste. 123, Fall River, MA 02723, idibona@umassd.edu), and Jesse Kanaley (Elec. Eng., Univ. of Massachusetts, Dartmouth, Fall River, MA)

An acoustic calibration system was developed primarily for underwater acoustic transducers, although applicable for a variety of acoustic applications. Historically, such systems have been large, highly customized, and very expensive. With the advent of compact cost-effective digital PC-based oscilloscopes and data acquisitions systems (e.g., devices from TiePie, Cleverscope, and Digilent), a development that began as a senior capstone-design project has recently been demonstrated. Our system is based on the TiePie Digital Oscilloscope, a PC based signal generation and acquisition systems, a power amplifier, optional preamplifier and stepper motor with controller for radiation patterns. More specifically, the receive sensor signals are acquired with a TiePie Handyscope HS6 DIFF (1000MS/s, USB 3.0 oscilloscope, with four differential analogue channels) and the transmit signal (signal generation, transmit voltage and current acquisition) utilizes a TiePie HS5-540XM 5 (500 MHz, 1-channel generator, 2-channel acquisition) measuring system. The calibration and data display is controlled with MATLAB and controlled through a graphical user interface (GUI). Examples of Transmit Pressure Response per Volt (TR/V or TVR), Free Field Voltage Sensitivity (FFVS), and Beam Pattern measurements will be provided. [Work supported by BTech Acoustics LLC.]

4:00

4pEA10. Technical and electroacoustic aspects of miniature omnidirectional sound sources used in acoustic scale modeling. Bartlomiej Chojnacki (AGH Univ. of Sci. and Technol., Mickiewicza 30, Cracow 30-059, Poland, bchojnacki@agh.edu.pl) and Tadeusz Kamisinski (AGH Univ. of Sci. and Technol., Cracow, Poland)

Acoustic similarity theory allows us to create scaled room acoustic models which are simple to construct but usually difficult to measure correctly. One of the most common problems is the sound source, which works in the wide frequency range (800–40000 Hz, 500–63000 Hz) and has an omnidirectional sound propagation pattern. The lecture will present the novel approach for miniature sound sources design and creation for the most common measurement functions—laboratory objects and materials measurements (absorption, diffusion) in the scaled reverberation chamber, sound insulation scale modeling, and the room acoustic measurements in reduced models. Each of those applications has special sound source requirements, which allow us to differentiate their construction type. The most important aspects of the designing process were investigated, such as the number of transducers used and their placement, types of speakers, enclosure shape, and dimensions. The lecture will expose the sound sources used in Technical Acoustic Laboratory in AGH, Cracow, for the given measurement functions. The design recommendations developed based on FEM modeling, and experimental testing will be discussed regarding the existing objects, allowing future users to recreate in similar shape or adjust to other, predefined scale modeling measurement functions.

4:15

4pEA11. Quasi-anechoic frequency response measurement of underwater acoustic devices using exponential sine sweeps. Andrew Dominijanni (MITRE, The MITRE Corp., 202 Burlington Rd., Bedford, MA 01730, adominijanni@mitre.org) and Jeffrey Rowan (MITRE, Bedford, MA)

As part of the development of the MITRE BlueTech Lab, an existing $106 \times 40 \times 15$ ft. underground tank is being repurposed as an underwater acoustic test facility. Here, we report on the development of a procedure to extract anechoic frequency responses from measurements made in the otherwise reverberant environment using exponential sine sweeps. A demonstration of the procedure applied to a small-aperture baffled receiver array is shown, and results are compared to time-gated stepped sine wave measurements. Simple geometric rules to maximize measurement bandwidth while ensuring far-field conditions for both transmitter and receiver array are presented.

4:30

4pEA12. Abstract withdrawn.

4:45

4pEA13. Leveraging audio hardware for underwater acoustics. Thomas E. Blanford (The Penn State Univ., The Penn State Univ., State College, PA 16804, teb217@psu.edu), Luke Garrett, J. Daniel Park, and Daniel C. Brown (The Penn State Univ., State College, PA)

Both active and passive underwater acoustic systems frequently use purpose built hardware that is designed for a particular application. Transducers, electronics, and data acquisition systems for field experiments, therefore, are often expensive, tailored to limited frequency bands, or packaged for integration with a specific platform. Reliance on custom hardware, however, can make initial experimental investigations of new sensing paradigms cost prohibitive. The expense also creates an entry barrier to experimental work for students and researchers outside the underwater

acoustics community. Audio hardware, however, is widely available, easily integrated using commercial data acquisition tools, and is often of relatively low cost. This presentation will describe two efforts to use audio hardware for inexpensive experimental investigations of underwater acoustics topics. The first, AirSAS, uses audio transducers and electronics to investigate synthetic aperture sonar problems in air that are analogous to those underwater. The second, the Citizen Scientist Hydrophone, integrates underwater transducers with consumer audio electronics to make an inexpensive, multipurpose passive sensing device. Challenges and limitations of using commercial audio hardware for underwater acoustics, especially as they relate to accuracy and data quality, will be discussed.

5:00

4pEA14. Autonomous system for in-field observation of responses to vocalizations of Alston's singing mouse. Colby W. Cushing (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, colby.cushing@utexas.edu), Samantha K. Smith, John T. Clyde (Dept. of Integrative Biology, The Univ. of Texas at Austin, Austin, TX), Preston S. Wilson (Appl. Res. Labs. and The Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX), and Steven M. Phelps (Dept. of Integrative Biology, The Univ. of Texas at Austin, Austin, TX)

Ranging throughout central America, Alston's singing mice (*Scotinomys teguina*) are known for their long songs, which consist of rapidly repeated, frequency-modulated notes. Singing mice are the first rodents shown to exchange calls at rates resembling human conversation, which are used in mate attraction and male-male competition. Despite being researched in laboratory settings, few studies have examined the impact of the vocalizations in a natural environment. Unlike singing mice, most small rodents make high-frequency sounds at nighttime, to avoid attracting predators. We aim to observe why singing mice can produce long-distance, low-frequency songs, despite the predation risk. To do so, we developed a Raspberry Pi-based system to generate artificial mouse vocalizations at regularly scheduled intervals and record audio data of approaching predators and singing mice. This device was designed to be powered with a standard car battery and a backup lithium-ion battery to function for approximately two weeks in the field without interruption. An Adam A3X studio monitor was deconstructed and retrofitted with a USB microphone, two-terabyte hard drive, backup battery, and amplifier. This presentation will describe the construction of the device and issues with functionality addressed during the design process. [Work supported by UT Integrative Biology Dept., SWAN Howard-McCarley Award.]

5:15

4pEA15. Photoacoustic LED imaging apparatus using a low-cost system. Leah E. Burge (Phys. Dept., U.S. Naval Acad., Annapolis, MD 21402, leahburge02@gmail.com) and Murray S. Korman (Phys., U.S. Naval Acad., Annapolis, MD)

The use of compact and cost-effective visible light emitting diodes (LEDs) as an alternative to Q-switched lasers, the conventional photoacoustic excitation source, has been explored for vascular imaging applications [T. J. Allen and P. C. Beard, "Light emitting diodes as an excitation source for biomedical photo-acoustics," *Proc. SPIE* **8581**, 85811F (2013)]. This research motivated our work. Experiments used a 619 nm LED (Luminus SST-90-R-F11-HH100) pulsed for 450 ns with period 34 ms utilizing a pulse generator, MOSFET driver and power MOSFET switch, discharging a bank of capacitors (990 μF) with 36 A peak current. A pair of 2.5 cm diameter aspherical acrylic lens focused the LED—with an emitting area of 3 mm $\times$ 3 mm—onto the surface of a vertical capillary tube (1.19 mm ID, 1.75 mm OD) filled with 5% McCormick Green food coloring or 5% Higgins India Ink and placed in the 10 cc acrylic water tank. An Olympus emersion transducer (1.27 cm diameter circular plane array, 1 MHz with 0.4 MHz bandwidth) was amplified using four home-made low noise amplifiers (87 dB overall gain, 0.5–30 MHz bandwidth). An Agilent DSO 7014 B oscilloscope recorded the photoacoustic signal pulse averaging 8196 trials.

Session 4pNS

Noise, Architectural Acoustics, Psychological and Physiological Acoustics, Computational Acoustics, and Signal Processing in Acoustics: Soundscape and Virtual Reality

Ming Yang, Cochair
School of Architecture, University of Sheffield, HEAD acoustics GmbH, Herzogenrath, 52134, Germany

Brigitte Schulte-Fortkamp, Cochair
HEAD Genuit Foundation, Ebert Straße 30 a, Herzogenrath, 52134, Germany

Chair's Introduction—1:00

Contributed Paper

1:05

4pNS1. On the development of virtual reality tools to aid in the design interior soundscapes. Semiha Yilmazer (Ray W. Herrick Labs., Purdue Univ., Ray W Herrick Lab 177, S Russell St., West Lafayette, IN 47907, syilmaze@purdue.edu), Patricia Davies (Ray W. Herrick Labs., Purdue Univ., West Lafayette, IN), and Cengiz Yilmazer (CSY R&D and Architecture Eng., West Lafayette, IN)

Can simulations of interiors spaces (visual and acoustic) be used to improve designs of spaces within buildings? The first step in the simulation of the sounds in a space is to characterize typical sound sources and their locations within the space. The next step is to characterize the paths from the sources to the user, based on geometry and materials used. Combining the contributions from all sources, is it possible to create sounds that are similar to sounds in the real environment? How close do simulated and measured sounds need to be to evoke similar emotional responses from users? The same questions arise for the visual experience and for interactions between the visual and acoustic experiences. Are there metrics that can be used to guide improvements in the virtual environment to match the real experience in the built environments? To start to answer these questions, a sound simulation program using an image source modeling approach was developed for spaces in an existing building. The impulse responses from the image source model and the measured impulse responses are compared, as are simulated and measured sounds. Room acoustics and sound quality metrics for the virtual and real sounds are also compared.

Invited Paper

1:20

4pNS2. Use of virtual reality in designing and developing soundscape for dementia care facilities. Arezoo Talebzadeh (Ghent Univ., 126 Tech Ln. Ghent Sci. Park, Ghent 9052, Belgium, arezoo.talebzadeh@ugent.be), Patricia De Vriendt (Artevelde Univ. College, Ghent, Belgium), Paul Devos, and Dick Botteldooren (Information Technol., Ghent Univ., Ghent, Belgium)

Sound plays an essential role in making people conscious of space and time; however, once the cognitive ability declines, the same stimulus can become a source of anxiety and disturbance. People with dementia have difficulties identifying and interpreting their sensory stimuli; perception of space can become confusing and disturbing. They perceive and understand the sonic environment differently; the most obvious difference is the wrong meaning they may give to the sounds they notice due to changing mental association. A well-designed soundscape can help to reduce anxiety and distress and improve sleep and overall quality of life. In this research, a personalized soundscape was implemented in the dementia unit, adding recognizable sounds to the sonic environment to give people a feeling of safety, elevating their mood and activating them. Virtual Reality is a tool to present the result to stakeholders and further implement soundscape into designing care facilities. Soundscape is the sonic environment as perceived in context; VR helps define the context. Walking through the space while hearing the soundscape demonstrates how sound helps spatial orientation and understanding of time. We need to remember that people with dementia have a different mental model, and VR can help feature diverse mental models. The goal is to use Virtual Reality to enhance the soundscape experience during the design and development, especially for care facilities.

1:40

4pNS3. Post Sonica: a speculative design project that cultivates community and sense of place within the city soundscape. Andrew Green (Design, Univ. of Utah, 153 First Ave. N, Salt Lake City, UT 84103, dmckgreen@gmail.com)

There is a strong history of soundscape reverence and reflection in Utah. From Messiaen to Ussachevsky, scholars and composers alike have been inspired by the state's sonic environment. Using these individuals' reflective works as a muse while simultaneously incorporating R. Murray Schafer's insights on soundscapes, I performed a sound sample study across Salt Lake City. Analyzing my samples led me to design a public installation that explores the ways in which our sonic environments affect daily life at the intersection of time, sound, and place. The concept for post sonica is a granite cylinder (4 ft. wide $\times$ 3 ft. high). Three arms rotate from the center at variable speeds in response to the strength and pitch of the sounds around them. Pads at the arms' ends wear away separate paths in the granite base with each pass, creating the sonic "fingerprint" of a place over time. User engagement is necessarily bidirectional; as passersby are moved to interact with the device, their moment of reflection is captured and physicalized by the spinning arms. In this way, post sonica is a reminder of an urban community's fundamental interconnectedness and the ever-present soundscape that functions as a part of it all.

1:55

4pNS4. Optimization of loudspeaker positions for virtual auditory displays in a free-field environment. Nathaniel Wells (Brigham Young Univ., BYU Dept. of Phys. and Astronomy, N283 ESC, Provo, UT 84602, nateswells@gmail.com), Scott D. Sommerfeldt, and Jonathan D. Blotter (Brigham Young Univ., Provo, UT)

Virtual acoustic environments are created by mimicking the pressures of real acoustic sources at a listener's ears. In this way, a listener can perceive virtual acoustic sources as coming from any desired direction. There are multiple methods to accomplish this using headphones or loudspeaker arrays, but not all are practical for everyday use. Some use cases limit the use of headphones or the number and position of loudspeakers, such as automobiles and home theater systems. These limitations also put bounds on the performance of each possible system, such as allowable head movement and the accuracy of localization. An exhaustive-search optimizer was used to find the performance of various sparse loudspeaker arrays, given a limited number of possible loudspeaker positions in a free-field environment for a single stationary listener. The optimizer sought to maximize allowable head rotation and translation and to minimize localization errors. Arrays consisting of two or four loudspeakers were considered using a spherical head model to find the necessary head-related transfer functions and cross-talk cancellation was used to create the virtual acoustic environment. From the results, pareto fronts were created that can be used to find the best loudspeaker array configuration for given design constraints.

2:10

4pNS5. Location-aware synthesis of walkable urban soundscapes in room-centered immersive virtual reality system. Mincong Huang (Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, huangm5@rpi.edu) and Jonas Braasch (School of Architecture, Rensselaer Polytechnic Inst., Troy, NY)

This work, situated at Rensselaer's Collaborative-Research Augmented Immersive Virtual Environment Laboratory (CRAIVE-Lab), demonstrates a system that utilizes the facility's panoramic display and multichannel wave field synthesis loudspeaker array to simulate navigable human-scale urban environments with automatically generated virtual soundscapes. The system positions the CRAIVE-Lab's virtual footprint within the Unity game engine and provides it with the capability to move within virtual space. With geolocation input, the system uses ArcGIS to extract geospatial features, such as urban topologies and building extrusions. The same input is also used to retrieve real-time weather data from open-source databases (i.e., OpenWeather). Based upon the extracted information, the system updates acoustic signatures of the virtual surroundings by performing a multichannel ray-tracing analysis at a fixed time frame. The resulting signatures are then used to generate environmental noise profiles and process autogenerated virtual sound sources present in the environment using wave field synthesis and an extension of multiple audio datasets typically used for model training in urban sound classification (i.e., UrbanSound8K). We present the results as part of an *in situ* audiovisual experience where users can stand in the CRAIVE-Lab's physical enclosure and walk about the virtual landscape in which they are immersed.

Session 4pPA

Physical Acoustics, Structural Acoustics and Vibration, Computational Acoustics, Engineering Acoustics, and Signal Processing in Acoustics: Machine Learning in Acoustic Metamaterials

Feruza Amirkulova, Cochair
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Valerie J. Pinfield, Cochair
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Chair's Introduction—1:00

Invited Papers

1:05

4pPA1. Generative design of acoustic metamaterials conditioned on desired transmission characteristics. Caglar Gurbuz (Chair of Vibroacoustics of Vehicles and Machines, Tech. Univ. Munich, Boltzmannstr. 15, Garching 85748, Germany, caglar.guerbuez@tum.de) and Steffen Marburg (Chair of Vibroacoustics of Vehicles and Machines, Tech. Univ. Munich, Garching, Germany)

Metamaterials gain increasing interest in acoustics due to their sound insulation abilities. The special arrangement of unit cells allows them to influence the way sound propagates in acoustic media. To date, a system-based approach is mainly adopted in the design process. This approach is accompanied with two drawbacks: Firstly, the design relies primarily on the expertise of specialists making the current process prone to subjective assessments. Secondly, the relevant target criteria can only be evaluated at the end of design phase. This is accompanied with substantial time and resource expenses and is even amplified when large scale simulations or physical experiments are involved. It is the aim of this contribution to introduce a design assistant which supports specialists in the design of acoustic metamaterials. The proposed assistant system aims at a criteria-centric design scheme. By this means, target criteria, which are only accessible at the final stages, can be considered early in the design process. Therefore, a conditional generative adversarial network is implemented. This network proposes cell candidates conditioned on the desired transmission characteristics for the metamaterial. To validate our method, simulations with the finite element method are performed. The findings of this investigation suggest an important role for generative adversarial networks in the design process of acoustic metamaterials. As such, the proposed framework allows to design cell candidates for problem-specific applications.

1:25

4pPA2. Acoustic metamaterial inverse design based on machine learning using Gauss–Bayesian models. Jing Yang (Key Lab. of Modern Acoust., MOE, Inst. of Acoust., Dept. of Phys., Nanjing Univ., 22 Hankou Rd., Gulou District, Nanjing, Jiangsu Province 210093, China, yangj@nju.edu.cn), Zi-Xiang Xu, An Chen, Bin Liang, and Jian-chun Cheng (Key Lab. of Modern Acoust., MOE, Inst. of Acoust., Dept. of Phys., Nanjing Univ., Nanjing, China)

The emergence of acoustic metamaterials (AMs) provides distinct response to acoustic waves unattainable in nature. However, conventional AMs design has relied on extensive experimentation and a trial-and-error approach requiring significant computational resource. Here, we introduce an inverse-design method based on machine learning using Gauss–Bayesian (GB) models and implement it on low-frequency broadband sound attenuating devices. By utilizing the superiority of GB model on multi-parameter design and optimization efficiency, we successively customize ventilated microperforated sound-absorbing structure, sound absorbers with coherently coupled weak resonances (CCWRs) and ventilated meta-silencers with acoustic consecutive Fano resonances (ACFRs). Under the judiciously adaptive acquisition function and Matérn kernel, the designed GB frameworks with small samples are available to seek the optimal solution established on above physical effects by the cycle of training and prediction with high accuracy and efficiency. Good agreements between numerical and experimental results further verify the effectiveness of the proposed method. With the advantage of low cost, high efficiency and flexibility, our GB-based machine learning method paves the way for tremendous opportunities in future design of various metamaterial-based applications, as well as far-reaching implications for acoustics-related fields.

1:45

4pPA3. Design of acoustic metasurfaces with independent phase and amplitude control. Arvind Krishna (Northwestern Univ., Evanston, IL), Steven R. Craig (Mech. Eng., Georgia Inst. of Technol., Atlanta, Georgia), Chengzhi Shi (GWW School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA), and Roshan V. Joseph (Georgia Inst. of Technol., School of Industrial and Systems Eng., Atlanta, GA 30332, roshan@gatech.edu)

We address the problem of designing acoustic metasurfaces for independent amplitude and phase control of acoustic waves. The reflection and transmission amplitude and phase of an acoustic wave are governed by the geometry of the acoustic metasurface. The geometry is modeled by discretizing the continuous space into a finite number of elements, where each element can either be filled with air, or solid material. Full wave simulations on the COMSOL Multiphysics software are performed to obtain the acoustic outputs for a given geometry. In case of smaller-sized elements, it is computationally infeasible to consider all geometries, which correspond to all possible combinations of solid or air-filled elements. To address this challenge, we propose a novel 'acoustic domain space expansion algorithm' that starts with a few geometries and adaptively adds geometries to the set such that they span the entire domain of the desired acoustic outputs, using only a small fraction of all the possible geometries. Our algorithm simulated only 24 000 geometries out of the 563 trillion possible geometries, which can be used to obtain any feasible combination of the acoustic outputs with a tolerance of 5.4% in the individual outputs.

2:05

4pPA4. Efficient metamodeling of acoustic metasurfaces with variable sized design domains. Tyler J. Wiest, Carolyn C. Seepersad (Walker Dept. of Mech. Eng. & Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Michael R. Haberman (Walker Dept. of Mech. Eng. & Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, haberman@utexas.edu)

Traditional approaches to numerically model acoustic metasurfaces (AMS) with deeply subwavelength features require tremendous computational expense. Metamodels created using machine learning (ML) methods are promising techniques to efficiently and accurately model macroscopic AMS behavior while considering subwavelength features. However, many ML methods require a fixed quantity of design features so that ML model input dimensionality is constant. This work presents a metamodeling approach that is valid for metamaterial systems whose design domain has variable input feature dimensionalities. An example is an AMS consisting of variable numbers of cells, each with potentially unique geometry. We achieve improved modeling efficiency by representing design feature connectivity as an attributed graph and training a neural network (NN)-based model to operate on the attributes of graph elements. The metamodel is trained to update each attribute from state-to-state in a dynamic process. This NN architecture enables a single model to represent the behavior of an asymmetrically absorbing AMS composed of variable numbers of cells and nonperiodic geometry. The NN architecture can therefore be trained on data that includes a large proportion of systems with a small domain size but that is generalized to predict performance for larger systems that require greater computational expense for analysis.

Contributed Papers

2:25

4pPA5. Acoustic metamaterial design framework using deep learning and generative modeling. Feruza Amirkulova (Mech. Engieering, San Jose State Univ., 1 Washington Sq., San Jose, CA 95192, feruza.amirkulova@sjsu.edu), Linwei Zhou, Anam Abbas (Mech. Engieering, San Jose State Univ., San Jose, CA), Peter Lai (San Jose State Univ., San Jose, CA), Cheng Qiu (Mech. Eng., San José State Univ., Concord, CA), and Tristan A. Shah (San Jose State Univ., San Jose, CA)

This talk presents our findings and research performed on the application of deep learning and generative modeling in acoustic metamaterial design. Specifically, we will discuss our research findings published in recent papers on the application of deep learning algorithms, generative neural networks, reinforcement learning models, and global optimization for the inverse design of 2-D and 3-D acoustic metamaterial structures. The examples will be shown for the implementation of neural networks models for the inverse design of 3-D pentamode structures resulting in a low shear modulus, high bulk modulus, and an impedance matched with water. The generative 2-D GLO-Nets and the reinforcement learning models producing broadband low scattering effects for 2-D planar configurations of scatterers under plane wave incidence will be presented. The current challenges encountered during the application of deep learning methods in scaled metamaterial design will be discussed.

2:40

4pPA6. Acoustic metamaterial design using Conditional Wasserstein Generative Adversarial Networks. Peter Lai (San Jose State Univ., 1 Washington Sq., San Jose, CA 95192, peter.lai@sjsu.edu) and Feruza Amirkulova (Mech. Engieering, San Jose State Univ., San Jose, CA)

This talk presents a method for generating planar configurations of scatterers with a reduced total scattering cross section (TSCS) by means of generative modeling and deep learning. The TSCS minimization via repeated forward modeling techniques, trial-error methods, and traditional optimization methods requires considerable computer resources and time. However, similar or better results can be achieved more efficiently by training a deep learning model to generate such optimized configurations producing low scattering effect. In this work, the Conditional Wasserstein Generative Adversarial Networks (cWGAN) is combined with Convolutional Neural Networks (CNN) to create the generative modeling architecture [1]. The generative model is implemented with a conditional proponent to allow the TSCS targeted design generation and is enhanced with the coordinate convolution (CordConv) layer to improve the model's spatial recognition of cylinder configurations. The cWGAN model [1] is capable of generating images of 2D configurations of scatterers that exhibit low scattering. The method is demonstrated by giving examples of generating 2-cylinder and 4-cylinder planar configurations with minimal TSCS. [1]. P. Lai, F. Amirkulova, and P. Gerstoft. "Conditional Wasserstein generative adversarial networks applied to acoustic metamaterial design," *J. Acoust. Soc. Am.* **150**(6), 4362–4374 (2021).

2:55–3:10 Break

Invited Papers

3:10

4pPA7. Acoustic metamaterial-based neural network for task solving. Jingkai Weng, Yujing Ding (Inst. of Acoust., Nanjing Univ., P. R. China, Nanjing, China), Bin Liang (Inst. of Acoust., Nanjing Univ., P. R. China, 22 Hankou Rd., Nanjing, Jiangsu 210093, China, liangbin@nju.edu.cn), Xuefeng Zhu (Huazhong Univ. of Sci. and Technol., Wuhan, China), Jing Yang (School of Phys., Nanjing Univ., Nanjing, Jiangsu Province, China), and Jian-chun Cheng (Inst. of Acoust., Nanjing Univ., P. R. China, Nanjing, China)

Simple and precise sound manipulation is of fundamental interest and practical significance in acoustics and many related fields. The recent emergence of acoustic metamaterials offers acoustic properties unattainable in the nature and opens up possibility for controlling sound, which may also find applications in diverse scenarios. However, acoustic metamaterials still cannot directly solve complicated tasks, such as recognition of objects that needs to analyze scattered wave. Deep learning technique helps to eliminate the conventional dependence on human experts, but is still arouse issues of computation complexity, energy supply, device size and cost. We build a passive neural network by using numerous metamaterial unit cells, whose phase shifts are chosen as learnable parameters for the machine learning training. Such a metamaterial-based neural network interacts with the scattered wave produced by the object and accurately redistributes the output energy on a detection plane, exhibiting the "intelligence" to perform complex machine-learning tasks. Its equivalence with a conventional neural network is analytically proved, and its task-solving performance is numerically and experimentally demonstrated. More importantly, detection and computation are performed simultaneously, and the computational complexity or energy consumption will not increase even if the object becomes more complicated.

3:30

4pPA8. Interval uncertainty propagation in metamaterial beams using machine learning based optimization. Lucas Van Belle (Dept. of Mech. Eng., KU Leuven & DMMS Lab, Flanders Make, Celestijnenlaan 300 box 2420, Heverlee 3001, Belgium, lucas.vanbelle@kuleuven.be), Felipe Igea (Dept. of Eng. Sci., Univ. of Oxford, Oxford, United Kingdom), Elke Deckers (Dept. of Mech. Eng., KU Leuven & DMMS Lab, Flanders Make, Diepenbeek, Belgium), and Alice Cicirello (Mech. and Phys. of Structures Section, TU Delft, Delft, Netherlands)

Metamaterials have recently emerged in the search for lightweight noise and vibration solutions. One of their appealing properties for noise control engineering is the ability to create stop bands, which are frequency ranges without free wave propagation. These stop bands arise from the sub-wavelength addition of identically tuned resonators in or on a host structure. However, when manufacturing metamaterials, variability in material properties and geometry is inevitably introduced. On the one hand, the metamaterial attenuation performance can deteriorate due to variability, while on the other hand, variability can even broaden their typically narrowband performance. During the early phases in the design, when often little information concerning the inherent variability is available, it would be desirable to be able to assess the input uncertainty effects on the performance of metamaterials. This work focuses on the numerical assessment of the impact of uncertainties in resonator properties on the vibration attenuation performance of metamaterial beams. To this end, a machine learning based optimization strategy, the so-called Bayesian optimization, is employed in a recently developed non-intrusive uncertainty propagation approach. This enables a very efficient evaluation of the upper and lower bounds on the metamaterial performance, with the uncertain resonator parameters defined as interval variables.

3:50

4pPA9. Designing acoustofluidic devices using a simplified physics-informed machine learning approach. Samuel J. Raymond (Climate and Sustainability Consortium, MIT, 161 Putnam Ave. Apt. 1, Cambridge, MA 02139, sjr@mit.edu), David Collins (Univ. of Melbourne, The University of Melbourne, Victoria, Australia), and John Willams (Civil and Environ. Eng., MIT, Cambridge, MA)

The safe positioning of particles within an acoustofluidic device is critical in biomedical and biological applications. Relating the design of acoustofluidic device walls and the internal acoustic field is a complex, nonlinear problem. The field of Physics-Informed Machine Learning (PIML) offers a number of potential approaches to simplify the design of these devices. One such PIML approach is learning from synthetic data. With large scientific data sets with rich spatial-temporal data and high-performance computing providing large amounts of data to be inferred and interpreted, the task of PIML is to ensure that these predictions and inferences are enforced by, and conform to the limits imposed by physical laws. The tools employed in PIML can include large, deep neural networks, Bayesian modeling, and deep reinforcement learning with sophisticated simulations of the environment. In this work, we show a simplified version of PIML using a combination of a small fully connected neural network and a 2D meshfree simulator of acoustic devices to predict the boundary shape for an acoustically actuated device. We will discuss the real-world results and applications, as well as the current limitations of this approach and the path ahead to scale and include more complexity for more applications and designs.

4:10

4pPA10. Efficient artificial intelligence techniques for inverse design and knowledge discovery in metamaterials. Mohammadreza Zandehshahvar, Yashar Kiarashinejad, Mohammad Hadighe Javani, Muliang Zhu, Tyler Brown, Daqian Bao (Elec. and Comput. Eng., Georgia Inst. of Technol., Atlanta, GA), and Ali Adibi (Elec. and Comput. Eng., Georgia Inst. of Technol., 778 Atlantic Dr. NW, Atlanta, GA 30004, ali.adibi@ece.gatech.edu)

A new artificial-intelligence (AI) approach based on dimensionality reduction techniques for the design and knowledge discovery in metamaterial structures will be presented. It is shown that reducing the dimensionality of the response and design spaces by using a pseudoencoder can reduce the computation time by 2–3 orders of magnitude. In addition, using pruning techniques can simplify the trained algorithm to further reduce the computation time while providing simpler paths for relating input-output relationships for understanding the role of design parameters (i.e., knowledge discovery). It is also shown that this approach can enable an evolutionary design method in which the initial design can be evolved intelligently into an alternative with favorable specifications. The application of these AI approaches for both optical and acoustic metamaterials will be presented.

4:30

4pPA11. Reinforcement learning for acoustic metamaterial design. Tristan A. Shah (San Jose State Univ., 1 Washington Square, San Jose, CA 95192, tristan.shah@sjsu.edu) and Feruza Amirkulova (Mech. Eng., San Jose State Univ., San Jose, CA)

This talk presents our findings and methodologies in applying reinforcement learning to the design of acoustic metamaterials [1]. Our reinforcement learning agents are capable of discovering configurations of cylindrical scatterers in water which minimize the scattering of an acoustic plane wave. These agents are successful in the optimization of two different parametric designs, i.e., radii and positions of each scatterer. The resultant designs produced by reinforcement learning algorithms such as double deep Q-learning network and deep deterministic policy gradient algorithms are comparable and in some cases superior to those produced by the gradient-based optimization solver such as fmincon. However, significant computational resources are required to train reinforcement learning models to completion. This poses a significant challenge when attempting to increase the complexity of a design; as training times increase to unfeasible levels. Methods to counteract this challenge are discussed in this talk. These methods include utilization of the Julia programming language as well as multiprocessing and multithreaded programming. Together they create a synergy which reduces training times by orders of magnitude. [1] T. Shah, L. Zhuo, P. Lai, A. Rosa-Moreno, F. Amirkulova, and P. Gerstoft, "Reinforcement learning applied to metamaterial design," *J. Acoust. Soc. Am.* **150**(1), 321–338 (2021).

4:45

4pPA12. Sound localization via deep learning, generative modeling, and global optimization. Wei-Ching Wang (Mech. Eng., San Jose State Univ., 375 Page st., San Jose, CA 95126, wangw9277@gmail.com)

An acoustic lens is capable of focusing incident plane waves at the focal point. This talk will discuss a new approach to design metamaterials with a focusing effect using effective and innovative methods by using machine learning. Specifically, the physics simulations use multiple scattering theory and machine learning techniques such as deep learning and generative modeling. The 2-D-Global Optimization Networks (2-D-GLOnets) model [1] developed initially for acoustic cloak design is adapted and generalized to design and optimize the acoustic lens. We supply the absolute pressure amplitude and gradient into the deep learning algorithms to discover the optimal scatterer positions that maximize the absolute pressure at the focal point in the confined region. We examine and evaluate the performance of the generative network, searching for an optimal configuration of scatterers over a range of parameters to produce desired features, such as broadband focusing and localization effects. The model will show examples of planar configurations of cylindrical structures producing focusing effect. [1] L. Zhuo and F. Amirkulova, "Design of acoustic cloak using generative modeling and gradient-based optimization," in *INTER-NOISE and NOISE-CON Congress and Conference Proceedings* (Institute of Noise Control Engineering, 2021), Vol. 3.

5:00

4pPA13. Pentamode metamaterial design via generative modeling and deep learning. Cheng Qiu (Mech. Eng., San José State Univ., 1790 Ellis St., Apt. 47, Concord, CA 94520, cheng.qiu@sjsu.edu), Anam Abbas, and Feruza Amirkulova (Mech. Eng., San José State Univ., San Jose, CA)

In this talk, the deep learning-assisted models will be presented for the design of pentamode unit cells to construct a 3-D lattice structure that can mimic the acoustic properties of water. The pentamode models were implemented and modified by altering the properties of the structure during the full-wave simulation performed on COMSOL Multiphysics software to ensure they meet the requirements of specific appliances for manufacture. The design is further improved by the inverse design technique. The implementation of conditional Wasserstein Generative Adversarial Networks with gradient penalty (cWGAN-GP) for the inverse design will be illustrated by showing examples of 3-D titanium pentamode structures on the hexagonal lattice. The cWGAN was set up using critic and generator with CoordConv layers, along with regressor to produce images matching the desired labels. The convolutional neural network was used as an auxiliary regressor to predict all design parameters for the cell images. The expected PM lattice structure has a high bulk modulus, low-shear modulus and behaves as metal water at a broad range of frequencies including higher frequencies.

Session 4pPP

Psychological and Physiological Acoustics: Human and Animal Physiology (Poster Session)

Bobby E. Gibbs, Cochair
Hearing and Speech Sciences, University of Maryland, College Park, 7251 Preinkert Dr., College Park, MD 20742

Michael L. Smith, Cochair
Department of Speech Language Sciences, University of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55455

All posters will be on display form 1:30 p.m. to 4:30 p.m. Authors of odd-numbered papers will be at their posters from 1:30 p.m. to 3:00 p.m. and authors of even-numbered papers will be at their posters from 3:00 p.m. to 4:30 p.m.

Contributed Papers

4pPP1. Feasibility study for the hearing assessment of classical music student using Otoacoustic Emissions. Stephen Dance (School of the Built Environment and Architecture, London South Bank Univ., Borough Rd., Torquay Rd., London SE1 0AA, United Kingdom, dances@lsbu.ac.uk) and Eric Ballestero (Dept. of Acoust., Univ. of Le Mans, Le Mans, France)

Since the introduction and enforcement of the Control of Noise at Work Regulations in 2008 continuing research has been undertaken on five thousand students of the Royal Academy of Music to understand their hearing acuity. Standard audiometric screening methods were employed for both entry and exit testing. It was established that the music students obtained results that in over 50% of cases presented negative hearing losses as compared against the Health and Safety Executive categories. This presented an interesting problem which has now been addressed in an attempt to understand the hearing acuity of classical musicians using an otoacoustic emission based methodology. A feasibility study was undertaken in July 2021 of 120 musicians using Hearing Coach software for the analysis and Path Medical OAE instrument to undertake the measurements. The results presented show similar trends and thus provide reassurance that OAE can be used to assess the hearing of music students.

4pPP2. Determining auditory brain stem response threshold using an adaptive stimulus selection procedure. Erik A. Petersen (Speech and Hearing Sci., Univ. of Washington, 1417 NE 42nd St., Seattle, WA 98105, erikp2@uw.edu) and Yi Shen (Speech and Hearing Sci., Univ. of Washington, Seattle, WA)

Auditory brain stem response (ABR) measurements provide a method to determine the hearing threshold of laboratory animals and humans who are unable to provide behavioral responses to auditory stimuli. Determining hearing threshold using ABR typically requires a large number of pre-selected frequency and stimulus levels, which may not be the most efficient method to sample the stimulus space. The goal of the current study is to develop an adaptive procedure that determines *in situ* the stimulus that will provide the best estimate of the threshold, based on data collected earlier in the procedure. A Gaussian Process model is iteratively fitted after each test level, and the subsequent test level is chosen at the interim threshold estimate predicted by the current model fit. Simulations of the adaptive procedure were conducted using previously collected toneburst ABR measurements from mice presented at 5 dB increments from 0 to 80 dB at seven frequencies ranging from 4 to 32 kHz. The thresholds determined by this procedure were compared to those produced by human raters who viewed the response at all stimulus levels. Initial results indicate that the threshold may be accurately estimated utilizing fewer stimuli levels than current methods, thereby reducing the duration of the measurements.

4pPP3. Relationship of noise history to perceived loudness and physiological arousal from noise. Alanalyn N. Pinaula-Toves (Univ. of Minnesota-Twin Cities, 924 17th Ave. SE, Apt. 203, Minneapolis, MN 55414, pinau003@umn.edu), Peggy Nelson (Ctr. for Applied/Translational Sensory Sci., Univ. of Minnesota, Minneapolis, MN), Steven P. Gianakas (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN), Jessica Sullivan (Communicative Sci. Disord., Hampton, Hampton, VA), and Matthew B. Winn (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN)

Chronic exposure to noise is a well-documented health hazard (Suter, 1991). In this study, we investigated whether an individuals' self-reported noise history (exposure, sensitivity, and annoyance) correlated with noise-elicited physiologic arousal. We measured pupil dilation responses and loudness judgments to varying types of noise and related those to self-identified noise history. We hypothesized that increased history of exposure or annoyance would be correlated with perceived loudness and arousal from noise. Five colors (spectra) of noise were generated with the same sound pressure levels within the band 1000–4000 Hz but with different slopes of spectral energy below and above the band. Changes in participant pupil dilation responses elicited by each noise type were used as an assessment of physiological stress response to noise stimuli. Participants also adjusted the loudness of two randomized noise spectra until they were determined to be of equal loudness. Preliminary data suggest that the perception of loudness and levels of arousal differed depending on the color spectrum of noise. Relationships between the loudness judgments, self-reported noise history, and observed physiologic data will be further discussed. Results have implications for understanding the behavioral and physiological effects of chronic exposure to noise. [Robert Young, NIDCD R01DC017114 (Winn).]

4pPP4. The difference between using context to predict versus using context to repair: a study of listening effort. Michael L. Smith (Speech-Language-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, smit8854@umn.edu) and Matthew B. Winn (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN)

Semantic context in sentences has been shown to increase intelligibility and reduce listening effort. However, context has typically been tested in situations where it is used to predict upcoming words rather than repair

earlier words. The current study attempts to directly compare predictive versus retroactive context for the first time for the same listeners in the same study. We hypothesized an increase in effort for retroactive context compared to the elevated effort of predictive context. Cochlear implant and Normal Hearing listeners heard sentences where a single content word was masked by noise, either early in the sentence (requiring retroactive repair) or later in the sentence (where it was predicted by earlier words). Changes in pupil dilation were measured as an index of listening effort linked with the moment of each word and its context. Effort increases when context is used to repair an earlier missing word, even if intelligibility is perfect. There is a smaller increase of effort when the missing word follows context that enabled prediction of the word. This study provides a novel approach in directly comparing the mental cost of using context to repair versus predict misperceived words in a sentence.

4pPP5. Aging effects on binaural listening and other auditory system assessments. Kerry A. Walker (Otolaryngol., Univ. of Colorado School of Medicine, 12631 E 17th Ave., Aurora, CO 80045, kerry.walker@cuanschutz.edu), Carol A. Sammeth (Physiol. & Biophys., Univ. of Colorado School of Medicine, Aurora, CO), Nathaniel Greene (Otolaryngol., Univ. of Colorado School of Medicine, Aurora, CO), Achim Klug, and Daniel Tollin (Physiol. & Biophys., Univ. of Colorado School of Medicine, Aurora, CO)

Older listeners are known to have decreased sound localization and hearing in noise abilities compared to young listeners, even when exhibiting clinically normal hearing. We hypothesize that these deficits may be related to decreased temporal precision of neural activity in the sound localization processing pathway, and/or other central auditory system deficits involving more complex functions such as speech recognition. Presented here are preliminary results from a study underway on the impact of aging on auditory functioning, and in particular, binaural processing. Subjects range from 21 to 89 years of age, have normal or near normal-hearing sensitivity, and are being evaluated on assessments of auditory system integrity and behavioral performance. Primary outcome measures include speech understanding in noise, auditory brainstem responses (ABRs) and the calculated ABR binaural interaction component, behavioral measures of spatial acuity, and a subjective questionnaire of hearing handicap. Secondary measures include temporal fine structure and spectro-temporal modulation sensitivity tests, otoacoustic emissions, and electrocochleography. In addition, extended-high-frequency hearing thresholds are measured and working memory assessed. Early results show an aging effect across decades of life on several of these measures. Greatest aging effects are seen in the more adverse listening conditions from behavioral testing. [Support: R01-DC017924.]

4pPP6. Neural markers of improved auditory selective attention following neurofeedback training. Hwan Shim (Univ. of Iowa, 250 Hawkins Dr., Iowa City, IA 52242, hwan-shim@uiowa.edu), Subong Kim (Purdue Univ., West Lafayette, IN), Jae Hee Lee, Leah Gibbs, Karsyn Rush, and Inyong Choi (Univ. of Iowa, Iowa City, IA)

We designed a neurofeedback training paradigm to enhance the attentional modulation of cortical auditory evoked responses. Two concurrent speech streams—a female voice repeating "Up" five times and a male voice repeating "Down" four times in three secondsi—were played from left and right loudspeakers, respectively. Subjects were instructed to attend one of those streams by a pre-stimulus visual cue (e.g., "Target: Up"). Attention was decoded from single-trial EEG signals. Subjects received neurofeedback (i.e., a visual object on the computer screen moves upward if the EEG decoder determines the "up" stream was attended; or vice versa). Subjects repeated this training for four sessions across four weeks. During the last session, compared to the first, subjects exhibited strengthened alpha oscillation in the right parietal cortex while they attended right and ignored left, indicating that spatial inhibitory processing to suppress sounds left was enhanced. The temporal cortex exhibited greater attentional modulation of beta oscillation after the four weeks of training, indicating enhanced neural activity to predict the target. The strength of attentional modulation on cortical evoked responses to sounds was also improved. These results prove that neurofeedback training effectively enhances the cortical processing for auditory selective attention.

4pPP7. Toward improved measurement of infant bone conduction auditory brainstem responses. Kristin M. Uhler (Physical Medicine and Rehabilitation, Univ. of Colorado School of Medicine, 12631 East 17th Ave., Academic Office 1, Ste. 1201, Aurora, CO 80045, kristin.uhler@cuanschutz.edu), Aoi A. Hunsaker (Speech and Hearing Sci., Univ. of Washington, Seattle, WA), Nathaniel Greene, Kerry A. Walker (Otolaryngol., Univ. of Colorado School of Medicine, Aurora, CO), and Andrew D. Brown (Speech and Hearing Sci., Univ. of Washington, Seattle, WA)

Newborn hearing screening has led to improved detection and treatment of hearing loss (HL) in infants and young children, supporting improved communication outcomes in early childhood and beyond. However, current screening protocols are not equally sensitive to all forms of HL, limiting diagnostic accuracy and, ultimately, identification of appropriate treatment options. Correspondingly, treatment outcomes in children with HL remain quite variable. One fundamental and persistent challenge is the assessment of *bone conduction hearing*, which is essential to differentiate conductive versus sensorineural pathology. Whereas hearing via air conduction can be effectively assessed using the auditory brainstem response (ABR), a noninvasive electrophysiologic measure, bone conduction ABR measurements suffer from comparatively poor signal quality (including significant stimulus artifact), transducer acoustic output limitations, and generally lower test-retest reliability. Here we evaluated the prospective benefits of measuring infant bone conduction ABRs using a modified bone conduction transducer designed to reduce stimulus artifact and thereby improve measurement quality. Measurements obtained using a standard bone conduction transducer were compared on several dimensions to measurements obtained using the modified transducer. Improved bone conduction measurement tools are expected to support improved detection and classification of conductive and mixed HL, leading to improved treatment outcomes for this important patient population.

4pPP8. Measuring ongoing delay with reverse noise. Philip X. Joris (Neurosciences, KU Leuven, Herestraat 49, Bus 1021, Leuven B-3000, Belgium, Philip.Joris@kuleuven.be)

Counts of spike coincidences provide a powerful means to compare responses to different stimuli or of different neurons, particularly regarding temporal factors. A drawback is that these methods do not provide an absolute measure of latency, i.e., the temporal interval between stimulus and response. It is desirable to obtain such a measurement within the same analysis framework of coincidence counting. Single neuron responses were obtained from several fiber tracts (nerve, trapezoid body, lateral lemniscus) in two species to a broadband noise and its polarity-inverted version. The spike trains in response to these stimuli are the "forward noise" responses. The same stimuli were also played time-reversed. The resulting spike trains are then again time-reversed: these are the "reverse noise" responses. The forward and reverse responses were then analyzed with the coincidence count methods we have introduced earlier. Latency measurements derived from correlograms between forward and reverse noise responses show maxima at twice the latency value measured with other methods, as expected. Correlograms were often asymmetric but, at low-characteristic frequencies, were well-predicted by crosscorrelation of the reverse-correlation function and its reverse. We conclude that reverse noise provides an easy and reliable means to estimate latency of auditory nerve and brainstem neurons.

4p THU. PM

4pPP9. Brain perturbation to reveal causal contributions of executive networks to the mechanisms of auditory attention. Kayla Howerton (Univ. of Iowa, 250 Hawkins Dr., Iowa City, IA 52242, kayla-howerton@uiowa.edu), Aaron Boes (Univ. of Iowa, Iowa City, IA), Barbara Shinn-Cunningham (Carnegie Mellon Univ., Pittsburgh, PA), and Inyong Choi (Univ. of Iowa, Iowa City, IA)

Listeners face the challenge of understanding speech every day. A key cognitive strategy for parsing speech in noisy environments relies on auditory selective attention, which recruits a network of multiple cortical regions. Our recent studies identified at least two different neural mechanisms of auditory attention. Space-based attention engages alpha oscillations in the parietal cortex, which are thought to cause lateral suppression of streams in the ignored portion in space. In contrast, talker identity-based attention modulates beta oscillations in the temporal lobe, suggesting that a

neural pitch prediction mechanism in the middle temporal gyrus (MTG) focuses attention to talker. Based on these findings, here we use a brain perturbation method to establish causal links between such oscillatory activity and control of different forms of auditory attention. We hypothesize that perturbing alpha oscillations in parietal cortex will impair spatial attention while perturbing beta oscillations in the MTG will impair speaker identity-based attention. Our ongoing study utilizes repetitive transcranial magnetic stimulation (rTMS) to induce such perturbations in the pre-determined regions of interests non-invasively and safely. This presentation will provide proof-of-concept data to validate the feasibility of this approach.

4pPP10. Two kinematic gains underlying the organ of Corti mechanotransduction. Anes Macic, Wei-Ching Lin (Mech. Eng., Univ. of Rochester, Rochester, NY), and Jong-Hoon Nam (Mech. Eng., Univ. of Rochester, 203 Hopeman Bldg, 140 Hutchison Rd., Rochester, NY 14627, jong-hoon.nam@rochester.edu)

The organ of Corti (OoC) is the sensory epithelium in the mammalian hearing organ where vibrations are encoded into neural signals. The OoC vibrates due to external and internal agitations. Externally, differential fluid pressures across the OoC cause vibrations. Internally, actuator cells (outer hair cells) are known to boost the vibrations. These two agitations result in distinct vibration patterns. Eventually, these vibrations deflect the stereocilia bundle to activate mechanotransduction channels in them. How OoC vibrations result in the hair cell mechanotransduction is critical to hearing research, but a fundamental information (kinematic gain) has not been quantified clearly. Using the optical coherence tomography, we measured 2-D vibrations of the OoC in cochlear turns acutely excised from young Mongolian gerbils. The excised tissues were stimulated either mechanically or electrically, which are comparable to external or internal agitations, respectively. The stereocilia deflection was estimated from relative motion between overlying and underlying structures. Two kinematic gains were defined. Passive kinematic gain represents the amplitude ratio between the stereocilia deflection and the basilar membrane transverse vibration. Active kinematic gain represents the amplitude ratio between the stereocilia deflection and the outer hair cell length change. (Work supported by NIH NIDCD R01 DC014685.)

4pPP11. Simultaneous measurement of human auditory nerve and brainstem potentials: Effects of upward spread of excitation. Skyler G. Jennings (Commun. Sci. and Disord., Univ. of Utah, 390 S. 1530 E., Salt Lake City, UT 84112, skyler.jennings@hsc.utah.edu) and Tabitha Whitmore (Commun. Sci. and Disord., Univ. of Utah, Salt Lake City, UT)

Acoustic clicks evoke strong synchrony of auditory-nerve (AN) fibers, thereby eliciting a compound action potential (CAP); however, synchrony is reduced by the delay of the cochlear traveling wave. Upward-frequency chirps compensate for this delay, resulting in larger CAPs evoked by chirps than clicks. Our model simulations of CAPs suggest that chirps presented at high intensities (100 dB peSPL) adapt basal AN fibers via upward spread of excitation of low-frequency chirp components. This adaptation results in less-than-or-equal-to CAP amplitudes and distorted CAP morphology compared to CAPs evoked by clicks. Simulations show that these effects on CAP amplitude and morphology can be avoided by reducing the amplitude of the low-frequency components of the chirp (i.e., modified chirp). Here we present CAPs from 12 young adults with NH, which for high stimulus intensities, exhibit reduced amplitudes and broader CAP morphology for chirps than clicks, whereas CAP amplitudes for modified chirps exceed those for clicks, consistent with model simulations. The distorted CAP morphology in response to high-level chirps is consistent with adaptation of AN responses and broad cochlear excitation. Interestingly, the deleterious effects of high-level chirps on CAP amplitude and morphology are reduced or absent in simultaneous recordings of the auditory brainstem response.

4pPP12. Binaural brainstem and spatial hearing deficits in a guinea pig model of noise-Induced cochlear synaptopathy. Monica A. Benson (Biophys. and Physiol., Univ. of Colorado Anschutz Medical Campus, 12800 E 19th Ave., Aurora, CO 80045, monica.benson@cuanschutz.edu), John Peacock (Biophys. and Physiol., Univ. of Colorado Anschutz Medical Campus, Aurora, CO), Nathaniel Greene (Otolaryngol., Univ. of Colorado School of Medicine, Aurora, CO), and Daniel Tollin (Biophys. and Physiol., Univ. of Colorado Anschutz Medical Campus, Aurora, CO)

Animal studies have revealed that moderate-level noise exposure can cause a permanent loss of ribbon synapses between inner hair cells and auditory-nerve fibers, but only temporary threshold shifts. Such noise-induced cochlear synaptopathy has been called ‘hidden hearing loss’ because while there is a significant degeneration of ribbon synapses, the resulting hearing dysfunction is effectively hidden from typical clinical assays. Here we used guinea pigs to study the mechanisms leading to hearing deficits resulting from a moderate noise. We measured distortion product otoacoustic emissions and auditory brainstem responses to assay peripheral hearing. We tested spatial hearing ability through the prepulse inhibition (PPI) of the acoustic startle reflex. PPI was used to measure hearing-in-noise ability, or spatial release from masking, as this task approximates the “cocktail party” effect. Finally, we performed immunohistochemistry to confirm synaptopathy. Results show that the noise exposure induces no permanent hearing threshold shift, but despite recovery of normal audibility, both behavioral and electrophysiological binaural hearing deficits persisted. Cochlear synaptopathy was objectively confirmed by visualizing the loss of ribbon synapses in the cochlea. The results demonstrate that cochlear synaptopathy causes deficits in brainstem circuits known to be critical for binaural and spatial hearing. [Work supported by Otolaryngology T32 DC012280.]

4pPP13. The impact of prior topic awareness on listening effort during speech perception. Steven P. Gianakas (Speech-Language-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, giana009@umn.edu) and Matthew B. Winn (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN)

Most standard speech perception tests use unrelated sentences where the topic changes for each stimulus. This unrealistic paradigm is expedient but might overlook an important aspect of speech processing and effort, which is how listeners use expectations and topic awareness to enable easier repair of misperceived words. We hypothesize that the effort of mentally repairing misperceived words will be reduced when individuals have advance topic awareness of upcoming sentences. In this study, individuals with normal hearing and hearing loss heard sentences with a masked target word that required retroactive mental repair using later context. Sentences were either preceded by the presence or absence of a related priming word presented visually to simulate advance topic knowledge. Listening effort for these sentences was measured by changes in pupil size that were linked with the exact timing of the word being repaired. When the target word is masked in a sentence, pupil dilation increases immediately after the missing word, and that dilation is modulated by the helpfulness of the prime before the sentence.

4pPP14. Pitch-timbre interactions as measured in discrimination and the frequency-following response. Ryan Anderson (Speech, Lang., and Hearing Sci., Indiana Univ., 200 S. Jordan, Bloomington, IN 47401, anderyan@indiana.edu), Yi Shen (Speech and Hearing Sci., Univ. of Washington, Seattle, WA), and William P. Shofner (Speech and Hearing Sci., Indiana Univ., Bloomington, IN)

Perceptual interactions between pitch and timbre have been demonstrated repeatedly using discrimination tasks in psychoacoustics. The current study investigates whether the pitch-timbre interaction is reflected in neural signatures of pitch encoding in the brainstem. Listeners discriminated the pitch of harmonic tone complexes with low- or high-spectral centroids (SC) in a yes/no task. Fundamental frequency (F0) varied by one just noticeable difference (JND), while SC could be varied by two- or five- JNDs. Discrimination performance increased when both F0 and SC were higher as opposed to when one was higher, and one was lower. This suggests that higher SCs bias listeners toward “higher pitch” responses, replicating previous studies. Frequency-following responses (FFR) were obtained in response to the same stimuli from the same listeners. The recorded FFR signals were analyzed using autocorrelation functions, to evaluate the coding of periodicity, and the Fourier spectra, as an indicator of the encoded F0. The encoding of pitch-relevant information was similar for all stimuli, regardless of SC. Contrary to bias observed in pitch discrimination performance imposed by SC, the lack of a significant influence of SC in the FFR measurements suggests that the pitch-timbre interaction was not observed at the level of the brainstem.

4pPP15. Variance of alpha oscillation power corresponds to behavioral performance in spatial auditory attention. Benjamin N. Richardson (Neurosci. Inst., Carnegie Mellon Univ., 5702 Darlington Rd., Apt. 3, Pittsburgh, PA 15217, bnrichar@andrew.cmu.edu), Barbara Shinn-Cunningham (Neurosci. Inst., Carnegie Mellon Univ., Pittsburgh, PA), and Jasmine Kwasa (Elec. and Comput. Eng., Carnegie Mellon Univ., Pittsburgh, PA)

To analyze an acoustic stream from a desired location, the brain focuses spatial selective attention, which causes shifts in parieto-occipital alpha power (~8i–12 Hz). We explored whether neurotypical (N = 10) and ADHD (N = 36) adults differ in performance and in the time course of alpha in a spatial attention task. Subjects reported the order of syllables presented in spatially separated streams while EEG was recorded. In each trial, subjects either maintained attention on a central "target" stream (focal attention) or reported the content of a possible "interrupter" from the left (broad attention). Previously, we found that performance correlates with individual differences in how strongly listeners suppress, in focal trials, event-related potentials elicited by the interrupter. We hypothesized that oscillatory alpha signatures would differ for focal and broad attention, reflecting top-down control of spatial attention. Instead, we found that higher consistency in alpha oscillations correlated with better behavioral outcomes. Rather than deploying spatial attention differently in focal and broad conditions, good listeners may use the same listening strategy in both, but be better in staying on task (i.e., overall "focus"). We propose that the unpredictable interrupter engages bottom-up attentional pathways that are functionally separate from previously demonstrated spatial attention steering mechanisms.

4pPP16. Characterizing the effects of distorted tonotopy on neural coding and perception in sensorineural hearing loss. Homeira I. Kafi (Weldon School of Biomedical Eng., Purdue Univ., 715 Clinic Dr., Lyles-Porter Hall, West Lafayette, IN 47907, hkafi@purdue.edu), Joshua M. Alexander, and Hari Bharadwaj (Speech, Lang., and Hearing Sci., Purdue Univ., West Lafayette, IN)

Patients with similar audiograms often experience varying levels of difficulty with understanding speech in noisy listening environments, even with prescriptive amplification using state-of-the-art hearing aids. With audibility mostly restored by amplification, these suprathreshold differences are often attributed to degraded frequency selectivity (broadened cochlear tuning) due to sensorineural hearing loss (SNHL), and non-peripheral factors. However, evidence from recent animal models of noise-induced SNHL suggests that distorted tonotopy—i.e., hypersensitivity of the tails of cochlear frequency tuning curves—may be a significant contributor to degraded speech coding, particularly in the presence of background noise. In this study, a combination of behavioral measures, otoacoustic emissions, and electroencephalography (EEG) was used to investigate: (1) whether distorted tonotopy degrades the neural coding of speech in human listeners with SNHL as measured using EEG, and (2) whether variations in the degree of distorted tonotopy contribute to the large individual differences in perceptual outcomes that persist despite prescriptive amplification. Preliminary results from individuals with mild-to-moderate SNHL who listened to stimuli amplified by a hearing-aid simulator implementing individualized prescriptions suggest that distorted tonotopy (as measured using the tuning-curve tip-to-tail ratio) can indeed partly account for individual variations both in neural coding and behavior.

4pPP17. Ambiguous lateralization of complex binaural stimuli emerge from neurons constrained to physiologically relevant ranges of ITD and frequency—The binaural aperture. Daniel J. Tollin (Dept. of Physiol.,Univ. of Colorado School of Medicine, RC1-N, Rm. 7106, 12800 E 19th Ave., Aurora, CO 80045, daniel.tollin@ucdenver.edu), Matthew J. Goupell (Hearing and Speech Sci., Univ. of Maryland-College Park, College Park, MD), and G. Christopher Stecker (Ctr. for Hearing Res., Boys Town National Res. Hospital, Omaha, NE)

Although we perceive a spatially contiguous, integrated world, sensory systems sample information discretely via small spatial and narrow frequency receptive fields (RFs) in vision and audition, respectively. Undersampling of stimuli by narrow RFs produces ambiguous representations (e.g., the classic "aperture problem" that visual motion cannot be uniquely determined by one RF). Here we posit a similar problem for encoding interaural time differences (ITDs). Current models of ITD processing propose large ensembles of ITD-sensitive neurons spanning wide ranges of preferred ITD and characteristic frequency (CF). To account for anomalous lateralization of narrowband but robust lateralization of broadband stimuli, such models require two paradoxical assumptions that have recently come into question: (1) the existence of neurons preferring larger than physiologically possible ITDs; and (2) population biases overemphasizing small ITDs. Here, we consider an alternative account based on a highly constrained ITDxCF aperture through which the stimuli are perceived. ITDs bounded by $\pm\pi$ interaural phase difference and CFs by dominance weighting, where ~600-750 Hz contributes most to lateralization. A more detailed consideration of this "binaural aperture" sheds light on how ambiguous lateralization data emerge from limited ensembles of neurons constrained to physiologically relevant ranges of ITD and frequency. [R01-DC011555 (DJT), R01-DC014948 (MJG), and R01-DC016643 (GCS).]

4pPP18. Comparison of click and chirp stimuli in evoking the binaural interaction component of the auditory brainstem response. Zoe Owrutsky (Univ. of Colorado, 1196 N. Grant St., Apt. 501, Denver, CO 80203, zoe.owrutsky@cuanschutz.edu) and Daniel Tollin (Univ. of Colorado, Aurora, CO)

Chirp stimuli enhance synchrony in the auditory nerve by compensating for frequency- and level-dependent delays accrued by the traveling wave in the cochlea. Chirps produce larger amplitudes in specific waves of the monaural auditory brainstem response (ABR). We test whether chirps increase the amplitude of the binaural interaction component (BIC) of the ABR, a potential biomarker for binaural hearing. ABR latencies were measured across multiple stimulus frequencies and intensities in chinchillas. Tone bursts (1–16 kHz) were presented ~10 dB below to 50 dB above threshold. Latencies of ABR waves I-IV and BIC were plotted against frequency and fit to a power function: Tau = k*f(-d), where tau is latency, f is frequency, and k and d are constants. The values of k and d were used to construct chirps for three intensity-levels based on monaural ABR waves I and IV, and BIC. Monaural ABR latencies decreased with increasing frequency as expected from cochlear delay. Monaural ABR amplitudes were enhanced by chirps relative to clicks. Surprisingly, BIC DN1 latencies did not show a similar shift but rather appeared constant across frequency. Traditional chirp stimuli appear ill-suited for eliciting optimal BIC amplitudes. Traditional broadband clicks or non-traditional chirps may be preferable for evoking the BIC.

4pPP19. The effects of noise exposure and aging on the acoustic reflex in normal-hearing people. Ward R. Drennan (Otolaryngol., Univ. of Washington, VM Blodel Hearing Res. Ctr., UW Box 357923, Seattle, WA 98195, drennan@uw.edu), Lauren Langley (Otolaryngol., Univ. of Washington, Seattle, WA), and Zeyu Wei (Statistics, Univ. of Washington, Seattle, WA)

Noise exposure and aging can cause subclinical hearing problems that are not identified with conventional audiometry. Nerve damage among the normal hearing could reduce afferent auditory-nerve activity consequently reducing the middle ear muscle reflex (MEMR). It was hypothesized that normal-hearing individuals reporting more noise exposure would have slower growth of MEMR with increasing levels. Data were analyzed from sixty-one young, *normal-hearing* people. Each completed the four MEMR assessments and the Life-Time Exposure to Noise and Solvents Questionnaire. Reflex amplitudes were measured as a function of sound level using clinical equipment. High-pass noise stimuli (1.5-4.0 kHz) and a 1-KHz tone were used as elicitors with a 226-Hz probe tone. Data were analyzed using a linear mixed effects model. Individuals reporting higher noise exposure with the LENS-Q had significantly reduced growth of MEMR with level, supporting the hypothesis. Independent effects of age and gender were also observed. The effect of the pure-tone average on MEMR growth functions did not reach significance at the 5% level (p = 0.052). No significant relationships were observed between the MEMR measures and extended high-frequency thresholds nor with word recognition in noise. The results suggest that the MEMR could indicate subclinical nerve damage.

THURSDAY AFTERNOON, 26 MAY 2022 PLAZA EXHIBIT HALL, 1:00 P.M. TO 5:00 P.M.

Session 4pSC

Speech Communication: Speech Perception I (Poster Session)

Emily Grabowski, Chair
UC Berkeley, Dwinelle Hall, Berkeley, CA 94710

All posters will be on display from 1:00 p.m. to 5:00 p.m. Authors of odd-numbered papers will be at their posters from 1:00 p.m. to 3:00 p.m. and authors of even-numbered papers will be at their posters form 3:00 p.m. to 5:00 p.m.

Contributed Papers

4pSC1. Recognition of English vowels, consonants, words, and sentences for Chinese college students. Junbo Liu (Woodbridge High School, Irvine, CA), Lihong Wang, Wei Hu, Libo Qiao (Tianjin Normal Univ., Tianjin, China), Sha Tao (Beijing Normal Univ., Beijing, China), and Chang Liu (Univ. of Texas at Austin, 2504A Whitis Ave., Austin, TX 78712, changliu@utexas.edu)

The goals of this study were to first, investigate the recognition of several types of English speech sounds such as vowels, consonants, words, and sentences in quiet and long-term speech-shaped (LTSS) noise for Chinese college students; second, examine relationship among the perception of these speech materials, e.g., whether students with better sentence recognition also showed better phonemic perception. Speech stimuli included twelve isolated English vowels, twenty-one English consonants in the /aCa/ context, NU6 words, the coordinate response measure (CRM), Hearing in noise test (HINT) sentences, and IEEE sentences. Preliminary data analysis indicated that at the phonetic level, Chinese students performed significantly better in consonant identification than in vowel identification; at the word level, students did significantly with CRM than NU6; and at the sentence level, students had significantly higher recognition scores for HINT sentences than for IEEE sentences. The correlations among the recognition of these speech materials will examined and discussed. [Work supported by China National Natural Science Foundation 31628009.]

4pSC2. Perception and expression of talker gender at the beginning and end of words. Matthew B. Winn (Speech-Language-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr. SE, Shevlin Hall Rm. 115, Minneapolis, MN 55455, mwinn@umn.edu) and Benjamin Munson (Univ. of Minnesota, Minneapolis, MN)

Among the common acoustic differences between the voices of women and men are the spectral characteristics of fricatives. Akin to pitch and formant frequencies, the frequency characteristics of fricatives are lower in men than in women. These fricative characteristics are learned social cues that help to express the talker's gender. Listeners reliably incorporate these differences into perceptual judgments by categorizing the same sound differently depending on what they perceive the gender of the talker to be; a fricative that is intermediate to "s" and "sh" will be labeled "s" when thought to be spoken by a man, but "sh" if by a woman. However, the acoustic descriptions and the perceptual consequences of these gender-related acoustic differences have been explored almost exclusively at syllable-onset position. Here we use both word-onset and -offset positions in both perception and production tasks. The perceptual accommodation of talker gender is weaker when the fricative is at syllable offset, inviting questions about whether the extent of gender-related changes in acoustics are simply commensurate with salience of word position. We will present both the perception and production data in an attempt to unify both phenomena in relation to each other.

4pSC3. Perception and imitation of prevoicing across language backgrounds. Emily J. Clare (Univ. of Toronto, Mississauga, ON, Canada) and Jessamyn Schertz (Univ. of Toronto, 3359 Mississauga Rd., Mississauga, ON L5L 1C6, Canada, jessamyn.schertz@utoronto.ca)

The present study investigates languages users' ability to perceive and imitate prevoicing, testing speakers of languages differing its contrastive status: (1) those where prevoicing serves as a primary cue to the laryngeal contrast (e.g., Portuguese); (2) those where prevoicing never occurs, with aspiration being the primary cue to the contrast (e.g., Cantonese); and (3) English, where prevoicing occurs in free variation. After hearing pairs of words differing in presence/absence of prevoicing, participants were asked to imitate them, followed by an ABX discrimination task. Preliminary findings show above-chance discrimination accuracy as well as significant imitation of the difference, in participants all language backgrounds, with place of articulation, consonantal, and vocalic cues all affecting the extent of prevoicing in production. As expected, speakers of languages where prevoicing is absent showed less overall prevoicing than other groups; however, there were not clear differences across language groups in the faithfulness of imitation, or in discrimination ability. While faithful imitation was usually

associated with accurate discrimination, good discrimination was also found on many tokens that were not faithfully imitated, indicating that factors other than perception are necessary to account for variability in imitative ability.

4pSC4. Syllable inference as a mechanism for spoken language understanding. Laura Dilley (Communicative Sci. and Disord., Michigan State Univ., 1026 Red Cedar Rd., East Lansing, MI 48824, ldilley@msu.edu)

A classic problem in spoken language comprehension is how speech is perceived as comprised of discrete words. A Syllable Inference account of spoken word recognition and segmentation is proposed, according to which alternative hierarchical models of syllables, words, and phonemes are dynamically posited. Estimates of context speech rate are combined with generative models, such that over time, models which result in local minima in error between predicted and recently experienced signals give rise to perceptions of hearing words. Evidence for this account comes from experiments using the visual world eye-tracking paradigm. Materials were sentences that were acoustically ambiguous in numbers of syllables, words, and phonemes they contained. Time-compressing, or expanding, speech materials permitted determination of how temporal information at, or in the context of, each locus affected looks to, and selection of, pictures with a singular or plural referent. Supporting our account, listeners probabilistically interpreted identical chunks of speech as consistent with a singular or plural referent to a degree that was based on the chunk's gradient rate in relation to its context. These results support the Syllable Inference account that arriving temporal information informs inferences about syllables, giving rise to perception of words separated in time.

4pSC5. Differential effects of pitch on perceived duration in linguistic and auditory tasks. Emily Grabowski (UC Berkeley, Dwinelle Hall, Berkeley, CA 94710, emily_grabowski@berkeley.edu)

Duration perception is a complex phenomenon in linguistic studies. While it can be instrumentally measured in a straightforward fashion, its relationship to other acoustic measurements is still not well understood. Previous work has found that duration can influence the perception of other acoustic cues, such as pitch (Brigner, 1988). This implies that it is important to go beyond isolated analysis of individual acoustic cues, and consider the synergistic effects of multiple acoustic cues. This paper presents the results of an experiment designed to test the influence of changing pitch on duration in two tasks, one using linguistic stimuli and one using acoustic stimuli. I find an effect of pitch on duration perception in both tasks. However, the results are not identical. In the auditory task, there was a small effect where higher tones were heard as longer than low tones. However, in the linguistic task, higher tones were heard as shorter. This indicates that pitch can have both a direct and indirect effect on duration perception and that there are complex interactions that may have a significant impact on our language perception.

4pSC6. The effect of second language experience on lexical encoding of segments and tones. Yen-Chen Hao (Modern Foreign Lang. and Literatures, Univ. of Tennessee, 701 McClung Tower, 1115 Volunteer Blvd., Knoxville, TN 37996-0470, yenchenhao@gmail.com)

While previous studies generally find that increasing second language (L2) experience contributes to more accurate phonetic perception of L2 sounds, whether it leads to more proficient lexical encoding remains an empirical question. The current study examines lexical encoding of Mandarin segments and tones by English speakers at different proficiency levels. Eleven English speakers naïve to Mandarin, fifteen intermediate and nine advanced L2 learners participated in a word-learning experiment. After learning 16 Mandarin disyllabic words, they judged the matching between sound and meaning pairs, with half of the pairs being complete matches while the other containing segmental or tonal mismatches. The results showed that all groups were more sensitive to segmental than to tonal mismatches. The two learner groups outperformed the Naïve group in detecting segmental mismatches, but the three groups were equally inaccurate in rejecting tonal mismatches. The reaction times revealed, however, that the learners but not the Naïve group attended to tonal variations. These findings suggest that L2 experience has a clearer benefit for L2 segmental encoding than for tonal encoding, probably due to learners' non-tonal language background. Experience in a tonal L2 enhances learners' attention to the tones, but does not necessarily improve their accuracy in tonal encoding.

4pSC7. Perception of gender in children's voices. Christopher E. Holt (Speech and Hearing Sci., The Ohio State Univ., 1070 Carmack Rd., Columbus, OH 43210, holt.465@osu.edu), Robert A. Fox, Ewa Jacewicz, and Lian J. Arzbecker (Speech and Hearing Sci., The Ohio State Univ., Columbus, OH)

Research shows that gender in the voices of prepubescent children can be identified relatively well, despite the absence of reliable anatomical differences related to their vocal tract morphology. This implies that information about differences in girls' and boys' voices also resides in behavioral and cultural aspects of speech and not exclusively in the acoustic cues corresponding to anatomical differences. The observed improved gender identification (gID) for sentences over syllables suggests that longer stretches of speech provide listeners with a richer acoustic basis for their decisions. Here, we introduce two additional factors that may affect gID: speaking style and children's dialect. Children ($n=92$) ages 8;0–12;4 years, boys ($n=45$) and girls ($n=47$) from Ohio, Wisconsin, and North Carolina produced isolated syllables, read sentences, and spontaneous utterances. Listeners were young adults from Ohio. Expectedly, gID accuracy for sentences/utterances was greater (66%) than for syllables (55%), and for older than younger children. Dialect was a significant predictor for younger (but not older) children, with the highest gID for Ohio. Overall, spontaneous utterances did not provide more gender information than read sentences except for older children from North Carolina, suggesting that cultural traits in conversational dynamics can also be useful to listeners in gID.

4pSC8. Perceptual assessment of contrastive voicing in 2-year-old children's speech by adult and school-age child listeners. Elaine R. Hitchcock (Commun. Sci. & Disord., Montclair State Univ., 1515 Broad St., Attn: Elaine Hitchcock, Bloomfield, NJ 07003, hitchcocke@montclair.edu) and Laura Koenig (Haskins Labs, New Haven, NY)

Past research assessing perception of voice onset time (VOT) in stops suggests that perceptual acuity improves as a function of age, with adults showing steeper labeling functions and narrower boundary widths. Most work has used synthetic stimuli varying a single acoustic cue, so extending such findings to natural speech may not be straightforward. Perceptual judgments may also be influenced by distributional properties of the dataset, e.g., VOT distributions along a continuum and/or the number of productions in each VOT category. This study will assess how distributional characteristics of naturally produced child speech stimuli, collected from six 2–3-year-old English-speaking children, might influence adult and child labeling behavior. Six exemplars per child were chosen with short-lag /b d/, short-lag /p t/, long-lag /b d/, long-lag /p t/. For each POA and VOT group, /b d/ and /p t/ VOTs were bimodally distributed (shorter for voiced targets), separated by a 5 ms gap. We will seek listening data from 20 adults and 20 children (aged 6–8). We anticipate high and similar accuracy levels for both groups when VOT values are appropriate for the target, but clearer group differences for inappropriate VOTs.

4pSC9. Distal context rate effects attenuated by attention. Christopher C. Heffner (Communicative Disord. and Sci., Univ. at Buffalo, 122 Cary Hall, South Campus, Buffalo, NY 14214, ccheffne@buffalo.edu)

Listeners use the rate of speech at the beginning of a sentence to segment ambiguous words that occur later in a sentence. For instance, the sentence "the drab mermaids will join the clean medics smoothly" can also be perceived as ending with "medic smoothly" if the words at the sentence onset are slowed down, showing the effects of what has been termed distal context rate. Here, I report a set of studies that assess the extent to which distal context effects are subject to modulation by attention. We manipulated two types of attention: explicit attention (in the form of overt task instructions) and implicit attention (employing unnatural F0 contours to attract or distract). These were designed to focus listeners on or divert listeners from the word segmentation ambiguities present in the signal. The

findings suggest that listeners who were cued to attend more to the word segmentation ambiguities were less affected by distal context effects than listeners who were cued to shift their attention away. This suggests that listeners who were cued to apply more "perception-oriented" attention (cf. McAuliffe and Babel, 2016) to experimental items may be less affected by context speech rate than those cued otherwise.

4pSC10. Perception and timing of acoustic distance. Matthew C. Kelley (Linguist., Univ. of Washington, GUG 407A, Seattle, WA 98195, mattck@uw.edu) and Benjamin V. Tucker (Linguist, Univ. of AB, Edmonton, AB, Canada)

The notion of acoustic distance figures into many aspects of phonetics, including phonological neighborhoods. A measurement of word-level acoustic distance useful for cognitive modeling must account for two listener characteristics: sensitivity to acoustic differences and sensitivity to duration discrepancies between words. The present work used dynamic time warping to measure how acoustic distance accumulates between words over time. The results of a distance rating task with synthesized vowels were used as a basis for selecting a mathematical function that best matched listener sensitivities. Additionally, the results of a reminder task with synthesized vowels were used to determine a just noticeable difference threshold for vowel duration. The results suggested that a distance function based on the 4.5-norm using a 30 ms radius for dynamic time warping best matched human behavior. A third analysis used these dynamic time warping configurations to model reaction times in an auditory lexical decision task and found that Euclidean distance and no temporal constraints on dynamic time warping best matched human behavior. These results are discussed in relation to spoken word recognition models, including how to assess the acoustic match between the speech signal and a word in the lexicon.

4pSC11. Phonological but not lexical processing alters the perceptual weighting of mean fundamental frequency and vocal-tract length cues for voice gender categorisation. Laura Rachman (Otorhinolaryngology/Head and Neck Surgery, Univ. Medical Ctr. Groningen, Hanzeplein 1, Groningen 9713 GZ, the Netherlands, l.rachman@rug.nl), Almut Jebens, and Deniz Baskent (Otorhinolaryngology / Head and Neck Surgery, Univ. Medical Ctr. Groningen, Univ. of Groningen, Groningen, the Netherlands)

Listeners use various voice cues to segregate different speakers, or to infer speaker-related information such as perceived gender. Two important anatomically related voices cues used for speaker identification, including perceived gender, are mean fundamental frequency (F0), related to the glottal pulse rate, and vocal-tract length (VTL), correlating with body size. Voice cue processing seems to be affected by linguistic processes, such that voice perception is more precise when listeners hear speakers in their native language compared to a non-native language. In addition, recent research shows that F0 and VTL sensitivity is lower for words compared to time-reversed words, either because time-reversed words are unintelligible or phonemes are distorted in voice-onset times and aspirations, pointing to effects of lexical or phonological processing. However, voice cue sensitivity and using these cues to infer speaker-related information may rely on different mechanisms. Here, we studied effects of lexical and phonological processing on F0 and VTL cue weighting for one aspect of speaker identification, namely voice gender categorisation, by manipulating these cues in three linguistic conditions: meaningful words; phonotactically plausible nonwords; and phonotactically implausible time-reversed nonwords. We found that F0 and VTL weighting for voice gender categorisation was affected by phonological but not by lexical processing.

4pSC12. Dominance patterns in concurrent vowel recognition. David Edwards (Linguist & TESOL, Univ. of Texas at Arlington, Box 19559 — 132 Hammond Hall, Arlington, TX 76019-0559, david.edwards@mavs.uta.edu)

When hearers encounter multiple people speaking at the same, they can often pick out the speech of one person from the others, but the mechanisms which they use in their acoustic discrimination process are not fully understood. To look into vowel discrimination processes, this project entailed combining sound files of minimal pairs generated by synthetic speakers. As a point of departure from many similar experiments, these files were standardized for pitch and intensity. Participants were asked to select the word they heard from a list of four potential words that shared the same beginning and ending consonants, differing only in the vowel. Participants were significantly more likely to select a word if its vowel had a higher F1 than the F1 of the other vowel of the combined audio file. Higher F2 and F3 frequencies and greater movement of F1 also contributed to increased selection rates, although not as much as the F1 ratio. Participants were also asked to describe a subset of the combined sound files. That task showed similar dominance patterns, such that participants were often unaware of the presence of one of the concurrent vowels.

4pSC13. Judgments of gay and heterosexual male speakers of American English: Which consonants do listeners rely on to form their sexual orientation judgments? Erik C. Tracy (Psych., Univ. of North Carolina Pembroke, PO Box 1510, Pembroke, NC 28372, erik.tracy@uncp.edu)

Prior research demonstrated that, upon hearing a single phone, listeners differentiated between gay and heterosexual male talkers of American English. For instance, researchers found that listeners relied on three consonants (e.g., /l/, /n/, and /s/) to form their judgments. It is unclear whether these findings could be replicated and whether there are additional consonants that listeners used. To further explore this, 23 consonants were examined (e.g., /b/, /v/, /s/, /n/, /t/, /f/, /ɛr/, /dʒ/, /θ/, /k/, /ɹ/, /m/, /l/, /g/, /d/, /ʒ/, /w/, /p/, /ɛl/, /ʃ/, /z/, /j/, and /ð/). For 21 of these consonants, listeners heard two tokens each, and heard one token of /θ/ and /ð/. Results were mixed. Listeners relied on both instances of /s/, /w/, /ɛl/, and /z/ to form their judgments, while they relied on only one instance of /b/, /n/, /ɛr/, /dʒ/, /k/, /ɹ/, /l/, /d/, /ʒ/, /p/, and /j/. Listeners didn't rely on any instances of /v/, /t/, /f/, /g/, /m/, and /ʃ/; they relied on /ð/ and did not rely on /θ/. The only consistent finding, across multiple experiments, is that participants used /s/. Under certain circumstances, they may use additional consonants, but it doesn't appear as if they use them consistently.

4pSC14. Clearly, fame isn't everything: Talker familiarity does not mitigate the perceptual consequences of talker variability. Emma R. Hatter (Psychol. and Brain Sci., Univ. of Louisville, Louisville, KY), Caleb J. King (Psychol. and Brain Sci., Univ. of Louisville, 317 Life Sci. Bldg., University of Louisville, Louisville, KY 40292, caleb.king.1@louisville.edu), Anya E. Shorey, and Christian E. Stilp (Psychol. and Brain Sci., Univ. of Louisville, Louisville, KY)

Familiarity with a talker's voice provides numerous benefits to speech perception, including faster response times and improved intelligibility both in quiet and amidst competing talkers. However, it is unclear whether familiarity provides resilience against talker variability, or the increased processing costs attributed to hearing speech from multiple different talkers compared to a single talker. Here, listeners completed a speeded word recognition task where stimuli were the words "do" and "to" excised from political speeches, presented in either single- or multiple-talker blocks. Talkers were either famous (the last five U.S. Presidents) or non-famous (other male politicians of similar ages). No information about the talkers was provided. As predicted, listeners categorized words more quickly when spoken by famous talkers. However, talker familiarity did not provide any resiliency to talker variability; reaction times increased (from single- to multiple-talker blocks) by greater amounts for famous talkers than non-famous talkers. In a post-task questionnaire, participants recognized famous talkers better than non-famous talkers from the "do"/"to" target words, a full sentence, or when shown their names and asked if they knew who the talker was. Results were not correlated with self-rated political interest. Thus, familiarity might not alleviate the perceptual consequences of talker variability.

4pSC15. Talker and sentence variability attenuates speaking rate normalization in speech perception. Caleb J. King (Psychol. and Brain Sci., Univ. of Louisville, 2301 S. 3rd St., Louisville, KY 40292, cjking03@louisville.edu), Chloe M. Sharpe (Xavier Univ., Cincinnati, OH), Anya E. Shorey, and Christian E. Stilp (Psychol. and Brain Sci., Univ. of Louisville, Louisville, KY)

Speech perception is shaped by acoustic context effects, where acoustic properties of earlier (context) sounds influence categorization of later

(target) sounds. These context effects are sensitive to talker characteristics. When context sentences were spoken by a different talker on each trial, spectral contrast effect (SCE) magnitudes affecting categorization of the target vowel were smaller than when sentences were spoken by a single talker [Assgari and Stilp, *J. Ascoust. Soc. Am.* (2015)]. Here, perceptual consequences of talker variability were examined in Temporal Contrast Effects (TCEs; also termed rate normalization). Each trial presented a context sentence (spoken at a fast or slow rate) followed by a target word varying from "deer" to "tier." Context sentences were the same stimuli from Assgari and Stilp (2015): one talker speaking the same sentence on each trial, one talker speaking a different sentence on each trial, and 200 talkers each speaking a different sentence on each trial. TCE magnitudes significantly decreased between One Talker/One Sentence and One Talker/200 Sentences, and again between One Talker/200 Sentences and 200 Talkers/200 Sentences. Conversely, in Assgari and Stilp, talker variability diminished SCEs but sentence variability for a single talker did not. These results suggest that variability in different acoustic domains restricts speech perception.

4pSC16. Rapid speech adaptation in younger and older normal-hearing adults: Distributional and lexically guided learning. Ruoqian Cheng (Dept. of Linguist., Univ. of Kansas, 1541 Lilac Ln., University of Kansas, Lawrence, KS 66045, rqcheng@ku.edu) and Allard Jongman (Linguist, Univ. of Kansas, Lawrence, KS)

In speech perception, when a primary acoustic cue (e.g., VOT) is ambiguous, listeners may increase the weight of a secondary cue (e.g., F0). In experiment 1, we compared the cue-weighting adjustment strategies across younger and older normal-hearing adults with a distributional learning paradigm. Two groups of native English listeners were exposed to voicing contrasts that were ambiguous in either VOT or F0. Additionally, listeners may access lexical information to help resolve the ambiguity in the acoustic signal. Older listeners have been reported to use lexical information to a greater degree than younger listeners. In experiment 2, using a lexically guided learning paradigm, we tested if younger and older adults differ in their use of lexical information when learning to interpret ambiguous acoustic tokens. There were four types of exposure, in which stimuli differed in lexical status (*day-*tay*; **doy-toy*) and the acoustic ambiguity involved either only VOT or both VOT and F0. Preliminary results from younger normal-hearing, listeners showed significant speech adaptation effects, with a significant change in cue weights in distributional learning and salient lexical bias in lexically guided learning. More data will be collected from older adults to assess the extent of perceptual learning relative to younger adults.

4pSC17. Does having control over background noise affect listening effort or speech perception? Alexander L. Francis (Speech, Lang. & Hearing Sci., Purdue Univ., Lyles-Porter Hall, 715 Clinic Dr., West Lafayette, IN 47907, francisa@purdue.edu), Nicole Kirk, Erin Frain, Kirsten Fong (Speech, Lang. & Hearing Sci., Purdue Univ., West Lafayette, IN), Paola Medina Lopez (Univ. of Puerto Rico, Mayagüez, Puerto Rico), Jordan Love (SLHS, Purdue Univ., West Lafayette, IN), and Jane E. Clougherty (Drexel Univ., Philadelphia, PA)

We have previously shown that individual personality traits, including noise sensitivity, can influence listeners' assessment of effort and frustration related to noise that interferes with a listening task. Here, we extend our previous research to employ stimuli from a commonly used speech-in-noise task, the coordinate response measure (CRM). On each trial, participants heard a sentence of the form "Ready, CALLSIGN, go to COLOR NUMBER now" presented in a background of steady-state speech-shaped noise. They responded by selecting from a multiple-choice display (two callsigns, four colors, and four numbers). After completing one block of 64 trials with 32 sentences presented at a -4 dB SNR and 32 at -8 dB SNR, half of the participants were allowed to choose to change the noise level, making it either "louder" or "softer" for the next 64 trials. The other participants were assigned to either a louder or softer SNR without a choice. Importantly, half of the trials in the "louder" and "softer" conditions were presented at the same SNR (−6 dB), permitting comparison of performance independently of chosen or assigned difficulty. Results will be discussed in terms of implications for future research on noise sensitivity and long-term health.

4pSC18. Influences of talker gender on second language speech perception. Yang Zhang (Speech-Language-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, zhang470@umn.edu), Keita Tanaka (Tokyo Denki Univ., Ishizaka, Hiki-gun, Saitama, Japan), and Toshiaki Imada (Ctr. for Frontier Medical Eng., Chiba Univ., Seattle, WA)

This study examined how talker gender information may influence second language speech perception. The participants were ten right-handed male Japanese adults with normal hearing. The speech stimuli were two minimal pairs of English words, "heed-hod" and "head-had", recorded from twenty male and twenty female native English speakers. The stimuli were delivered binaurally at a sensation level of 50 dB. In the gender recognition task, the subjects were asked to judge talker gender. In the vowel recognition task, they were asked to identify the vowel category. Magnetoencephalography data were recorded during the experiment. For gender recognition, there was no significant difference in identification accuracy between the two word pairs while reaction time was significantly faster for the difficult "head-had" pair. In the phoneme recognition task, there was a significant phoneme*gender interaction for the difficult "head-had" pair but not for the easy "heed-hod" pair. The MEG data for gender recognition showed right hemisphere dominance, and the phoneme recognition task showed overall bilateral activation. Time-frequency analysis further revealed distinctions between male and female voices and the difficulty differences for the two word pairs in the phoneme recognition task. The results suggest a complex role of paralinguistic gender information in L2 speech perception.

4pSC19. The perception of synthetic speech versus natural speech stimuli in adult listeners. Kathryn Cabbage (Commun. Disord., Brigham Young Univ., 1190 N. 900 E, TLRB 161, Provo, UT 84604, kcabbage@byu.edu), Thomas Carrell (Commun. Disord., Univ. of Nebraska Lincoln, Lincoln, NE), Melannee Ipsen (Commun. Disord., Brigham Young Univ., Provo, UT), and Elaine R. Hitchcock (Commun. Sci. & Disord., Montclair State Univ., Bloomfield, NJ)

The majority of speech perception research investigates interpersonal perception through presentation of two types of stimuli: synthetic speech and/or natural speech. Synthetically created speech stimuli allows for precisely controlled manipulation of individual acoustic variables, but can sound unnatural calling into questions its ecological validity. Alternatively, natural speech preserves the full complement of redundant acoustic cues yet limits control of specific acoustic parameters. In this study, we investigate speech discrimination comparing perception of specified acoustic parameters (formant transition duration, F3 onset frequency, F3-F2 distance) across three types of speech stimuli: synthetic speech stimuli, natural speech of a standard adult speaker, and each participant's own voice. We use the Wide Range Acoustic Accuracy Scale (WRAAS), a computer-based perceptual assessment that uses a parameter estimation by sequential testing (PEST) adaptive-tracking algorithm to rapidly identify a listener's ability to discriminate sounds. Most trials are presented near the Just-Noticeable-Difference (JND) of a stimulus continuum; thus, few trials are "wasted" with obviously same or different sounds. This study of typical adult listeners (n = 20, enrollment is ongoing) will provide baseline data for future work investigating perception of these same stimuli in children.

4pSC20. Do individual differences in lexical reliance reflect states or traits? Nikole Giovannone (Univ. of Connecticut, 2 Alethia Dr., Unit 1085, Storrs, CT 06269, nikole.giovannone@uconn.edu) and Rachel M. Theodore (Univ. of Connecticut, Storrs, CT)

Individual differences in listeners' reliance on lexical information for speech perception have been established in the literature. However, it is not yet clear whether these individual differences reflect a context-dependent "state" or whether they reflect a stable listener "trait." To examine this question, listeners completed three measures of lexical processing and one measure of acoustic-phonetic processing at two time points (n = 80 at session 1; n = 41 at session 2). Results to date suggest limited within-session convergence among the three measures of lexical processing, inconsistent with past results. However, strong between-session relationships for two of the three measures of lexical processing and the measure of acoustic-phonetic processing and were found, suggesting that performance in these tasks may reflect stable individual traits. A strong trading relationship between reliance

on lexical versus acoustic-phonetic information was only found when both sources of information were available in a given task, suggesting that individual differences in cue use may emerge most strongly when cues conflict in speech input. Collectively, the current results suggest that individuals are stable in their acoustic-phonetic and lexical cue use strategies for a given task, but that an individual's cue use strategy may be task-dependent.

4pSC21. Effects of noise, native language, age, and speaker gender on intelligibility in a large corpus of read speech. Richard A. Wright (Dept. of Linguist., Univ. of Washington, Box 352425, Seattle, WA 98195-2425, rawright@uw.edu), Benjamin V. Tucker (Linguist., Univ. of AB, Edmonton, AB, Canada), Matthew C. Kelley, and Marina Oganyan (Linguist., Univ. of Washington, Seattle, WA)

Because speech communication takes place under noisy conditions, investigating the effect of noise on intelligibility is an essential part of understanding speech perception. One challenge is having sufficient numbers of speakers and unique sentences to control for sentence and speaker effects. In an online experiment, we used a corpus of 720 IEEE sentences read by 20 native English speakers (10 male, 10 female) from the Pacific Northwest (WA, OR, ID), which were embedded in three corpus-shaped noise conditions (−2, 0, + 2 dB). Stimuli were presented to undergraduate students at the Universities of Alberta and Washington using a design matrix which ensured that no listener heard a sentence more than once. Participants entered responses into a form which appeared immediately after the stimulus completed playing. Data collection is ongoing, but the current number of listeners is 1269, who responded to 120 items each. We use Levenshtein and Jaro distance measures to compare listener transcription accuracy. We investigate the effects of listener native language, age, and gender. We also investigate effects of the speaker's gender on transcription accuracy. [Work supported by NIH NIDCD R01 DC006014.]

4pSC22. Effects of word context on formant discrimination. Alyssa Strickler (Linguist, Univ. of Colorado Boulder, Boulder, CO) and Rebecca Scarborough (Dept. of Linguist., Univ. of Colorado Boulder, 295 UCB, Boulder, CO 80309, rebecca.scarborough@colorado.edu)

Studies of phonetic variation identify systematic, but often very small, acoustic differences in the realization of vowels. However, it is not known whether such differences are perceptually relevant in realistic contexts. We examine perceptibility of incrementally manipulated differences in F1 and F2 in vowels embedded in real words of English (experiment 1), and vowels in isolation (experiment 2), via an AX discrimination task. Forty listeners in each experiment, recruited through Amazon Mechanical Turk, heard 27 monosyllabic words or vowels in pairs containing a no-change token and a token edited to differ up to ±100 Hz in F1 or ±150 Hz in F2 relative to the original and were asked to say whether tokens were same or different. Discrimination increased with bigger vowel differences (both F1 and F2) in both experiments; however, even maximally different tokens were correctly discriminated less than half the time. F2 discrimination was better in isolated vowels (48%) than in whole words (33%). Compared to previous results in optimized difference limen studies (e.g., Kewley-Port and Watson, 1994), our results show that untrained listeners in more typical listening conditions (non-synthetic speech heard in non-laboratory conditions in real word contexts) require much bigger differences to be perceptible.

4pSC23. What mechanisms underlie phonetic and phonological learning ability in monolingual and bilingual populations? A closer look at the connection with cognitive and sensory functions. Laura Spinu (Communications & Performing Arts, City Univ. of New York-Kingsborough Community College, 2001 Oriental Blvd. Rm. E329, Brooklyn, Brooklyn, NY 11235, laura.spinu@kbcc.cuny.edu), Laura M. Muscalu (Psych., Hobart and William Smith Colleges, Ithaca, NY), Mariana Vasilita, Crystal Gilbert (Commun. Sci. and Disord., Brooklyn College, Brooklyn, NY), and Julia Wallace (Communications & Performing Arts, City Univ. of New York-Kingsborough Community College, Brooklyn, NY)

Despite considerable controversy surrounding the "bilingual advantage" phenomenon, recent meta-analyses reveal a majority of studies on the topic have reported that bilinguals outperform monolinguals on tasks related to executive functioning, such as mental flexibility and inhibitory mechanisms. A more recent line of investigation also shows that bilinguals typically score higher than monolinguals on tasks involving phonetic and phonological learning as well as auditory sensory memory. In this study, we aim to gain insight into how these different functions are related by exploring the correlations between four cognitive tasks in 30 bilinguals and 30 monolinguals. The tasks were designed to test: (1) inhibition; (2) mental flexibility; (3) phonetic and phonological learning (PPL); and (4) auditory sensory memory (ASM). All the participants were tested and data analysis is currently underway. Based on existing literature, we hypothesize enhanced bilingual performance on all of these tasks, as well as a connection between PPL and ASM. While the literature does not allow for a straightforward prediction, we will also investigate whether executive functions play a role in PPL and are correlated with ASM. Our study will thus lead to better understanding of the coupling between sensory and cognitive functions.

4pSC24. Cents and Shenshibility: The role of reward in talker-specific phonetic recalibration. Hannah Mechtenberg (Univ. of Connecticut, 247 Hanks Hill Rd., Storrs Mansfield, CT 06268, hannah.mechtenberg@uconn.edu), Sahil Luthra (Carnegie Mellon Univ., Pittsburgh, PA), and Emily B. Myers (Univ. of Connecticut, Storrs, CT)

We, as listeners, encounter many challenges during speech comprehension that extend far beyond resolving ambiguities in the acoustic phonetic signal. An understudied component of naturalistic speech comprehension is how motivated a listener is to attend to various aspects of the speech signal. This includes, when listening to multiple novel talkers, whether we give talkers equal attention when adapting to their idiosyncratic productions. We asked if listener adaptation is influenced by an external reward that is intended to bias attention towards one talker versus the other. In a lexically guided perceptual learning paradigm, participants heard two talkers of different genders—"Jane" and "Austin"—with idiosyncratic productions of the /s/ and /ʃ/ fricatives. Participants were more likely to receive a small monetary reward for one talker than the other following correct responses during a phoneme-monitoring cover task. We hypothesized that participants would show greater phonetic recalibration for the more-rewarded talker than the less-rewarded talker; however, our results indicate that external reward may undermine learning for both talkers. The role of reward and motivation in talker-specific phonetic recalibration raises interesting questions for how these domain-general factors influence language processing more broadly.

4pSC25. Pupil response to familiar and unfamiliar talkers in the recognition of noise-vocoded speech. Terrin N. Tamati (Dept. of Otolaryngol., Ohio State Univ., 1608 Aschinger Blvd., Columbus, OH 43212, terrintamati@gmail.com), Lars Bakker, Stefan Smeenk, Almut Jebens, Thomas Koelewijn, and Deniz Başkent (Dept. of Otorhinolaryngology/Head and Neck Surgery, Univ. Medical Ctr. Groningen, Groningen, the Netherlands)

In some challenging listening conditions, listeners are more accurate at recognizing speech produced by a familiar talker compared to unfamiliar talkers. However, previous studies have found little to no talker-familiarity benefit in the recognition of noise-vocoded speech, potentially due to limitations in the talker-specific details conveyed in noise-vocoded signals. Although no strong effect on performance has been observed, listening to a familiar talker may reduce the listening effort experienced. The current study used pupillometry to assess how talker familiarity could impact the amount of effort required to recognize noise-vocoded speech. Four groups of normal-hearing, listeners completed talker familiarity training, each with a different talker. Then, listeners repeated sentences produced by the familiar (training) talker and three unfamiliar talkers. Sentences were mixed with multi-talker babble, and were processed with an 8-channel noise-vocoder; SNR was set to a participant's 50% correct performance level. Preliminary results demonstrate no overall talker-familiarity benefit across training groups. Examining each training group separately showed differences in pupil response for familiar and unfamiliar talkers, but the direction and size of the effect depended on the training talker. These preliminary findings suggest that normal-hearing, listeners make use of limited talker-specific details in the recognition of noise-vocoded speech.

4pSC26. Categories in speech perception and production: How closely are they related? Kuniko Nielsen (Linguist, Oakland Univ., Human Health Bldg., 433 Meadow Brook Rd., Rochester, MI 48309-4452, nielsen@oakland.edu)

A large body of research shows that fine phonetic details are not only used by listeners in processing speech (e.g., McMurray *et al*., 2009), but these details also affect listeners' subsequent speech productions (e.g., Goldinger, 1998), suggesting a robust link between speech perception and production. However, evidence for a direct perception-production link is mixed, and the relationship between categories in perception and production is largely unknown. Newman (2003) found correlation between a perceptual prototype and mean VOT production for /pa/, while there were no production-perception correlations for other stop consonants. Nielsen (2021) found that perceptual boundaries in VOT vary widely across speakers, but that there is no apparent link between categorical boundary in perception of voicing contrast ([p]-[b]) and production variables of isolated speech (e.g., mean VOT for /p/ or /b/, the center of gap between two categories, the distance between the two category means), confirming Bailey and Haggard (1973). The current study further explores the relationship between perception and production categories by examining categorical boundaries of English stops using minimal pairs (e.g., pear-bear), their perceptual prototypes through a goodness rating task, and production variables in connected and isolated speech.

4pSC27. Perceptual consequences of clear speech. Tifani Biro (Neurology, Univ. of Pennsylvania, 328 Goddard Labs., Philadelphia, PA 19104, tifbiro@gmail.com), Navin Viswanathan, and Anne J. Olmstead (Commun. Sci. & Disord., Penn State Univ., University Park, PA)

When faced with communication challenges talker often modify their way of speaking producing *clear speech*. In a production study we investigated how American English talkers adapt medial flaps (e.g., *petal* and *pedal* are pronounced as [ˈpɛɾ̃əl]) following miscommunications with a computer partner. When subjects said a word (e.g., *petal*) the computer either interpreted the word correctly or incorrectly (it guesses *pedal*, *kettle* or ???). Talkers produced longer words, vowels, and stop closure in response to an incorrect versus correct guess. Talkers also changed their flapped productions to stops and lengthened stop closures more following voicing errors compared to other error types. Thus, talkers' adjustments appear to depend on the miscommunication type. Next, we investigated how listener perceptions were influenced by the talkers' adjustments. Listeners heard talkers' productions and indicated what they heard. Listener perception was more accurate for lengthened compared to shortened productions and better for stops than flaps. However, listeners misperceived /d/ for /t/ when talkers lengthened their /d/ closures. Moreover, production changes talkers made following a misrecognition improved listener perception for medial /t/, but not medial /d/, which was at-chance. These findings indicate that talker adjustments made to remedy a misrecognition do not universally aid listeners' perception.

4pSC28. The burden of face masks on speech in Parkinson's disease: Impact of clear and loud speech styles. Thea Knowles (Communicative Disord. and Sci., Univ. at Buffalo, 103D Cary Hall, South Campus, Buffalo, NY 14215, theaknow@buffalo.edu) and Gursharan Badh (Communicative Disord. and Sci., Univ. at Buffalo, Buffalo, NY)

Face masks, now ubiquitous as a means of reducing COVID-19 transmission, have been shown to act as a low pass acoustic filter on speech. People with Parkinson's disease (PwPD) and dysarthria may be especially vulnerable to the detrimental effects of masks, which may have negative consequences on spoken communication. The speech of PwPD has been shown to have lower concentrations of high frequency spectral energy and is perceptually characterized by a low speech volume and a breathy-hoarse voice quality. Increased vocal effort may aid in overcoming the filtering effects of masks. The purpose of this study was to quantify the effects of three masks (none, surgical, KN95) and three speech styles (habitual, clear, and loud) on the spectral characteristics of speech in PD. Fifteen PwPD and fifteen controls read aloud sentences with and without masks in three speech conditions. Measures of spectral tilt were taken from the long-term average spectra and modelled as a function of group, mask, and style. While both masks were associated with overall attenuation (steeper spectral tilt), a greater decrease was found for the PwPD (group by mask interaction). Loud speech, followed by clear speech, was effective in overcoming these effects for both groups.

4pSC29. The neurobiology of the adolescent brain: Interaction between hobby and speech processing. Yanxin Wang, Connie Huang, Connor Huang (Herricks High School, New Hyde Park, NY), Jeffrey Yang (The Bronx High School of Sci., Bronx, NY), and Yan H. Yu (Commun. Sci. & Disord., St. John's Univ., 4631 216 St., Bayside, NY 11361, yanhyu@gmail.com)

Experience shapes brain development. However, whether and how different types of hobbies influence speech processing in the developing brain is poorly understood. To increase our understanding of auditory neural responses in individuals with different hobbies, we compared teenagers with different hobbies (music, visual art, sports, and control) in their automatic neural sound encoding accuracy. Event-related potentials were recorded using 65-channel electroencephalogram sensor nets. Participants listened to English syllable contrasts in a passive listening paradigm. Mismatch negativity (MMN) and P3a components of the auditory event-related potentials were examined. We hypothesized that larger MMN will be observed in the musician brain than the rest of the groups. No MMN and P3a differences will be observed in teenagers with hobbies of visual arts and sports. Whether and how hobby modulates neural sound processing will be discussed.

4pSC30. Cortical plasticity for speech processing in English & Mandarin-speaking adolescents: The effect of transcranial direct current stimulation. Yanxin Wang (Herricks High School, New Hyde Park, NY) and Yan H. Yu (Commun. Sci. & Disord., St. John's Univ., 4631 216 St., Bayside, NY 11361, yanhyu@gmail.com)

Language experience and brain maturation both shape speech processing at the cortical level. To examine whether transcranial direction stimulation (tDCS) shows any effect on automaticity of neural speech processing in relation to language experience in adolescents, we recorded event-related potentials (ERP) to a speech contrast both before and after administering tDCS. The participants were adolescents (14–19 years old) from monolingual English-speaking households and bilingual English-Mandarin speaking households. We used a passive listening oddball paradigm and recorded ERP responses before and after 10 minutes of tDCS. The mismatch negativity (MMN) and late negativities (LN) were examined. Both groups showed clear neural discrimination. We observed language group differences in the amplitudes of MMN and stimulus condition differences in the amplitudes of LN. The preliminary analyses of ERP responses before and after tDCS suggest that one short session of tDCS does not influence neural discrimination of speech in typically-developing adolescents.

Session 4pSP

Signal Processing in Acoustics, Computational Acoustics, Underwater Acoustics, and Physical Acoustics: Model Based Signal Processing, Bayesian Learning, and Machine Learning II

Ananya Sen Gupta, Cochair
Department of Electrical and Computer Engineering, University of Iowa, 103 S Capitol Street, Iowa City, IA 52242

Scott H. Hawley, Cochair
Department of Chemistry and Physics, Belmont University, 1900 Belmont Blvd, Nashville, TN 37212

Ning Xiang, Cochair
School of Architecture, Rensselaer Polytechnic Institute, 110 Eighth Street, Troy, NY 12180

Chair's Introduction—1:00

Invited Papers

1:05

4pSP1. Feature selection for seabed characterization using non-linear regression and decision trees. Noah Roselli, Diego Rios, Akaash Patel, Yousef Sayes (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)

Knowledge of the seabed properties in an oceanic waveguide facilitates the understanding of wave propagation in the medium and, eventually, the implementation of tasks such as target detection and localization. Seabed characterization is, thus, a topic of extensive interest in the ocean acoustics research community. In this work, two approaches are implemented towards the task of seabed identification after the application of a preprocessing step that extracts characteristic features from received time-series following the transmission of a broadband sound signal. First, non-linear regression is applied, employing the extracted features and exploiting the relationship between those and geoacoustic properties. This approach leads to estimates of parameters relating to the sediment. Second, a decision tree is implemented, along with Principal Component Analysis, for identifying sediment types using the same features (or subsets of those). This technique is a classification task which maps propagation environments to specific types. Both approaches, tested on synthetic data from five sediment classes, lead to the successful extraction of information in regards to sediment properties. [Work supported by NSF and ONR.]

1:25

4pSP2. Features analysis to label sound sources by long-term monitoring. Domenico De Salvio (Dept. of Industrial Eng., Univ. of Bologna, Viale Del Risorgimento, 2, Bologna 40136, Italy, domenico.desalvio2@unibo.it), Michael J. Bianco, Peter Gerstoft (Marine Physical Lab., Univ. of California San Diego, Scripps Inst. of Oceanogr., La Jolla, CA), Dario D'Orazio, and Massimo Garai (Dept. of Industrial Eng., Univ. of Bologna, Bologna, Italy)

Background noise, speech privacy, and productivity are strictly related in offices. Employees' speech may affect the performance of colleagues disturbing their concentration. On the other hand, HVAC noise can mask irrelevant speech noise, improving in some way employee's comfort. Thus, it becomes important to achieve the ability to individuate in an unattended way the different sound sources during working hours: mechanical (HVAC, devices, computers, …), human, traffic, activities in nearby spaces. Clustering techniques provide tools to find patterns among unlabeled data. In a previous study, Gaussian Mixture Model and K-means clustering were applied to sound level meter measurements carried out during working hours. The reliability of results encourages the investigation to find robust features to label the sound sources. In the present work, a sound level measurement has been carried out alongside the audio recording of the working activities throughout a whole day. Besides the clustering performed via Gaussian Mixture Model and K-means clustering over the sound pressure levels strings, a Variational Autoencoder was used to find latent features from the recording. Correlations between the methods explore the chance to obtain a broader understanding of data obtained by long-term monitoring.

1:45

4pSP3. Acquisition functions in Bayesian optimization of ocean acoustic waveguides using Gaussian processes. William F. Jenkins (Scripps Inst. of Oceanogr., UC San Diego, 9500 Gilman Dr., La Jolla, CA 92093, wjenkins@ucsd.edu) and Peter Gerstoft (Univ. of California, San Diego, La Jolla, CA)

Geoacoustic model optimization and inversion are computationally expensive endeavors. In cases where a parameter grid search is prohibitively expensive, optimization produces an approximated solution through sampling techniques such as Markov chain Monte Carlo, simulated annealing, and genetic algorithms. More recent work proposes replacing such sampling with a Bayesian approach that uses a Gaussian process as a surrogate model of the objective function. The surrogate model represents the posterior on the objective function and is updated with each sample evaluation. As an alternative to sampling methods listed above, acquisition functions incorporate the uncertainty in the posterior to select the next point to be sampled and can greatly speed the optimization. In this study, four common acquisition functions are described encompassing approaches that use manual tuning (upper confidence bound), improvement criteria (probability of improvement and expected improvement), and information criteria (entropy search). Results are presented from an acoustic parameter search for a shallow water waveguide for each function. Expected improvement is found to be the preferred acquisition function for the simulations, converging on the optimal solution more quickly than the other acquisition functions.

2:05

4pSP4. Comparison of maximum entropy and ResNet-18 inferences of sediment sound speed using surface ship noise. Tracianne B. Neilsen (Phys. and Astronomy, Brigham Young Univ., N251 ESC, Provo, UT 84602, tbn@byu.edu), Michael C. Mortenson, Jacob R. Nuttall, Stephen M. Amos, Mark K. Transtrum (Phys. and Astronomy, Brigham Young Univ., Provo, UT), David P. Knobles (Phys., Knobles Sci. and Anal., Austin, TX), and William Hodgkiss (Marine Physical Lab., Scripps Inst. of Oceanogr., San Diego, CA)

The impact of individual seabed properties on sound propagation in the ocean depends on many factors including source-receiver range and frequency band of interest. In this talk, estimates of the sound speed ratio across the water-sediment interface are obtained using a maximum entropy approach and ResNet18, a supervised machine learning model. The input data are spectrograms of surface ship noise from shipping lanes. Synthetic spectrograms are modeled using a ship noise source spectrum and a range-independent normal mode model, ORCA, with a wide range of environments and ship parameters. Experimental data from the New England Mud Patch are used with both inverse methods. The maximum entropy approach uses data-model mismatch to obtain a posterior probability distribution for the parameters of interest. The ResNet18 is trained on the synthetic spectrograms, augmented with additive noise, and then applied to the experimental data. A comparison of the results from these two methods for a variety of ships using different frequency bands will be presented, along with a discussion of the advantages and limitations of each method.

2:25

4pSP5. Estimation and interpretation of multipath channel activity using braiding techniques. Ryan A. McCarthy (Marine Physical Lab., Scripps Inst. of Oceanogr., 103 S. Capitol St., Iowa City, IA 52242, ryan-mccarthy-1@uiowa.edu), Ananya Sen Gupta, and Madison Kemerling (Elec. and Comput. Eng., Univ. of Iowa, Iowa City, IA)

Robust underwater acoustic channel estimation is critical towards improving communications efforts and enhancing awareness of changing environments. To explore these channels in-depth, machine learning algorithms are developed through feature geometric representations, referred to as "braiding," to interpret multipath ray bundles within shallow water acoustic channels in two ways. The first application of this work predicts the number of reflections an acoustic signal may undergo through the environment by applying known physical parameters and braid features. The second application explores the importance of a braid feature within the acoustic channel for estimation purposes by using braid path information. Three unique machine learning techniques are trained to predict the applications using a diverse set of shallow water acoustic channels generated through the BELLHOP model. Machine learning models developed for the applications demonstrate high testing accuracies with an accuracy of 86.70% in the first application and an accuracy of 99.94% in the second application. As a further demonstration, braid feature representations and model predictions are used for channel estimation and determining the number of reflections using SPACE08 field data.

2:45–3:00 Break

Contributed Papers

3:00

4pSP6. Depth estimation as a sequential process following array invariant range estimation. Paul Hursky (Appl. Ocean Sci. LLC, 4825 Fairport Way, San Diego, CA 92130, paul.hursky@gmail.com), Christopher Verlinden, and Tessa Munoz (Appl. Ocean Sci. LLC, Fairfax Station, VA)

Range estimation based on the array invariant concept exploits the inherent properties of shallow-water waveguide propagation. Multipath arrivals observed on a vertical line array, indexed on vertical arrival angle and time difference of arrival, form what is known as a beam migration pattern in the shape of an ellipse that is a function of source range and the acoustic invariant beta. Fitting the beam migration pattern to an ellipse yields the source range. A property of this beam migration pattern is that it is independent of source depth. Source depth determines how the multipath arrivals are distributed along that ellipse. We will extend previous work on estimating range and present how to estimate depth. This decouples the two estimates and reduces a 2D search process to two 1D processes. We will explore the CRLB structure of the range and depth estimates and demonstrate the results of processing experiment data from the RADAR 2007 experiment in shallow water off the coast of Portugal.

3:15

4pSP7. Statistical-based feature extraction and classification of active sonar data. Bernice Kubicek (Dept. of Elec. and Comput. Eng., Univ. of Iowa, 103 South Capitol St., Iowa City, IA 52242, bernice-kubicek@uiowa.edu), Ananya Sen Gupta (Dept. of Elec. and Comput. Eng., Univ. of Iowa, Iowa City, IA), and Ivars Kirsteins (NUWC, Newport, RI)

Sonar target recognition is difficult due to the potential nonlinear overlap within an acoustic color response due to various backscatter and clutter within the ocean. This talk presents initial results from using a statistical model of feature vectors in conjunction with machine learning classifiers.

Canonical correlation analysis (CCA) seeks to find two linear combinations of data by maximizing the correlation between the linear combinations while maintaining unit variance. In this application, CCA is used as a feature extraction method before target classification of active sonar data experimentally collected during the Shallow Water Active Classification (SWAC)-1 and SWAC-2 sea trials in the Malta Channel. The database consists of 20 targets; three were analyzed using this method. The data are generated by taking windows of consecutive pings from the ping-vs-time domain and performing CCA. The intuition behind using CCA is that there are persistent features within the data that morph over time due to changing target aspect angles and platform positions which can be represented by the maximally correlated linear combinations of data among consecutive pings. The resulting linear combinations are feature vectors used to train a single hidden-layer neural network classifier. Results are reported as overall classification accuracy and confusion matrices.

3:30

4pSP8. Signal classification with machine learning. Manton J. Guers (Acoust., Penn State Univ., PO Box 30, State College, PA 16804, mjg244@psu.edu) and Tyler P. Dare (Acoust., Penn State Univ, State College, PA)

This paper investigates and evaluates several Machine Learning techniques for the proper identification and classification of analytical signals. Signals having different "shapes" and periods were defined analytically to have pre-determined class associations. Supervised Machine Learning techniques were then investigated to evaluate the Machine Learning methodology's ability to properly classify the analytical signals based on characteristics of interest.

OPEN MEETINGS OF TECHNICAL COMMITTEES

Technical Committees of the Acoustical Society of America will hold open meeting on Tuesday, Wednesday, and Thursday evenings.

All meetings will begin at 7:30 p.m., except for Engineering Acoustics which will meet starting at 4:45 p.m. on Tuesday and Computational Acoustics which will meet starting at 4:30 p.m. on 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 discussion.

Committees meeting on Tuesday

Engineering Acoustics (4:45 p.m.)	Governors Square 10
Acoustical Oceanography	Governors Square 14
Animal Bioacoustics	Governors Square 17
Architectural Acoustics	Plaza Ballroom A
Physical Acoustics	Governors Square 11
Psychological and Physiological Acoustics	Plaza Ballroom D
Signal Processing in Acoustics	Governors Square 16
Structural Acoustics and Vibration	Governors Square 12

Committees meeting on Wednesday

Biomedical Acoustics	Governors Square 15

Committees meeting on Thursday

Computational Acoustics (4:30 p.m.)	Governors Square 12
Musical Acoustics	Directors Row H
Noise Plaza	Plaza Ballroom D
Speech Communication	Plaza Ballroom E
Underwater Acoustics	Governors Square 14

Session 5aAA

Architectural Acoustics, ASA Committee on Standards and Noise: Courts and Municipal Buildings

Jessica S. Clements, Cochair
Special Technologies, Newcomb & Boyd, 303 Peachtree Center Ave. NE, Suite 525, Atlanta, GA 30303

Dennis Paoletti, Cochair
Paoletti Consulting, 708 Foothill Drive, San Mateo, CA 94402

Chair's Introduction—8:30

Invited Papers

8:35

5aAA1. Architecture and acoustics: Effective collaboration for judicial facility design. Kristine B. Johnson (HOK, 3223 Grace St. NW, Washington, DC 20007, kristine.johnson@hok.com)

Designing public buildings, like judicial facilities, requires careful planning and thoughtful integration of all building systems. Judicial facilities can be stressful environments and the planning, design, and performance of these buildings can enhance or detract from a user's experience of the space. This presentation will address best practices for courthouse planning and explore opportunities for architects and acousticians to collaborate effectively in designing spaces that meet the needs of their stakeholders.

8:55

5aAA2. Conversations with the Administrative Office of the U.S. Courts and changes to the US Courts Design Guideline. Jessica S. Clements (Acoust., Newcomb & Boyd, LLP, 303 Peachtree Ctr. Ave. NE, Ste. 525, Atlanta, GA 30303, jclements@newcomb-boyd.com)

The Administrative Office (AO) of the U.S. Courts oversees the design and construction of courts spaces within new federal court buildings following the U.S. Courts Design Guide. Conflicts between the U.S. Courts Design Guide and the General Services Administration (GSA) PBS P-100 are known to raise questions for designers. Conversations with Facility Program Managers and interviews with other designers and users from the AO and GSA have provided invaluable insight into their needs and challenges. These insights have been used to inform changes to the acoustical design chapter of the U.S. Courts Design Guide. This presentation will outline the insights gathered through the interviews and present the changes that are included in the updated acoustics design chapter of the U.S. Courts Design Guide.

9:15

5aAA3. Addressing federal court facility acoustical design challenges. Brandon M. Westergaard (Special Technologies, Newcomb & Boyd, 303 Peachtree Ctr. Ave. NE, #525, Atlanta, GA 30303, bwestergaard@newcomb-boyd.com)

Modern federal courthouse acoustical designs mandate a multidisciplinary approach in which architects, engineers, consultants, contractors, and court administrators work together to provide cost-effective designs which meet explicit speech privacy, room acoustic, and building systems background noise requirements. It is a primary responsibility of the acoustical consultant to understand and support implementation of owner standards, the U.S. Courts Design Guide criteria, and various tenant adjacency requirements, all of which present unique challenges. Owner standards vary from project to project in accordance with functional requirements, building location, economic constraints, and other project-specific needs. A new edition of the U.S. Courts Design Guide containing significant updates is expected to be published in the near future. Tenant adjacency requirements are often ambiguous and conflict with each other and with other project criteria. This paper examines a case study of three federal courthouse designs, focusing on challenges associated with intertwined project requirements, cost constraints, and the design-build process. The study translates federal judiciary and courthouse tenant requirements and provides insight into established design precedents. Solutions to design challenges are presented, which ensure future designs meet the acoustical needs of federal court facilities.

9:35

5aAA4. Lessons learned that impact acoustics from the California Courts Program. Dennis Paoletti (Paoletti Consulting, 708 Foothill Dr., San Mateo, CA 94402, dpaoletti88@gmail.com) and Clifford Ham (Administrative Office of the Courts, Former Project Director, Judicial Council of California (Retired), Oakland, CA)

Architect Clifford Ham served as the Project Director and Principal Architect for the Judicial Council of California Administrative Office of the Courts. He was responsible for overseeing more than 450 court facilities throughout California. Projects ranged from renovations and small single court buildings in remote locations to new multistory (31 and 71 multiple purpose courtrooms) in dense major urban areas. As an architect, with courtroom design experience, he was able to efficaciously support all aspects of design, engineering, and specialty consulting. Throughout the process, projects were implemented to effectively meet the needs of various stakeholders. California Trial Court Facilities Standards, covering all aspects of architectural design and engineering, had been developed when the program began in 2006, with minor revisions leading up to the new 2020 edition. Revisions gained from first-hand experience emphasized changes in the guiding principles and updated technical areas which influenced the design and cost of courthouses. This presentation will review the discuss significant aspects of the overall courts program with specific emphasis on acoustics. Case study examples will present some interesting subtleties in the planning and design process that should be realized to avoid costly follow-up repair solutions after occupancy.

9:55–10:25
Panel Discussion

FRIDAY MORNING, 27 MAY 2022 GOVERNORS SQUARE 11, 8:00 A.M. TO 10:45 A.M.

Session 5aPA

Physical Acoustics: General Topics in Physical Acoustics II

Konstantin I. Matveev, Chair
Washington State University, Sloan Hall, Pullman, WA 99164

Contributed Papers

8:00

5aPA1. Combined thermoacoustic cooling and ortho-para conversion of cryogenic hydrogen. Konstantin I. Matveev (Washington State Univ., Sloan Hall, Pullman, WA 99164, matveev@wsu.edu) and Jacob I. Leachman (Washington State Univ., Pullman, WA)

Hydrogen is a promising fuel for our future, as it can be produced with renewable energy sources and it does not emit harmful emissions when reacting with oxygen. To make the hydrogen economy viable, hydrogen needs to be transported as a dense cryogenic liquid. Since efficiencies of current cryocoolers are relatively low, improvements of the cooling technologies are required. In this study, an innovative solution involving thermoacoustic flow-through cooler is considered. Acoustic waves passing through porous medium can pump heat and cool hydrogen flowing through the system. An important feature of cryogenic hydrogen is the shift between its equilibrium ortho and para isomer fractions, which requires additional removal of heat. Since the natural conversion is slow, it must be augmented by catalysts for fast transition. The advantage of using the flow-through thermoacoustic system to cool hydrogen is that a catalyst can be placed in the porous matrix where hydrogen flows, so it can be cooled and undergo conversion at the same time. A mathematical model describing this process has been developed. Results showing achievable coefficients of performance, temperature drops, and flow rates of hydrogen are presented for both standing-wave and travelling-wave systems.

8:15

5aPA2. Study of electronic band structure in a 75-nm-wide n-GaAs/AlGaAs quantum well by surface acoustic waves. Alexey Suslov (NHMFL, 1800 East Paul Dirac Dr., Tallahassee, FL 32310, suslov@magnet.fsu.edu), Irina Drichko, Ivan Smirnov, Mikhail Nestoklon (Ioffe PTI, St. Petersburg, Russian Federation), Dobromir Kamburov, Kirk Baldwin, Loren Pfeiffer, Ken West (Princeton Univ., Princeton, NJ), and Leonid Golub (Ioffe PTI, St. Petersburg, Russian Federation)

From measured magnetic field dependences of attenuation and velocity of surface acoustic waves (SAW), we calculated the real part of electron the conductivity in a 75-nm-wide n-GaAs/AlGaAs quantum well in magnetic fields of up to 18 T at frequencies of 30–300 MHz in the temperature range 20–500 mK. We observed slow magneto-oscillations of the conductivity depended on SAW intensity and occurred due to elastic intersubband scattering between the symmetric (S) and antisymmetric (AS) subbands formed by Coulomb repulsion between the electrons. Our theoretical description of the magneto-oscillations allowed estimating the quantum and the intersubband scattering times. The experimentally obtained subband splitting value 0.42 meV was close to the theoretically found 0.57 meV. Additionally, in tilted magnetic fields of 0.5–1.5, T we observed crossings of the Landau levels between the S and AS subbands. According to the developed theory, the crossings at the in-plane field below 1.5 T occur due to different dependences of cyclotron energies in the subbands on the in-plane field. Namely, the cyclotron energy in the S (AS) subband decreases (increases) with the rising of the in-plane field. Work supported by "BASIS," NSF DMR-1644779 and DMR-1420541, GBMF4420, and State of Florida.]

8:30

5aPA3. Photoacoustic optical absorption measurement of aerosols. John A. Case (PSU/ARL, P.O. Box 30, State College, PA 16804, jac7175@psu.edu) and Robert W. Smith (PSU/ARL, State College, PA)

Atmospheric optical absorption, including the contribution from aerosols, is important in modeling propagation of laser beams and for climate research. Photoacoustic methods have been used in the past for this purpose, as they directly measure absorption, rather than the sum of absorption and scattering. Acoustic operating frequency determines the sensed aerosol size range. The design of a windowless double-Helmholtz photoacoustic aerosol sensing system, operating at an acoustic frequency near 150 Hz, with the capability to measure an optical absorption with a 1/*e* length of 1 Mm, will be presented. In this design, environmental noise limits the instrument sensitivity. To improve immunity to environmental noise with an open resonator, an aluminum acoustic enclosure was built around the system, with welded vacuum flanges for inputs and outputs. Inlet and outlet acoustic mufflers were designed to limit the in-band noise entering the measurement system. A modulated 10 W 1064 nm fiber laser was used as the excitation source, and a high-sensitivity microphone placed within the open-cell cavity was used to measure the internal photoacoustic pressure fluctuations. The system was modeled using various computational tools, including MATLAB, Simulink, and ANSYS, and then was constructed and tested. Lab measurement results will be presented.

8:45

5aPA4. Design and characterization of a broadband multi-node bulk acoustic wave device for acoustofluidic applications. Nathan Jeger-Madiot (PMMH, ESPCI-CNRS, Barre Cassan - Bât A 1er étage Case 18-7 Quai Saint Bernard, Paris 75005, France, nathan.jeger-madiot@espci.fr), Mauricio Hoyos, and Jean-Luc Aider (PMMH, ESPCI-CNRS, Paris, France)

The classic approaches to generate the Acoustic Radiation Force (ARF) in a standing wave cavity are based on the use of a specific height for the cavity and a narrow-band acoustic frequency to reach the resonance. We present a robust multi-node cavity working over a broadband range of frequency. This approach allows large acoustophoresis manipulations. The design of a transparent cavity and the use of a broadband ultrasonic transducer allowed the characterization of the acoustic energy and the comparison with a simple 1-D model. The acoustic properties of the system were estimated through the effects of the acoustic radiation forces on particles suspension. From the particle velocities induced by the ARF, measured with a Particle Image Velocimetry (PIV) approach, we deduced the acoustic energy over a large frequency range. The automation of the setup allowed the acquisition of a large amount of data to make parametric studies. The results show a wide continuous operating range for the acoustic radiation forces. Furthermore, the occurrence of the resonance peaks allows an increase by a factor ten of the force magnitude. This broadband effect also enables an accurate control of the node positions by tuning the proper acoustic frequency.

9:00

5aPA5. Modeling nonlinear acoustic damping due to flow separation. Joseph Day (Mech. and Aerosp. Eng., Univ. of Colorado Colorado Springs, 1420 Austin Bluffs Pkwy, Colorado Springs, CO 80918, jday@uccs.edu) and J. M. Quinlan (Mech. and Aerosp. Eng., Univ. of Colorado Colorado Springs, Colorado Springs, CO)

Nonlinear acoustic damping has been observed in many high-amplitude acoustic systems as a result of flow separation and shear layer vortical motion, eventually transforming some of the acoustical energy into heat. The amount of nonlinear acoustic damping helps to determine the nonlinear limit cycle amplitude, e.g., damping caused by baffle blades in a liquid rocket engine to reduce combustion instabilities. The damping mechanism is dependent on both the location and phase of flow separation. Identifying the flow separation is a function of both the boundary layer growth and the acoustically imposed pressure gradient. When the acoustic pressure gradient is adverse, the boundary layer is more prone to separation. Using this as a basis, a model can be created that is applicable to general geometry, which will then be used to approximate the nonlinear acoustic damping in various situations. The constructed model will be compared to established cases, such as an orifice in a duct, to validate the model.

9:15

5aPA6. Acoustic streaming in the cochlea. Charles Thompson (Elec. and Comput. Eng, UMASS Lowell, 1 Univ. Ave. Lowell, MA 01854, charles_thompson@uml.edu), Kavitha Chandra, Emi Aoki, and Aidan Keefe (Elec. and Comput. Eng, UMASS Lowell, Lowell, MA)

This presentation examines the fluid motion resulting from airborne and bone conduction excitation in the cochlea. The time-averaged velocity, termed acoustic streaming, driven by the gradient of the Reynolds stress, is of particular interest. The effect of asymmetry in the oval and the round window velocity and its influence pressure gradient across the cochlear partition and acoustic streaming is explored. It is shown that the streaming flow is a function of the reciprocal of the Strouhal, 1/S and the oscillatory Reynolds number, 1/R, have a magnitude much less than one for frequencies of interest. The work presented will focus on the parameter range $R/S^2 = O(1)$. The streaming velocity is obtained in terms of asymptotic expansion in terms of the reciprocal of the Strouhal number. Solution of the streaming flow outside the viscous region is obtained using boundary conditions derived from a matched asymptotic analysis.

9:30

5aPA7. Ultrasonic scattering in two-phase polycrystalline materials. Showmic Islam (Mech. & Mater. Eng., Univ. of Nebraska Lincoln, Lincoln, NE 68588, sislam2@unl.edu) and Joseph A. Turner (Mech. and Mater. Eng., Univ. of Nebraska-Lincoln, Lincoln, NE)

Typically, many of the polycrystalline materials found in nature and those created using different manufacturing processes consist of multiple phases. Careful nondestructive characterization of these materials is of fundamental importance for quality assurance because microstructure is strongly correlated with mechanical performance. Ultrasonic scattering measurements have shown high sensitivity to material microstructure such that this approach is of interest for multi-phase materials. A fundamental understanding of scattering from idealized two-phase materials would provide essential information regarding the predictive nature of scattering measurements to quantify such microstructures. For this purpose, samples were created using spark plasma sintering (SPS) from mixtures with three different ratios (with respect to mass) of Cu and Fe powders. These elements do not alloy such that the resulting samples will not have any intermediate phases. Moreover, one pure Cu sample and one pure Fe sample were also sintered using SPS. The samples were imaged using electron backscatter diffraction (EBSD) to quantify the material organization. Ultrasonic backscatter and attenuation measurements were performed on the samples and the results were used for comparison with scattering models. Prospects for nondestructive evaluation of such materials are also discussed. [Work supported by Air Force Research Laboratory (AFRL) under prime contract FA8650-15-D-5231.]

9:45–10:00 Break

10:00

5aPA8. Ultrasonic scattering predictions of two-phase polycrystalline materials based on digital microstructures. Showmic Islam (Mech. & Mater. Eng., Univ. of Nebraska Lincoln, Lincoln, NE 68588, sislam2@unl.edu) and Joseph A. Turner (Mech. and Mater. Eng., Univ. of Nebraska-Lincoln, Lincoln, NE)

Many materials found in nature and used in industry consist of multiple phases, and the mechanical performance of such materials is controlled by their material organization. Ultrasonic scattering is one approach to characterize these microstructures nondestructively. In this presentation, previous ultrasonic scattering theories for two-phase materials are discussed and implemented using discrete synthetic three-dimensional microstructures as a means to assess the material statistics. These models predict the ultrasonic wave speed, attenuation, and diffuse ultrasonic backscatter with respect to the volume fraction and grain size distribution of each constituent. The computational approach is then focused on predictions for samples created using spark plasma sintering (SPS) of Cu and Fe powders with varying volume fractions as well as pure Cu and pure Fe samples. DREAM.3D is used to generate digital synthetics representative of the SPS samples as characterized using electron backscatter diffraction (EBSD). A set of 30 realizations for each volume fraction is used for each to quantify the statistical variations expected. The combined theoretical and computational model is used to study the dependence of ultrasonic wave propagation on the material organization for these two-phase materials. [Work supported by Air Force Research Laboratory (AFRL) under prime contract FA8650-15-D-5231.]

10:15

5aPA9. Anisotropy of elastic and photoelastic properties of lithium niobate crystals. Farkhad Akhmedzhanov (Lab. of Thermophysics of Multiphase Systems, Inst. of Ion-plasma and Laser Technologies of the Acad. of Sci. of Uzbekistan, 33 Durmon Yuli St., Tashkent, Tashkent 100125, Uzbekistan, akhmedzhanov.f@gmail.com) and Ulugbek Abdirakhmonov (Lab. of Thermophysics of Multiphase Systems, Inst. of Ion-plasma and Laser Technologies of the Acad. of Sci. of Uzbekistan, Tashkent, Uzbekistan)

The elastic and photoelastic constants of lithium niobate crystals and the attenuation coefficients of acoustic waves at room temperature have been determined by the Bragg diffraction of light by sound. The components of the photoelasticity tensor were determined by the modified Dixon method. The obtained values of the constants were used to determine the anisotropy of the effective elastic and photoelastic constants with a change in the direction of propagation of acoustic waves in the (100) symmetry plane, for different directions of the wave vector and polarization of light relative to the sound wave vector. It is shown that, for certain geometries of the Bragg light diffraction, there is a correlation between the orientation dependences of the effective elastic and photoelastic constants in the investigated plane. The results of the study can be used to analyze the anisotropy of the photoelastic properties of lithium niobate crystals and determine the diffraction geometry with the highest efficiency of Bragg light diffraction.

10:30

5aPA10. Attenuation of acoustic waves and Grüneisen tensor in lithium fluoride crystals. Farkhad Akhmedzhanov (Lab. of Thermophysics of Multiphase Systems, Inst. of Ion-plasma and Laser Technologies of the Acad. of Sci. of Uzbekistan, 33 Durmon Yuli St., Tashkent, Tashkent 100125, Uzbekistan, akhmedzhanov.f@gmail.com) and Jakhongir Kurbanov (Lab. of Thermophysics of Multiphase Systems, Inst. of Ion-plasma and Laser Technologies of the Acad. of Sci. of Uzbekistan, Tashkent, Uzbekistan)

The anisotropy of acoustic attenuation in lithium fluoride crystals in the range of 0.4–1.6 GHz was studied by the method of Bragg light diffraction on acoustic waves. The obtained values of the velocity and attenuation coefficient of acoustic waves were used to determine all independent real and imaginary elastic constants. Based on the Akhiezer attenuation mechanism, the anisotropy of acoustic attenuation was determined for the propagation of longitudinal and transverse waves in the (001) and (110) planes. All directions along which pure longitudinal and transverse acoustic waves propagate are found. To describe the anisotropy of acoustic attenuation, the Grüneisen acoustic tensor is used for the first time, the components of which have been determined from the values of the attenuation coefficient of acoustic waves along the crystallographic axes [100], [110], and [111]. It is shown that the orientational dependence of the effective Grüneisen constant, defined in terms of the components of this tensor, describes well the anisotropy of the attenuation of acoustic waves in the studied frequency range. The proposed approach can be used to all cubic crystals in which the main attenuation mechanism is the phonon–phonon Akhiezer mechanism.

Session 5aSC

Speech Communication: Speech Perception II (Poster Session)

Ye-Jee Jung, Chair
Purdue University, 215 Nimitz Dr., Apt 4, West Lafayette, IN 47906

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

Contributed Papers

5aSC1. The relationship between perceived speech clarity and dysphonia: Preliminary investigation. 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), Julia K. Ellerston (Otolaryngol. - Head and Neck Surgery, Univ. of Utah, Salt Lake City, UT), and Sarah H. Ferguson (Commun. Sci. and Disord., Univ. of Utah, Salt Lake City, UT)

Perceived listener ratings of sentence "clarity" have been used in studies involving clear speech and have been shown to demonstrate a strong correlation with vowel intelligibility as well as a robust clear speech effect (Ferguson and Kerr, 2009; Ferguson and Morgan, 2018). However, it remains unclear what aspects of the speech signal listeners are using to form their perception of speech clarity. It is possible that disruptions in the voice source (i.e., poor voice quality) affect listeners' perception of clarity. Given the increased prevalence of dysphonia in elderly adults (a target population for clear speech), it is important to determine whether voice quality negatively affects perceptions of speech clarity. The current study aims to examine the relationship between perceived sentence clarity and acoustic and perceptual measures of voice quality. Eighty sentences produced by 10 female talkers (5 healthy young adults and 5 older adults with hearing loss) already rated for perceived clarity by young adult listeners will be analyzed in terms of acoustic measures of voice quality (CPPS, HNR, percent glottal fry, and spectral slope) and perceptual measures of voice quality (CAPE-V evaluation performed by two speech-language pathologists with training in voice disorders).

5aSC2. Phonetic entrainment of Cantonese-speaking children with autism spectrum disorder (ASD). Yitian Hong (Dept. of Chinese and Bilingual Studies, The Hong Kong Polytechnic Univ., Hung Hom, Hong Kong, ytian.hong@connect.polyu.hk), Fang Zhou, Si Chen, Angel Wing Shan Chan, Tempo Po Yi Tang, Eunjin Chun, Bei Li, Phoebe Choi, Chakling Ng, Fiona Cheng, and Xinrui Gou (Dept. of Chinese and Bilingual Studies, The Hong Kong Polytechnic Univ., Hung Hom, Hong Kong)

Individuals with Autism Spectrum Disorder (ASD) typically show less engagement in social interactions. Previous studies in verbal communication found that they are less able to entrain the phonetic features to their interlocutors, compared to their Typically Developing (TD) counterparts. In this study, we examined the phonetic adjustment of 15 Cantonese-speaking ASD children (mean age = 8.5 years, range = 6–10.8) and 9 Cantonese-speaking TD children (mean age = 7.9 years, range = 6.4–9.6) when using the designed sentences to answer questions raised by the same experimenter. There are three main findings: (1) ASD children tended to disentrain the minimum f0 from the experimenter while TD children showed consistent minimum f0 through the experiment, possibly because TD children noticed the convergence made by the experimenter; (2) TD children significantly entrained the intensity towards the experimenter, but no entrainment was found in ASD children; and (3) both groups demonstrated an increase of speech rate, catching up with the speech rate of the experimenter. Although children at this age range might not fully acquire entrainment skills, our results suggested that compared to TD children, ASD children started to show atypicality of phonetic adjustment in conversations. This study of Cantonese speakers makes cross-linguistic contribution to the literature of ASD children's language acquisition.

5aSC3. Non-native speech perception and the role of phonological memory: A re-examination. Amy Hutchinson (Purdue Univ., 2403 Neil Armstrong Dr., Apt 2C, West Lafayette, IN 47906, hutchi25@purdue.edu) and Olga Dmitrieva (Purdue Univ., West Lafayette, IN)

Previous laboratory-based research has found phonological memory to significantly affect the perception of non-native speech sounds (Inceoglu, 2019). The present study aims to bolster findings reported in Inceoglu (2019) and provide support for the efficacy of online research by replicating the study's laboratory-based research in an online setting. Data were collected from 32 native English-speaking learners of French (12 male, 20 female; mean age: 33 years old) online using the Prolific data collection platform (www.prolific.com). The study consisted of two major components: an L2 French nasal vowel identification task (Inceoglu, 2019) and a nonword repetition task designed to assess phonological memory (Anderson, 2012; Grey *et al*., 2015; Inceoglu, 2019; Kissling, 2014). Results of a simple linear regression revealed that participants with higher scores on the nonword repetition task (indicative of better phonological short-term memory) were significantly likely to perform more target-like on the nasal vowel identification task than those with lower scores ($t = 12.57$; $p < 0.001$). These findings not only replicate the original results outlined in Inceoglu (2019), but they also support the efficacy of online research. The latter finding is crucial as research begins to adjust to accommodate a more socially distanced world.

5aSC4. Multimodal training using pitch gesture improves Mandarin tone recognition for children with cochlear implant. Hongwei Ding (Shanghai Jiao Tong Univ., 800 Dong Chuan Rd., Shanghai 200240, China, hwding@sjtu.edu.cn), Jing Zhang (Sonova China, Shanghai, China), Hao Zhang (Ctr. for Clinical Neurolinguistics, School of Foreign Lang. and Lit., Shandong Univ., Jinan, China), and Yang Zhang (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN)

Prelinguallydeafened Mandarin-speaking children with cochlear implants (CI) encounter a significant challenge in perceiving lexical tones accurately due to their limited language experience and insufficient pitch information provided by CI devices. To facilitate their tonal perception, we examined the role of pitch gestures in the training program, which was reported to be beneficial in acquiring Mandarin lexical tones for foreign learners of Chinese. In the current study, 18 prelingually deafened preschoolers with CI were recruited in Shanghai. They were randomly assigned to two groups. The experimental group was trained with audio and pitch gestures, and the control group was trained with audio only. Three tone-identification tests were conducted before, in the middle, and after the eight training sessions. Although the two groups had identical performance before

the training, the experimental group demonstrated significantly better performance than the control group after training sessions, especially in noise conditions. The results thus showed that multimodal training with pitch gestures improved the tone recognition ability of CI children more than sheer auditory training. Our findings offer more evidence to support that learning to perceive lexical tones can be facilitated by multimodal cues and also provide important implications for optimizing rehabilitation training after implantation.

5aSC5. Older adults' perception of medically related sentences in three listening conditions. Tessa Bent (Speech, Lang. and Hearing Sci., Indiana Univ., 2631 East Discovery Parkway, Bloomington, IN 47408, tbent@iu.edu), Melissa M. Baese-Berk (Univ. of Oregon, Eugene, OR), Erica E. Ryherd (Architectural Eng., Univ. of Nebraska - Lincoln, Omaha, NE), and Sydney Perry (Speech, Lang. and Hearing Sci., Indiana Univ., Bloomington, IN)

During hospital stays, crucial information about diagnoses, treatment plans, and discharge instructions is provided. Although understanding this information is essential, several barriers to successful communication exist that may have differing impacts depending on a patient's age. First, hospitals are noisy places, which could challenge speech understanding for older adults due to the increased incidence of hearing loss. However, older adults' larger vocabularies may support their comprehension of infrequent or less familiar medical words. To investigate how these factors impact intelligibility for older adults, recordings of medically related sentences with different familiarity/frequency types (low/low, high/low, high/high) were presented to 79 listeners (average age = 66 years; range 60–81) for transcription in quiet, speech-shaped noise, or hospital noise. Performance was best in quiet and worst in hospital noise. Both word frequency and familiarity impacted performance. Sentences with high familiarity but low frequency had lower accuracy than those with high familiarity and frequency; sentences with low frequency and familiarity showed low accuracy, particularly in hospital noise. Thus, hospital noise may impact the understanding of essential medical information. [Work by the IU Institute for Advanced Study and the James S. McDonnell Foundation, https://doi.org/10.37717/2021-3028.]

5aSC6. Employer ratings of cochlear implant users from speech samples. Valerie Freeman (Oklahoma State Univ., 042 Murray Hall, Dept. of Commun. Sci. & Disord., Stillwater, OK 74078, valerie.freeman@okstate.edu)

This study continues a project on speech-based perceptual "first impressions" of early-implanted cochlear implant (CI) users. Hiring managers with typical hearing (TH, N = 72) from various industries heard speech samples from CI users and TH young adults and rated their personalities and attractiveness as applicants for various types of jobs. As with college-student peer listeners in prior studies, employers rated TH speakers most positively, CI users with high speech intelligibility (CI-Hi) as intermediate, and CI users with lower intelligibility (CI-Lo) most negatively. Within this pattern, CI-Hi were rated somewhat lower by managers hiring for white-collar jobs (versus blue-collar) and by those who placed high importance on communication skills, who also rated CI-Lo lower on communication skills and "hirability" but higher on extroversion. These patterns underline the practical importance of CI users' speech intelligibility on their employment opportunities, particularly when entering white-collar and highly communicative jobs, where biases may exist against deaf speech and other unfamiliar patterns. However, managers with more knowledge of and experience with deafness rated CI-Hi communication skills more positively, supporting the value of increasing opportunities for interaction at school and work, as well as education about deafness, its effects on speech, and communication-repair strategies within the workplace.

5aSC7. Implicit learning versus explicit instruction in morphophonological learning. Shiloh Drake (Dept. of Linguist, Univ. of Oregon, 1290 University of Oregon, Eugene, OR 97403, sdrake@uoregon.edu), Isabel Preligera, and Melissa M. Baese-Berk (Dept. of Linguist, Univ. of Oregon, Eugene, OR)

This study uses an artificial grammar learning task to probe the learning of abstract morphophonological structure. Two sets of nonce words were created, one with plural forms using concatenative morphology (similar to English) and the other using non-concatenative morphology (similar to Arabic). Both sets had multiple phonologically conditioned allomorphs. Half of the participants were provided with instruction and feedback, and half were not. These results show that even minimal instruction and feedback leads to learning abstract morphological structure through the use of nonce words. While instruction and feedback at this level may not make non-concatenative morphology more straightforward to English speakers, their subconscious understanding of the morphophonological system is improved and leads to higher quality guesses on the derivations of the nonce words than on previous work with implicit learning of non-concatenative morphology alone (e.g., Drake, 2018). Feedback provides negative input for a second-language learner to narrow possibilities when following an otherwise abstract morphophonological pattern, particularly in a morphophonological system unlike that employed by their primary language.

5aSC8. Evaluating contributions of relative vocabulary size and demand for use on masked-speech recognition for bilingual children with normal hearing. Tiana Cowan (Ctr. for Hearing Res., Boys Town National Res. Hospital, 401 N 46 St., 4107, Omaha, NE 68132, Tiana.cowan@boystown.org), Lori Leibold (Ctr. for Hearing Res., Boys Town National Res. Hospital, Omaha, NE), Lauren Calandruccio (Dept. of Psychol. Sci., Case Western Reserve Univ., Cleveland, OH), Ryan McCreery (Ctr. for Hearing Res., Boys Town National Res. Hospital, Omaha, NE), Barbara Rodriguez (Dept. of Speech and Hearing Sci., Univ. of New Mexico, Albuquerque, NM), and Emily Buss (Dept. of Otolaryngology/Head and Neck Surgery, Univ. of North Carolina, Chapel Hill, NC)

The Children's English and Spanish Speech Recognition Test (ChEgSS) is a tool for assessing masked-speech recognition. An audiologist could test a bilingual child's speech recognition in both English and Spanish, but if time is limited, which language should they choose? We evaluated how two different measures of language dominance, relative demand for use and relative vocabulary size impacted word identification in noise and in a two-talker speech masker for 82 bilingual children (4–17 years) with normal hearing. The participants were exposed to Spanish from birth and varied in their initial exposure to English. Each child completed ChEgSS in English and Spanish using a 4AFC procedure. We measured vocabulary using standardized tests and demand for use using questionnaires. We created language dominance variables by standardizing values then subtracting Spanish scores from English ones. Linear regression evaluated how age and the interaction between length of exposure, relative vocabulary size, and relative language use predicted performance. Relative vocabulary size impacted thresholds, and this effect was larger for children with less exposure in the test language. Results suggest relative vocabulary size and length of exposure should inform the test language(s) used to assess clinical speech recognition.

5aSC9. Perception of post-focus compression as a cue to focus in L2 English. Sheyenne Fishero (Dept. of Linguist, Univ. of Kansas, 1541 Lilac Ln., Blake Hall Rm. 427, Lawrence, KS 66045, sfishero@ku.edu) and Jie Zhang (Dept. of Linguist, Univ. of Kansas, Lawrence, KS)

English and Mandarin both mark focus with post-focus compression (PFC) of F0 and intensity. Previous production research indicates PFC may be a universally marked feature difficult to transfer between languages, but the generally narrower pitch range and lack of fluency indicative of nonnative speech impede the ability to conclude the marked status of PFC based on production data alone. The present study utilized accuracy scores on a forced-choice prosodic restoration perception task to identify whether non-native learners of a PFC language (English) with a native PFC language L1 (Mandarin) are capable of perceiving PFC as a cue to focus when no other focus cues are available. The role of L2 proficiency in the ability to use PFC

as a cue to focus was also examined. Results indicate that native Mandarin listeners had significantly poorer accuracy compared to native English listeners at using PFC as a cue to focus in English regardless of proficiency, thus supporting PFC's role as a marked feature not easily transferred into an L2, but some Mandarin listeners' above chance performance on the task indicate it is not impossible to perceive PFC as a cue to focus in an L2 context.

5aSC10. The role of proficiency in the interlanguage speech intelligibility benefit. Sheyenne Fishero (Dept. of Linguist, Univ. of Kansas, 1541 Lilac Ln., Blake Hall Rm. 427, Lawrence, KS 66045, sfishero@ku.edu), Allard Jongman (Dept. of Linguist, Univ. of Kansas, Lawrence, KS), and Joan Sereno (Dept. of Linguist, Univ. of Kansas, Kansas City, KS)

Previous research indicates nonnative listeners may have an advantage at understanding nonnative speech of talkers with the same L1 due to shared interlanguage knowledge. The present study offers a comprehensive analysis of various factors that may modulate this advantage, including the proficiency of interlocutors, the mapping of phonemes between the L1 and L2, and the acoustic properties of the phonemes. Accuracy scores on a lexical decision task were used to identify whether native Mandarin learners of English are better than native English listeners at perceiving Mandarin-accented English speech, as well as whether they are better at perceiving Mandarin-accented English compared to native English speech. Results indicated an overall native English interlocutor advantage, an advantage for common-phoneme over unique-phoneme words in nonnative speech, and some evidence of gradience in ISIB effects based on the proficiency of talkers and listeners. No strong relationship between type of English input (native or accented) and the ability to understand Mandarin-accented English speech was found. The nonnative listener advantage at perceiving nonnative speech is relatively small and depends on various factors, including listener proficiency, speaker proficiency, phoneme characteristics, and the acoustics of specific speech tokens.

5aSC11. Perceptual training affects 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, Jacqueline M. Albor, and Olivia A. Billetdeaux (Commun. Sci. and Disord., The Penn State Univ., University Park, PA)

Understanding speech in noisy conditions is a problem faced by all listeners. Previous studies have shown that listeners understand target speech better when background speech (masker) is in a different language. This is called Linguistic Release from Masking (LRM). In the current study, we examined whether training on identifying speech in noisy conditions can modulate LRM. In a pre-test/training/post-test design, 60 monolingual American English listeners transcribed English sentences presented in noisy backgrounds. In the pre-test and post-test, all listeners transcribed sentences presented with both English and Dutch maskers without feedback. During training, participants were randomly assigned to transcribe target sentences with Dutch, English, or white noise maskers and received feedback. Results showed an LRM effect in the pre-test; participants transcribed the target sentences better with a Dutch than with an English masker. After training, participants improved in all conditions, but greater improvements in the English masker condition eliminated LRM. Results provide insight into the role of perceptual learning as well as into the nature of informational masking effects underlying LRM. This study serves as a basis for future research examining improvement for speech in speech recognition and changes in LRM.

5aSC12. Non-native talkers and listeners and perceptual benefits of clear speech. Ye-Jee Jung (Purdue Univ., 215 Nimitz Dr., Apt 4, West Lafayette, IN 47906, jung292@purdue.edu) and Olga Dmitrieva (Purdue Univ., West Lafayette, IN)

Native clear speech aids speech perception for various native populations such as hearing-impaired adults (Picheny *et al*., 1985). Compared to native speech, little is known about the benefit of non-native clear speech (Smijlanic and Bradlow, 2011). The current study investigates whether non-native clear speech can aid both native and non-native listeners, using every combination of talkers and listeners (native talker/listener, and non-native talker/listener). Non-native participants were L1 Korean speakers, while native participants were L1 American English speakers. Each group had 32 participants listening to semantically anomalous English sentences recorded by four native and four non-native talkers and presented with speech-shaped noise at 0 dB SNR in each speaking style (casual and clear). Each sentence had four keywords, and the number of correct keywords was converted to rationalized arcsine units. The results show significant effects of talker L1, listener L1, and speaking style. Native speech was more intelligible than non-native one and native listeners outperformed non-native listeners. Clear speech was more intelligible than casual speech. However, none of the interactions among the factors reached statistical significance, indicating the two groups of talkers and the two groups of listeners behaved in a parallel way in terms of producing and receiving clear speech benefits.

5aSC13. Children's perception of accent distance for native and nonnative varieties. Malachi Henry (Speech, Lang., & Hearing Sci., Indiana Univ., 2631 East Discovery Parkway, Bloomington, IN 47408, mjhenry5519@gmail.com), Tessa Bent (Speech, Lang., & Hearing Sci., Indiana Univ., Bloomington, IN), and Rachael F. Holt (Speech and Hearing Sci., Ohio State Univ., Columbus, OH)

Adults are sensitive to the presence and strength of accents different from their own. Children also are sensitive to the presence of unfamiliar accents, but much less is known about their awareness of accent strength. To address this gap, this study used a ladder task, in which listeners rank talkers based on their perceived distance from the home standard. Female adult talkers representing four native, six nonnative, and one bilingual English accents were included. Six-year-olds, 12-year-olds, and adults completed two ladders in which all talkers produced the same sentence and one ladder with a unique sentence for each talker. Average rankings were significantly correlated across age groups, with greater agreement for consistent than unique sentence ladders. Strategies used by six-year-olds differed from older children and adults. The youngest children arranged talkers into two or three distinct groups with more horizontal groupings while older children and adults primarily arranged talkers in a vertical array. These results support previous findings that young children distinguish their own accent from unfamiliar accents. Furthermore, the study provides new data demonstrating that sensitivity to accent distance emerges in the early school-aged years with continued maturation during middle childhood. [Work by NSF grants 1941691 and 1941662.]

5aSC14. Effects of age and musical expertise on speech-on-speech perception in adults. Laura Rachman (Otorhinolaryngology/Head and Neck Surgery, Univ. Medical Ctr. Groningen, Univ. of Groningen, Hanzeplein 1, Groningen 9713 GZ, Netherlands, l.rachman@rug.nl), Eleanor Harding, Ryan Gray, Stefan Smeenk (Otorhinolaryngology/Head and Neck Surgery, Univ. Medical Ctr. Groningen, Univ. of Groningen, Groningen, Netherlands), Anastasios Sarampalis (Dept. of Psych., Univ. of Groningen, Groningen, Netherlands), Etienne Gaudrain (Lyon Neurosci. Res. Ctr., CNRS UMR 5292, Inserm U1028, UCBL, UJM, Lyon, France), and Deniz Baskent (Otorhinolaryngology/Head and Neck Surgery, Univ. Medical Ctr. Groningen, Univ. of Groningen, Groningen, Netherlands)

Speech-on-speech (SoS) perception relies on perceptual mechanisms, such as discriminating mean fundamental frequency (F0) and vocal-tract length (VTL), and cognitive mechanisms, such as selective attention and working memory. Older adults may be less sensitive to F0 differences, possibly affecting their ability to perceive different speakers. Age-related cognitive changes may lead to difficulties in attention direction and inhibition. Compared to non-musicians, musicians are reported to possess enhanced processing of acoustic features such as F0, as well as enhanced cognitive abilities such as auditory attention skills and working memory. While this intuitively could lead to a musician advantage for SoS perception, reports of musicians outperforming non-musicians on SoS tasks are inconsistent across both younger and older adults. Differences in SoS paradigms across the literature have made it difficult to directly compare musicianship advantages in SoS perception in younger and older adults, or to clarify related underlying mechanisms. In this study, we investigated the extent to which older compared to younger adults benefit from voice differences in an SoS

perception task, by manipulating differences in F0 and VTL cues between target and masker speakers. Furthermore, we investigated the effect of musical expertise in younger and older adults within the same SoS task.

5aSC15. Emotion recognition in speech and music of Cantonese children with autism spectrum disorder. Fang Zhou (Hong Kong Polytechnic Univ., GH 709, Hong Kong Polytechnic University, Hong Kong 999077, Hong Kong, fangzhou@polyu.edu.hk), Si Chen, Xi Chen, Angel Wing Shan Chan, Tempo Po Yi Tang, Natalie Mak, Bebob Cheung, Sara Yee, and Claudia Fung (Hong Kong Polytechnic Univ., Hong Kong, Hong Kong)

Autism Spectrum Disorder (ASD) is a neurological and developmental disorder and people with ASD are found to have difficulties in speech communication and social interactions. Previous studies revealed that the performance of autistic children in emotion recognition from music and speech was less accurate compared to typically developing (TD) controls. However, there are few studies on emotion recognition from tonal languages and a direct comparison between speech and music. The current study examined emotion recognition by Cantonese-speaking children with and without ASD aged from 8 to 11. The stimuli for speech perception were recorded by an actor and the clips for music perception were segmented from piano songs, presenting different emotions (angry, happy, sad, fear, and tender). Results showed that ASD group showed a higher accuracy rate in recognizing fear and tender and a lower rate in recognizing angry, happy, and sad in the emotion recognition from speech, though the results did not reach significance. The identification rate from music differed significantly from the TD group, showing a lower accuracy in general. The outcome indicates that Cantonese ASD children may have difficulties in emotion recognition more from music and may have a better judgment on certain negative emotions like fear.

5aSC16. Re-evaluate the data of English vowel perception in quiet and noise for native and non-native listeners: Sensitivity and response bias. Yao Chen (Speech, Lang., and Hearing Sci., The Univ. of Texas at Austin, 2504A Whitis Ave. Stop A1100, Austin, TX 78712, chenyao@utexas.edu), Can Xu, Chang Liu (Speech, Lang., and Hearing Sci., The Univ. of Texas at Austin, Austin, TX), Mingshuang Li (Dept. of Commun. Disord. and Sci., California State Univ. at Northridge, Austin, TX), Lin Mi (The Affiliated Brain Hospital of Guangzhou Medical Univ., Guangzhou, China), and Sha Tao (State Key Lab. of Cognit. Neurosci. and Learning and IDG/McGovern Inst. for Brain Res., Beijing Normal Univ., Beijing, China)

For decades, research on vowel perception has been primarily focused on the percentage correct of identification. Limited studies take listeners' response bias into consideration and applied signal detection theory (SDT) in the data analysis of vowel identification. The goal of this study was to investigate English vowel identification in quiet, long-term speech-shaped noise (LTSS), and multi-talker babble (MTB) for English-native (EN), Chinese-native in the US (CNU), and Chinese-native in China (CNC) listeners by computing their sensitivity (measured by d') and response bias (measured by c) using the SDT. Results showed that (1) in all three listening conditions, EN listeners showed higher sensitivity and lower bias than CNU and CNC listeners; (2) in quiet and MTB, CNU group demonstrated higher sensitivity than CNC group with similar sensitivity in LTSSN between two Chinese groups; and (3) in MTB, CNU group had smaller bias than CNC group, while in quiet and LTSSN condition, two Chinese groups showed similar bias. These results suggest that the US residency may not improve Chinese-native listeners' vowel identification capacity, instead, may shift their response strategy toward native speakers' pattern. [Work by the University of Texas at Austin Research Grant and China National Natural Science Foundation 31628009.]

5aSC17. The effects of phonological content, sentence context, and vocoding on voice cue perception. Thomas Koelewijn (Otorhinolaryngology, Univ. of Groningen, Univ. Medical Ctr. Groningen (UMCG), Hanzeplein 1, Groningen 9713 GZ, Netherlands, t.koelewijn@rug.nl), Etienne Gaudrain (CNRS, Lyon Neurosci. Res. Ctr., Lyon, France), Thawab Shehab (Otorhinolaryngology, Univ. of Groningen, Univ. Medical Ctr. Groningen (UMCG), Groningen, Netherlands), Tobias Treczoks (Otorhinolaryngology, Univ. of Groningen, Univ. Medical Ctr. Groningen (UMCG), Oldenburg, Germany), and Deniz Baskent (Otorhinolaryngology, Univ. of Groningen, Univ. Medical Ctr. Groningen (UMCG), Groningen, Netherlands)

Recently, we showed the effect of lexical content on just-noticeable-differences (JNDs) in voice pitch (F0) and vocal-tract length (VTL) voice cue perception. Participants performed an adaptive 3AFC task with non-vocoded and vocoded auditory stimuli. When presented with words, participants showed smaller VTL JNDs compared to time-reversed words, also when vocoding was applied. These outcomes inspired two follow-up studies using similar methods. The first study compared words, time-reversed words, and non-words, to investigate if the benefit is related more specifically to lexical (words) and/or phonological (non-words) content. Irrespective of vocoding, reversed words had a detrimental effect on VTL perception, replicating earlier findings. However, VTL JNDs with non-words did not significantly differ from those with words, suggesting linguistic content benefits for VTL perception at a phonological level. The second study compared JNDs with words and sentences and showed, again regardless of vocoding, a benefit in processing a full sentence compared to a single word in both F0 and VTL perception, suggesting that the amount of acoustic speech information and/or semantic context led to increased voice cue sensitivity. These results improve our understanding of the interactive speech and voice perception processes, inspiring rehabilitation tools for CI users with limited access to voice information.

5aSC18. Audiovisual high variability phonetic training promotes second language lexical tone learning: An event-related potential study. Yang Zhang (Speech-Language-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, zhang470@umn.edu), Xiaojuan Zhang, Bing Cheng (Xi'an Jiaotong Univ., Xi'an, Shaanxi, China), and Yue Wang (Linguist, Simon Fraser Univ., Burnaby, BC, Canada)

This study investigated the efficacy of a modified high variability phonetic training (HVPT) protocol in second language learning. The target Mandarin lexical tones in the training stimuli were acoustically exaggerated at four levels in terms of duration, pitch range, and pitch contour. Seven audiovisual perceptual training sessions were designed to deliver the training stimuli adaptively based on the listener's identification score. Pre- and post-tests used behavioral identification and discrimination tasks with natural speech, synthetic speech, and non-speech control stimuli. ERP experiments were also conducted with synthetic speech stimuli in a passive listening oddball paradigm. A total of 24 adult monolingual American English speakers were randomly assigned to the training group and control group. Behavioral data showed significant improvement in identifying the four lexical tones in the trainees but not in the controls. There was also training-induced enhancement of categorical perception of the lexical tones for the synthetic speech stimuli. Posttest versus pretest comparison in the trainees showed an increased mismatch negativity (MMN) response with decreased MMN peak latency for the across-category lexical tone stimuli. Collectively, the results demonstrate fundamental neural sensitivity changes at the pre-attentive level underlying training-induced behavioral identification and discrimination changes in second language learners.

5aSC19. Impact of face masks on second language word identification. Yu-Jung Lin (Foreign Lang., Literatures, and Cultures, College of the Holy Cross, 1 College St., Worcester, MA 01610, yjlin@holycross.edu), Joshua Isakson (Chinese Studies/Sociology, College of the Holy Cross, Worcester, MA), and Emma Keane (German Studies/Psych., College of the Holy Cross, Worcester, MA)

The current study investigated the effects of face masks on the intelligibility of second language (L2) speech. Specifically, we examined whether L2 learners of Mandarin and English identify words in their L2s less accurately when the speakers spoke through masks. Seven Mandarin native

speakers whose L2 is English and seven English native speakers whose L2 is Mandarin were asked to identify the words they heard in videos, where English and Mandarin native speakers pronounced monosyllabic words in their native languages with and without surgical masks. The first languages (L1s) of these 14 subjects, the language of the videos, the mask conditions (with mask versus without mask), the noise conditions (quiet versus noisy), and the speaker gender (male versus female) were included in the experimental design. Preliminary results suggested that L2 word identification was significantly impacted when the background noise was present. Furthermore, the accuracy of L2 word identification in the male-speaker condition was significantly lower than the female-speaker condition when masks were worn. These findings were compared to the L1 perception data to demonstrate how the use of masks may negatively impact L2 perception to a greater degree than L1 perception.

5aSC20. Neural correlates of speech in noise perception differences between combined electric-acoustic stimulation and standard cochlear implants. Jean Hong (Dept. of Otolaryngol. Head and Neck Surgery, Univ. of Iowa Hospitals and Clinics, 516 Grandview Court, Iowa City, IA 52246, jean-hong@uiowa.edu), Hwan Shim (Univ. of Iowa, Iowa City, IA), Subong Kim (Purdue Univ., West Lafayette, IN), Marlan Hansen, Bruce Gantz (Dept. of Otolaryngol. Head and Neck Surgery, Univ. of Iowa Hospitals and Clinics, Iowa City, IA), and Inyong Choi (Univ. of Iowa, Iowa City, IA)

Cochlear implants have evolved to utilize residual acoustic hearing that combines to electric stimulation, known as electric-acoustic stimulation (EAS). However, there are mixed expectations about the benefits of EAS. A positive perspective expects that contributions from residual hearing provide better access to acoustic cues that is helpful for speech-in-noise perception. An opposing view concerns potentially poorer spectral resolution of EAS electrodes' stimulation as those electrodes are often inserted close to lateral wall, which may cause poorer speech-in-noise perception. This study aimed to directly compare neural processes of speech-in-noise perception between EAS and standard CI users to provide an answer to the above alternative expectations. We used 64-channel EEG to measure cortical evoked responses to (1) background noise and (2) target word while listeners perform a word-in-noise task. Then, we compared the amplitude ratio of evoked responses to the target word and background noise, referred to as "internal SNR," which reflects how well target sound is unmasked from the mixture of speech and noise. Based on the comparison of 55 EAS and 22 standard CI users, internal SNR was significantly larger in EAS CI users. This result indicates that EAS provides enhanced neural processes for speech unmasking.

5aSC21. Perception advantages of foreign directed speech. Rebecca Wheeler (Linguist, Univ. of Colorado, Boulder, 295 UCB, Dept. of Linguist, Boulder, CO 80309, Rebecca.Wheeler-1@colorado.edu)

Foreign directed speech (FDS) is a listener directed speech style used when native speakers interact with non-native listeners of a language. This study considers if native and non-native listeners benefit from the phonetic features of FDS in English. 43 native English speakers and non-native speakers were recruited on Amazon Mechanical Turk. Participants were presented with an audio clip and two pictures. They were asked to click on the correct image based on the audio given with reaction times recorded. Each participant was given a randomized order of speech tokens: 12 tokens from the two speech conditions, native speech (NS) and FDS. The data showed that speech condition had a significant effect on reaction time, where FDS yielded faster reaction times than native speech ($est = -0.024$, $t = 2.035$, $p = 0.04$). Listener language background did not have a significant effect, with both groups performing similarly in reaction times across both speech conditions. The results of this study show that the phonetic features of FDS are beneficial for comprehension when compared to native speech regardless of listener language background. This may be because the phonetic features of FDS are similar to the characteristics of other listener directed speech styles.

5aSC22. The cognitive effort associated with speaking clearly in noise: A pilot study. Keiko Ishikawa (Speech and Hearing Sci., Univ. of Illinois, Urbana-Champaign, 901 S. 6th St., Champaign, IL 61820, ishikak@illinois.edu), Elisabeth Coster, and Hannah Li (Speech and Hearing Sci., Univ. of Illinois, Urbana-Champaign, Champaign, IL)

Individuals with voice and speech disorders often struggle to implement therapy techniques in real-world environments. Background noise is a known distractor that deteriorates performance in cognitive and non-speech motor tasks. The noise may disturb talkers in a similar manner; however, its effect has been poorly described. This study evaluated the effect of background noise on talkers' ability to maintain clear speech, a well-documented intelligibility enhancement technique, and corresponding cognitive effort. Participants of the study were 10 native English speakers who have no history of voice and speech disorders. The participants were asked to read sentences and describe pictures in habitual speech and clear speech within two noise environments: quiet and two-talker babble noise. The participants rated their perceived level of overall effort, mental effort, physical effort, frustration, and performance using a modified NASA-TLX scale for each noise condition. The degree of cognitive effort during the tasks was also measured by pupillometry. Their voice recordings were acoustically analyzed for intensity, spectral slope, and speech rate. Preliminary findings indicate that (1) production of clear speech requires greater cognitive effort, especially in noise, and (2) the degree of agreement between the subjective rating and physical measurement of mental effort varies across the talkers.

5aSC23. Vocal emotion recognition by native Turkish children with normal hearing and with hearing aids. Gizem Babaoğlu (Audiol., Hacettepe Univ. Health Sci., Hacettepe Univ., Hacettepe Faculty of Health Sci., Ankara 06100, Turkey, g.babaoglu@umcg.nl), Başak Yazgan, Pınar Erturk (Audiol., Hacettepe Univ. Health Sci., Hacettepe Univ., Ankara, Turkey), Etienne Gaudrain, Laura Rachman (Otorhinolaryngology/Head and Neck Surgery, Univ. Medical Ctr. Groningen, Univ. of Groningen, Groningen, Netherlands), Leanne Nagels (Ctr. for Lang. and Cognition Groningen, Univ. of Groningen, Groningen, Netherlands), Stefan Launer (Phonak, Sonova AG, Staefa, Switzerland), Gurjit Singh (Phonak, Sonova AG, Mississauga, ON, Canada), Monita Chatterjee (Ctr. for Hearing Res., Boys Town National Res. Hospital, Omaha, NE), Esra Yücel, Gonca Sennaroğlu (Audiol., Hacettepe Univ. Health Sci., Hacettepe Univ., Ankara, Turkey), and Deniz Başkent (Otorhinolaryngology / Head and Neck Surgery, Univ. Medical Ctr. Groningen, Univ. of Groningen, Groningen, Netherlands)

Development of vocal emotion recognition in children with normal hearing takes many years before reaching adult-like levels. In children with hearing loss, decreased audibility and potential loss of sensitivity to relevant acoustic cues may additionally affect vocal emotion perception. Hearing aids (HAs) are traditionally optimized for speech understanding, and it is not clear how children with HAs are performing in perceiving vocal emotions. In this study, we investigated vocal emotion recognition in native Turkish normal hearing children (NHC, age range: 5–18 years), normal hearing adults (NHA, age range: 18–45 years), and children with HAs (HAC, age range: 5–18 years), using pseudo-speech sentences expressed in one of the three emotions, happy, sad, or angry (Geneva Multimodal Emotion Portrayal (GEMEP) Corpus by Banziger and Scherer, 2010; EmoHI Test by Nagels *et al.*, 2021). Visual inspection of the preliminary data suggests that performance increases with increasing age for NHC and that in general, HAC have lower recognition scores compared to NHC. Further analyses will be presented, along with acoustical analysis of the stimuli and an exploration of effects of HA settings. In addition, for cross-language comparison, these data will be compared to previously collected data with the same paradigm in children from the UK and the Netherlands.

5aSC24. Visual and acoustic influences on linguistic discrimination during interviews. Kathleen Siren (Speech-Language-Hearing Sci., Loyola Univ. Maryland, 4501 N. Charles St., Baltimore, MD 21210, ksiren@loyola.edu)

This study investigates the influence of bias during the interview process. University students of different races and ethnicities recorded predetermined responses to questions that might be asked during a graduate admissions interview. These recorded responses were played for a group of

university professors and a group of non-academicians. The listeners heard student responses under three conditions: (1) audio only, (2) audio paired with a matched picture of the student, and (3) audio paired with a picture of a student of a different race and ethnicity. Listeners rated responses along several continua, including knowledge of field, competency, and motivation. Student responses were compared across several acoustic features including fundamental frequency, pitch range, and rate. Results are discussed in terms of potential linguistic discrimination that may be exacerbated during face-to-face graduate admissions interviews and acoustic characteristics of speech that may influence perceptual bias based on race.

5aSC25. Facial posture as a phonetic prime for memory retrieval. Arian Shamei, Sijia Zhang (Linguist, UBC, Vancouver, BC, Canada), Noah Luntzlara (Linguist, UBC, 2613 West Mall, Vancouver, BC, Canada, noahidea@student.ubc.ca), Yangshuying Zhou, Sonja Frazier, Gillian de Boer, and Bryan Gick (Linguist, UBC, Vancouver, BC, Canada)

Facial postures influence the perception of speech towards sounds with similar motor configurations [Yeung and Scott, *J. Exper. Psych.* **150**, 983 (2021)]. However, it remains unknown whether facial posture can serve as a phonetic prime for the recall of speech. The present study tests whether maintaining a smile improves the recall of speech sounds with similar kinematics and somatosensory input (e.g., high front vowel /i/) [Ogane *et al.*, *J. Acoust. Soc. Am.* **148**, EL279–EL284 (2020)]. In the training phase, a list of monosyllabic words containing /i/, /a/, and /u/ vowels is presented aurally to participants maintaining a neutral expression. In the recall phase, participants are asked to recall these words in one of two conditions, maintaining either a smile or a neutral expression. Improved recall for tokens containing /i/ would suggest facial postures act as phonetic primes for memory retrieval. The experiment is ongoing; results will be presented and discussed. [Work by the NIH and NSERC].

FRIDAY MORNING, 27 MAY 2022 GOVERNORS SQUARE 14, 9:00 A.M. TO 12:00 NOON

Session 5aUW

Underwater Acoustics: General Topics in Underwater Acoustics III

Ivars P. Kirsteins, Cochair
NUWCDIVNPT, 1176 Howell St., Newport, RI 02813

Chad M. Smith, Cochair
The Pennsylvania State University, Applied Research Laboratory, State College, PA 16804

Contributed Papers

9:00

5aUW1. Passive suppression of radiation of low-frequency underwater noise. Oleg A. Godin (Phys. Dept., Naval Postgrad. School, Monterey, CA 93943, oagodin@nps.edu)

Using an exact solution of the problem of spherical wave diffraction on a fluid sphere, it has been predicted [Godin and Baynes, *J. Acoust. Soc. Am.* **143**, EL67–EL73 (2018)] that passive, broadband suppression of radiation of low-frequency sound can be achieved by placing an acoustically compliant object in the vicinity of the source. In underwater acoustics, the compliant object can be realized as an air-filled bladder. One of the promising applications of the technique is to suppress the noise generated by ship propellers. If implemented on a wide scale, the passive noise suppression technique would help to decrease the noise levels due to commercial and recreational vessels and reduce the detrimental effects of anthropogenic noise on marine life. This paper extends the earlier theoretical treatment of the problem to more realistic models of the compliant object such as a gas volume surrounded by a stretchable membrane (a balloon) and a thin shell. The effect of the finite depths of low-frequency noise source and the compliant object on noise levels in the far field will be discussed.

9:15

5aUW2. Error modeling of Doppler velocity logs using a point-based scattering model. Joshua Purviance (The Penn State Univ., 201 Appl. Sci. Bldg., Graduate Program in Acoust., C/O: Josh Purviance, University Park, PA 16802, jfp5785@psu.edu), Thomas E. Blanford, and Daniel C. Brown (The Penn State Univ., State College, PA)

As Doppler Velocity Log (DVL) systems become smaller, cheaper, and increasingly capable, they are becoming standard options on even low-end undersea vehicles. DVLs are the principal sensor that provide ground-referenced velocity measurements to compensate for the bias and drift errors found in inertial navigation systems. Therefore, DVLs fill an essential role in achieving an accurate navigation solution for unmanned undersea vehicles. Although there are many modeling and simulation techniques for comparing a fused-sensor inertial navigation solution to earth-reference, there is a sparsity in the literature for models and simulation environments that are suited for estimating the performance of a standalone DVL system. A practical model must incorporate sufficient environmental fidelity so as to match with empirical results, while still remaining computationally efficient. The point-based scattering model described by Brown *et al.* [JASA Elect. Lett. i(2017)] may be a useful tool for modeling DVL performance. The implemented model incorporates seafloor roughness, scatterer density,

textures, and is well-suited for integration into Monte Carlo simulations. This presentation will compare the DVL performance predicted by this point-based scattering model to several other error models and discuss the utility for predicting performance in natural marine environments.

9:30

5aUW3. Cancellation of multiple interferers in communication signal. Jiyoung Song (Korea Maritime and Ocean Univ., 727 Taejong-ro, Busan 49112, Republic of Korea, jysong.ualab@gmail.com), Donghyeon Kim (KIOST-KMOU OST School, Busan, Republic of Korea), and J. S. Kim (Korea Maritime and Ocean Univ., Busan, Republic of Korea)

The broadband impulse signal emitted by marine animals is difficult to separate from the communication signal and contaminates the phase of the symbol, causing performance reduction. In a previous study, the Green's function of this undesired signal was estimated for each path, and the impulse signal was spatially canceled by combining it with adaptive time-reversal processing [Kim *et al.*, *J. Acoust. Soc. Am.* **150**, A195 (2021)]. However, when the number of impulse signals to be canceled exceeds the number of receivers, the degree of freedom in the process of designing the weight vector is limited. In this presentation, we propose an approach that provides degrees of freedom by treating the signal vectors from multiple arrivals as a single constraint. The spatially nulling impulse signal without distorting the phase of the communication signal showed a remarkable improvement in communication performance in the seagoing experimental data. [Work supported by the Agency for Defense Development, South Korea, under Grant UD200010DD.]

9:45

5aUW4. Analysis of split-beam spatial coherence statistics for discerning compact from non-compact littoral sonar clutter. Chad M. Smith (Appl. Res. Lab., The Penn State Univ., The Penn State Univ., Appl. Res. Lab., State College, PA 16804, chad.smith@psu.edu), Daniel C. Brown (Appl. Res. Lab., The Penn State Univ., State College, PA), Anthony P. Lyons (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, Durham, NH), and Thomas Gabrielson (Appl. Res. Lab., The Penn State Univ., State College, PA)

Active sonar systems operating in the littoral environment are often reverberation-limited, which impacts detection performance by reducing the effective signal-to-noise ratio. Additionally, irregularities in the environment lead to excessive false alarms commonly referred to as sonar clutter. Clutter is found in all environments, but shallow-water littoral regions have been observed to be especially challenging. The irregularities that cause sonar clutter can have spatial scales ranging from much smaller to much greater than the dimensions of the sonar resolution cell. This talk will discuss physical interpretation and experimental assessment of signal statistics related to the transverse horizontal spatial coherence of reverberation estimated via split-beam processed data obtained from a line array. Multiple boundary interaction in shallow water depths and spatial variability randomize reverberation and may allow the use of Gaussian noise models. While match-filtered signal intensity is an optimal detection statistic in a Gaussian noise environment, spatial reverberation coherence statistics may provide complimentary information of the spatial scale of clutter features relative to the width of the beampattern. It is found these statistics may be used to help discern clutter types that are spatially compact from those that are non-compact when compared to the width of the sonar resolution cell.

10:00

5aUW5. Mitigating ocean random medium effects in array processing by using a cross-correlator beamformer instrumentation. Ivars P. Kirsteins (Naval Undersea Warfare Ctr., 1176 Howell St., Newport, RI 02813, i.kirsteins@gmail.com)

Ocean random medium effects such as internal waves can cause distortions to signal wave fronts that are highly detrimental to detecting and localizing signals with acoustic arrays. Motivated by the lucky imaging techniques used in astronomy [1], we have previously found evidence in real array data that when the data are looked at a much finer time scales, there can be brief moments when the wave front has little distortion [2]. However, the challenge has been developing metrics for ranking the quality of the data frames in order to exploit these lucky scintillations. In this paper, we propose a simple frame quality metric based on a cross-correlator instrumentation of the delay-sum beamformer [3] evaluated by arithmetically averaging the higher-order spatial correlation lags along the time delay trajectory corresponding to the beamformer focal location and then utilizing only the highest ranked frames in the localization. We connect this metric to beamformer focus sharpness by showing that the main lobe curvature increases monotonically with the number of lags used and demonstrate the approach on real underwater acoustic array data. [1] https://www.ast.cam.ac.uk/research/lucky [2] I.P. Kirsteins, *JASA* **143**, 1976 (2018). [3] E. Ruigrok *et al.*, "Cross-correlation beamforming," J. Seismol. *21*, 495–508 (2017).

10:15–10:30 Break

10:30

5aUW6. Simulating elastic targets for sonar algorithm development. Kyle S. Dalton (Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, ksd5377@psu.edu), Thomas E. Blanford, and Daniel C. Brown (Penn State Univ., State College, PA)

Targets interrogated during synthetic aperture sonar (SAS) applications may exhibit elastic scattering behavior and re-radiate sound after the initial geometric returns. This behavior can be a discriminating feature in target identification, but may also obscure nearby targets and make the full scene difficult to comprehend. Advancing algorithms to leverage elastic scattering for target localization and identification requires data with accurate ground truth. There is a shortage of field data containing elastic targets with sufficient quality for algorithm development. Stanton's 1984 model presents the scattering amplitude of an elastic cylinder as a function of frequency and spherical angles. This presentation uses Stanton's scattering model to simulate the time series of pressure scattered by the cylinder. This approach allows the target scattering model to be readily coupled with time series models for the environment to generate high fidelity simulated data. Simulated time series are generated for SAS collection geometries. The presentation will explore this modeling approach as a substitute for field data during initial stages of algorithm development for improved target localization. It will investigate the impact of elastic effects on imagery and discuss the limitations of methods used to isolate such effects.

10:45

5aUW7. Using the coherence of the frequency-difference autoproduct to facilitate rough surface information identification. Nicholas J. Joslyn (Appl. Phys., Univ. of Michigan, 450 Church St., Ann Arbor, MI 48109, njoslyn@umich.edu), Peter H. Dahl (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), and David R. Dowling (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

Acoustic waves, with wavenumber k and incidence angle θ, forward scattered from a randomly rough surface, with root-mean-square roughness height h, lose coherence with the incident field as $kh\cos\theta$ increases. Recovering reflected field-coherence is possible via the frequency-difference autoproduct, a quadratic product of complex field amplitudes at nearby frequencies within the signal bandwidth. By downshifting recorded frequencies, the apparent roughness of a surface is reduced and coherence can be regained. An additional consideration exists, however, as the nonlinearity of the frequency-difference autoproduct introduces a significant dependence on the surface autocorrelation function, and consequently, the surface power spectrum by Fourier transform. The relationship between coherence recovery, surface autocorrelation function, and surface power spectrum is discussed and results are shown. Furthermore, this presentation investigates the potential utility afforded by these relationships for environmental characterization in rough surface scattering. By employing coherence-based frequency-difference autoproduct methods, remote identification of the spectral content, lateral statistics, and vertical statistics of a rough surface are explored. The work uses acoustic data collected at sea during SW06 (off New Jersey, depth 80 m), where the geometry, signal bandwidth, and anisotropic ocean surface conditions put $kh\cos\theta > 2.5$. [Work supported by ONR and by the US DoD through an NDSEG Fellowship.]

11:00

5aUW8. The frame synchronization robust to low signal power in underwater acoustic communications. Kyeong pil Yang (Jeju National Univ., 102 Jejudaehak-ro, Jeju-si, Jeju 63243, Republic of Korea, did5878@naver.com), SEUNGHWAN KIM (Jeju National Univ., Jeju-si, Jeju Special Self-Governing Province, Republic of Korea), Wan-Jin Kim (Agency for Defense Development, Changwon, Republic of Korea), and Seokjun Ko (Jeju National Univ., Jeju-si, Jeju Special Self-Governing Province, Republic of Korea)

In underwater acoustic communications, it is difficult to find the starting time of received signals because of noise, multipath, transmission time delay, and Doppler frequency. In this paper, we solve the problem by using 2-STEP frame synchronization in the receiver. The transmitted signal is modulated by using burst PSK. The frame is composed of preamble of PN code and random data. The 2-STEP frame synchronization consist of non-coherent correlation, sliding FFT, and coherent correlation. First, in the STEP 1, the non-coherent correlation and sliding FFT are simultaneously performed to obtain an approximate start time of a frame in time domain and a Doppler frequency in frequency domain. However, through the results of the marine experiments, we can see that it is difficult to guarantee the starting time point in STEP 1 when the received signal power is weak and there is high Doppler frequency. Therefore, in the STEP 2, we use the coherent correlation to get more accurate start time stably after compensating the Doppler frequency of FFT in STEP 1. Finally, we can assure that the performance of the proposed 2-STEP frame synchronization is robust to low power of received signal and high Doppler frequency through marine experiments.

11:15

5aUW9. Joint equalizer and phase-locked loop for time variability in underwater acoustic communications. Seunghwan Kim (Jeju National Univ., Jeju National University, 102 Jejudaehak-ro, Jeju-si, Jeju Special Self-Governing Province 63243, Republic of Korea, tmdghks0704@gmail.com), Hyung-In Ra (Korea Maritime and Ocean Univ., Busan, Republic of Korea), Hyun-Woo Jeong, Chang-Hyun Youn (Dept. of Radio Commun. Eng., Korea Maritime and Ocean Univ., Pusan, Republic of Korea), Kyeong pil Yang (Jeju Univ., Jeju-si, Jeju, Republic of Korea), In-Soo Kim (Agency for Defense Development, Changwon, Republic of Korea), and Seokjun Ko (Electronics, Jeju National Univ., Jeju-si, Jeju Special Self-Governing Province, Republic of Korea)

In this paper, we verify the performance of the equalizer and phase-locked loop (PLL) operating jointly for channel time variability in underwater acoustic communications. We apply the decision feedback equalizer in order to eliminate channel multipath and PLL to phase rotation due to Doppler effects. In the PLL, a decision-directed $\tan^{-1}$ type phase detector is adopted. As shown in previous papers, it is difficult to make PLL to be steady state because of time-varying ISI occurred by multipath. Also, the phase rotation incurs the equalizer tap rotation. Therefore, we can see that the two subsystems of equalizer and PLL complement each other perfectly. Another issue is the convergence speed; how quickly we can make the joint equalizer and PLL to be stable state within the preamble period (known data). In order to verify the performance, the sea experiments were conducted in the condition of moving transmitter. The transmitted signal is modulated by using PSK and consists of five burst frames that are composed of preamble of PN code and random data. As a result, if we use the initial multipath value and Doppler frequency offset, we can make the joint equalizer and PLL to be the steady state more quickly.

11:30

5aUW10. Long-range underwater acoustic communication of M-eyry spread spectrum with cyclic prefix. Hyung-In Ra (Dept. of Radio Commun. Eng., Korea Maritime and Ocean Univ., 727, Taejong-ro, Yeongdo-gu, Busan, 49112, Rep. of KOREA, Busan KS012, Republic of Korea, babavivi@g.kmou.ac.kr), Chang-Hyun Youn, Ki-Man Kim (Dept. of Radio Commun. Eng., Korea Maritime and Ocean Univ., Pusan, Republic of Korea), and In-Soo Kim (Dept. of Radio Commun. Eng., Korea Maritime and Ocean Univ., Changwon, Republic of Korea)

The Spread Spectrum is an effective transmission method for complex channels in underwater acoustic (UWA) communication because of the high correlation of pseudo-random noise (PN) sequence. In particular, the spread spectrum method is effective in an environment with a low signal-to-noise ratio. The bandwidth of UWA communication channel is limited due to the fact that sound wave attenuation in water is positively associated with frequency. The limited bandwidth directly leads to the low data rate of conventional direct-sequence spread spectrum UWA communication. M-ary spread spectrum (MSS) schemes using multiple orthogonal PN sequences were proposed as a solution to this problem. However, even in the MSS method, the performance may be limited by the delay spread characteristics in the multipath propagation channels. We propose a more robust communication method in the UWA channel by including Cyclic Prefix (CP) to achieve a more robust performance in the multipath channel. In the proposed method, a CP is inserted for every symbol. In the 160 km long-range UWA communication experiment conducted in the East Sea, the proposed method showed more reliable results than the conventional MSS method. [Work supported by the Agency for Defense Development, South Korea, Grant No. UD200010DD.]

11:45

5aUW11. Underwater acoustic communication based on chirp spread spectrum with cyclic shifted PN sequences. Chang-Hyun Youn (Dept. of Radio Commun. Eng., Korea Maritime and Ocean Univ., 727, Taejong-ro, Yeongdo-gu, Busan, 49112, Rep. of KOREA, Pusan ASI KR KS012 PUSAN, Republic of Korea, younch265@g.kmou.ac.kr), Hyung-In Ra, Hyun-Woo Jeong (Dept. of Radio Commun. Eng., Korea Maritime and Ocean Univ., Pusan, Republic of Korea), Seung-Hwan Kim, Kyeong-Pil Yang (Dept. of Electron. Eng., Jeju National Univ., Jeju-do, Republic of Korea), Ki-Man Kim (Dept. of Radio Commun. Eng., Korea Maritime and Ocean Univ., Pusan, Republic of Korea), and Wan-Jin Kim (Agency for Defense Development, South Korea, Changwon, Republic of Korea)

In this presentation, we propose a new underwater acoustic communication technique that combines a binary PN (Pseudo Noise) sequence to which cyclic shift is applied and a CSS (Chirp Spread Spectrum) method that is robust against low Doppler shift. PN sequences are known to have sharp autocorrelation function and cyclic properties. In the case of the previous CSS method, 1-bit per symbol was mapped according to the sign of the slope in the time-frequency domain of the chirp signal, but the proposed method maps 2-bit per symbol according to the cyclic shift direction of PN sequence and the slope of the chirp. As a result, the transmission data rate is doubled. For example, "00" symbol is expressed as an up-chirp and right cyclic shifted PN sequence, and "11" symbol can be expressed as a down-chirp and left cyclic shifted PN sequence. This combination can have excellent autocorrelation properties and Doppler shift immunity. The performance of the proposed method will be compared with that of the previous method by presenting the results of the sea trial. [Work supported by the Agency for Defense Development, South Korea, Grant No. UD200002DD.]

ETHICAL PRINCIPLES OF THE ACOUSTICAL SOCIETY OF AMERICA FOR RESEARCH INVOLVING HUMAN AND NON-HUMAN ANIMALS IN RESEARCH AND PUBLISHING AND PRESENTATIONS

The Acoustical Society of America (ASA) has endorsed the following ethical principles associated with the use of human and non-human vertebrate animals in research, and for publishing and presentations. The principles endorsed by the Society primarily follow the form of those adopted by the American Psychological Association (APA), along with excerpts borrowed or modified from the Council for International Organizations of Medical Sciences (CIOMS) and International Council for Laboratory Animal Science (ICLAS), and the American Institute of Physics Publishing (AIPP). The ASA acknowledges the difficulty in making ethical judgments, but the ASA wishes to set minimum socially accepted ethical standards for publishing in its journals and presenting at its meetings. These Ethical Principles are based on the principle that the individual author or presenter bears the responsibility for the ethical conduct of their research and its publication or presentation.

Authors of manuscripts submitted for publication in a journal of the ASA or presenting a paper at a meeting of the Society are obligated to follow the ethical principles of the Society. Failure to accept the ethical principles of the ASA shall result in the immediate rejection of manuscripts and/or proposals for publication or presentation. False indications of having followed the Ethical Principles of the ASA may be brought to the Ethics and Grievance Committee of the ASA.

I. USE OF HUMAN SUBJECTS IN RESEARCH-Applicable when human subjects are used in the research

The ASA endorses the view that all research involving human subjects requires approval by an existing appropriate governing authority (e.g., institutional review board [IRB], Health Insurance Portability and Accountability Act [HIPPA], or by other governing authorities used in many countries) whose policies are consistent with the Ethical Principles of the ASA and adopts the requirement that all research must be conducted in accordance with an approved research protocol as a precondition for participation in ASA programs. If no such governing authority exists, the research should have met the following criteria:

Informed Consent

When obtaining informed consent from prospective participants in a research protocol, authors must have clearly and simply specified to the participants beforehand:

1. The purpose of the research, the expected duration of the study, and all procedures that were to be used.
2. The right of participants to decline to participate and to withdraw from the research in question after participation began.
3. The foreseeable consequences of declining or withdrawing from a study.
4. Anticipated factors that may have influenced a prospective participant's willingness to participate in a research project, such as potential risks, discomfort, or adverse effects.
5. All prospective research benefits.
6. The limits of confidentiality.
7. Incentives for participation.
8. Whom to contact for questions about the research and the rights of research participants. That office/person must have willingly provided an atmosphere in which prospective participants were able to ask questions and receive answers.

Authors conducting intervention research involving the use of experimental treatments must have clarified, for each prospective participant, the following issues at the outset of the research:

1. The experimental nature of the treatment;
2. The services that were or were not to be available to the control group(s), if appropriate;
3. The means by which assignment to treatment and control groups were made;
4. Available treatment alternatives if an individual did not wish to participate in the research or wished to withdraw once a study had begun; and
5. Compensation for expenses incurred as a result of participating in a study including, if appropriate, whether reimbursement from the participant or a third-party payer was sought.

Informed Consent for Recording Voices and Images in Research

Authors must have obtained informed consent from research participants prior to recording their voices or images for data collection unless:

1. The research consisted solely of naturalistic observations in public places, and it was not anticipated that the recording would be used in a manner that could have caused personal identification or harm, or
2. The research design included deception. If deceptive tactics were a necessary component of the research design, consent for the use of recordings was obtained during the debriefing session.

Client/Patient, Student, and Subordinate Research Participants

When authors conduct research with clients/patients, students, or subordinates as participants, they must have taken steps to protect the prospective participants from adverse consequences of declining or withdrawing from participation.

Dispensing with Informed Consent for Research

Authors may have dispensed with the requirement to obtain informed consent when:

1. It was reasonable to assume that the research protocol in question did not create distress or harm to the participant and involves:
    a. The study of normal educational practices, curricula, or classroom management methods that were conducted in educational settings
    b. Anonymous questionnaires, naturalistic observations, or archival research for which disclosure of responses would not place participants at risk of criminal or civil liability or damage their financial standing, employability, or reputation, and confidentiality
    c. The study of factors related to job or organization effectiveness conducted in organizational settings for which there was no risk to participants' employability, and confidentiality.
2. Dispensation is permitted by law.
3. Research involving the collection or study of existing data, documents, records, pathological specimens, or diagnostic specimens, if these sources are publicly available or if the information is recorded by the investigator in such a manner that subjects cannot be identified, directly or through identifiers linked to the subjects.

Offering Inducements for Research Participation

(a) Authors must not have made excessive or inappropriate financial or other inducements for research participation when such inducements are likely to coerce participation.

(b) When offering professional services as an inducement for research participation, authors must have clarified the nature of the services, as well as the risks, obligations, and limitations.

Deception in Research

(a) Authors must not have conducted a study involving deception unless they had determined that the use of deceptive techniques was justified by the study's significant prospective scientific, educational, or applied value and that effective non-deceptive alternative procedures were not feasible.

(b) Authors must not have deceived prospective participants about research that is reasonably expected to cause physical pain or severe emotional distress.

(c) Authors must have explained any deception that was an integral feature of the design and conduct of an experiment to participants as early as was feasible, preferably at the conclusion of their participation, but no later than at the conclusion of the data collection period, and participants were freely permitted to withdraw their data.

Debriefing

(a) Authors must have provided a prompt opportunity for participants to obtain appropriate information about the nature, results, and conclusions of the research project for which they were a part, and they must have taken reasonable steps to correct any misconceptions that participants may have had of which the experimenters were aware.

(b) If scientific or humane values justified delaying or withholding relevant information, authors must have taken reasonable measures to reduce the risk of harm.

(c) If authors were aware that research procedures had harmed a participant, they must have taken reasonable steps to have minimized the harm.

II. HUMANE CARE AND USE OF NON-HUMAN VERTEBRATE ANIMALS IN RESEARCH-Applicable when non-human vertebrate animals are used in the research

The advancement of science and the development of improved means to protect the health and well-being of both human and non-human vertebrate animals often require the use of animals in research, education, and testing. The ASA remains committed to ensuring the health and welfare of vertebrate animals used for these purposes. Vertebrate animal experiments should have been undertaken only after due consideration of the relevance for health, conservation, and the advancement of scientific knowledge. (Modified from the Council for International Organizations of Medical Sciences (CIOMS) and International Council for Laboratory Animal Science (ICLAS) document: “International Guiding Principles for Biomedical Research Involving Animals-2012”).

The ASA endorses the view that all research involving non-human vertebrate animals, hereinafter referred as “animals,” requires approval by an existing appropriate governing authority (e.g., an institutional animal care and use committee [IACUC]) whose policies are consistent with the Ethical Principles of the ASA and adopts the requirement that all research must be conducted in accordance with an approved research protocol as a precondition for participation in ASA programs. If no such governing authority exists, the research should meet the following criteria:

1. Animals have been used only when necessary and when no alternative methods, such as non-animal approaches, mathematical models, or computer simulation, are available to achieve the scientific goals.
2. Investigators have handled all animals in compliance with all current federal, state, and local laws and regulations, and with professional standards.
3. Investigators have made all reasonable efforts to minimize the number of animals used in research to achieve the scientific goals.
4. Investigators are experienced in the care of laboratory animals, supervise all procedures involving animals, ensure all subordinates who use animals have received proper training in methodology and animal care, and assume responsibility for the comfort, health, and humane treatment of experimental animals under all circumstances.
5. The health and welfare of animals are the primary considerations in making decisions of animal care including acquisition, housing, veterinary care, and final disposition of animals.
6. All surgical procedures have been conducted under appropriate anesthesia and followed techniques that avoided infection and minimized pain during and after surgery.
7. Investigators have made all reasonable efforts to monitor and mitigate any possible adverse effects to animals as a result of the experimental protocol. Strategies to manage, mitigate, and minimize any pain and/or distress in animals should be developed in consultation with a qualified veterinarian or scientist. Animals that suffer chronic pain, distress or discomfort that cannot be relieved should be removed from the study and/or euthanized using a procedure appropriate for the species and condition of the animal.
8. Investigators proceed to rapidly and humanely terminate an animal’s life when it is necessary and appropriate, always minimizing pain and always in accordance with accepted procedures as determined by a veterinarian and/or appropriate review board.

III. PUBLICATION and PRESENTATION ETHICS-For publications in ASA journals and presentations at ASA sponsored meetings

Statement of Ethics and Responsibilities

The mission of the ASA is to generate, disseminate, and promote the knowledge and practical applications of acoustics. To that end, it is essential that all authors of papers in ASA journals and presentors at ASA-sponsored meetings conduct themselves in accord with the highest level of professional ethics and standards.

By submitting a manuscript to an ASA journal, each author explicitly confirms that the manuscript meets the highest ethical standards. The same is required for material presented at meetings. Authors submitting to ASA journals should also adhere to the policies included in the particular journals’ Instructions for Contributors.

This section is mainly based on the policies of the American Institute of Physics Publishing.

Plagiarism

Plagiarism is the unauthorized and unacknowledged use of someone else’s words, ideas, processes, data, or results in a manner that can mislead others into thinking the material is your own. Plagiarism can also be in the form of text recycling, also called self-plagiarism, where an author reuses portions of text from their own work that isn’t properly credited. Plagiarism or self-plagiarism constitutes unethical scientific behavior and is never acceptable.

Publication Credit

Authorship should be limited to those who have made a significant contribution to the concept, design, execution or interpretation of the research study. All those who have made significant contributions should be offered the opportunity to be listed as authors. The author who submits a paper for publication or an abstract for presentation and publication should ensure that all coauthors have seen the final version of the paper or abstract and have agreed to its submission. Other individuals who have contributed to the study should be acknowledged, but not identified as authors.

Proper acknowledgment of the work of others used in a research project must always be given. Information obtained privately, as in conversation, correspondence, or discussion with third parties, should not be used or reported without explicit permission from the investigator with whom the information originated. Information obtained in the course of confidential services, such as refereeing manuscripts or grant applications, cannot be used without permission of the author of the work being used.

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The results of research should be recorded and maintained in a form that allows analysis and review, both by collaborators before publication and by other scientists for a reasonable period after publication. Exceptions may be appropriate in certain circumstances in order to preserve privacy, to assure patent protection, or for similar reasons.

Reporting Errors in Publication

All coauthors have an obligation to provide prompt retractions or correction of errors in published works.

Fabrication of Data and Selective Reporting of Data

Fabrication of data is an egregious departure from the expected norms of scientific conduct, as is the selective reporting of data with the intent to mislead or deceive, as well as the theft of data or research results from others.

Disclosure of Conflicts of Interest

A conflict of interest is anything that interferes with, or could reasonably be perceived as interfering with, the full and objective presentation of articles in the ASA journals and presentations at the ASA meetings. Author(s) have the obligation to disclose any personal interest or relationship that has the potential to be affected by publication of the submitted manuscript or presentation at ASA meeting:

1. The complete affiliation(s) of each author and sources of funding for the published or presented research should be clearly described in the paper or publication abstract.
2. If the publication or presentation of the research would directly lead to the financial gain of the author(s), then a statement to this effect must appear in the acknowledgment section of the paper or presentation abstract or

in a footnote of a paper. Authors must report any financial interest in corporate or commercial entities dealing with the subject matter of the manuscript or presentation.

3. If the research that is to be published or presented is in a controversial area and the publication or presentation presents only one view in regard to the controversy, then the existence of the controversy and this view must be provided in the acknowledgment section of the paper or presentation abstract or in a footnote of a paper. It is the responsibility of the author to determine if the paper or presentation is in a controversial area and if the person is expressing a singular view regarding the controversy.

Authors must submit corrections if conflicts of interests are revealed after publication.

Approved by the Executive Council on 9 December 2019.

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The Acoustical Society is grateful for the financial assistance being given by the Sustaining Members listed below and invites applications for sustaining membership from other individuals or corporations who are interested in the welfare of the Society.

Application for membership may be made to the Executive Director of the Society and is subject to the approval of the Executive Council. Dues of $1000.00 for small businesses (annual gross below $100 million) and $2000.00 for large businesses (annual gross above $100 million or staff of commensurate size) include a subscription to the *Journal* as well as a yearly membership certificate suitable for framing. Small businesses may choose not to receive a subscription to the *Journal* at reduced dues of $500/year.

Additional information and application forms may be obtained from Elaine Moran, Office Manager, Acoustical Society of America, 1305 Walt Whitman Road, Suite 110, Melville, NY 11747-4300. Telephone: (516) 576-2360; E-mail: elaine@acousticalsociety.org

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The Bylaws provide that any person, corporation, or organization contributing annual dues as fixed by the Executive Council shall be eligible for election to Sustaining Membership in the Society.

Dues have been fixed by the Executive Council as follows: $1000 for small businesses (annual gross below $100 million); $2000 for large businesses (annual gross above $100 million or staff of commensurate size). Dues include one year subscription to *The Journal of the Acoustical Society of America* and programs of Meetings of the Society. Please do not send dues with application. Small businesses may choose not to receive a subscription to the *Journal* at reduced dues of $500/year. If elected, you will be billed.

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Applicants may apply for one of four grades of membership, depending on their qualifications: Student Member, Associate Member, Corresponding Electronic Associate Member or full Member. To apply for Student Membership, fill out Parts I and II of the application; to apply for Associate, Corresponding Electronic Associate, or full Membership, or to transfer to these grades, fill out Parts I and III.

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Corresponding Electronic Associate: Any individual residing in a developing country who wishes to have access to ASA's online publications only including *The Journal of the Acoustical Society of America* and Meeting Programs [see http://acousticalsociety.org/membership/membership_and_benefits]. Dues $50 per year.
Member: Any person active in acoustics, who has an academic degree in acoustics or in a closely related field or who has had the equivalent of an academic degree in scientific or professional experience in acoustics, shall be eligible for election to Membership in the Society. A nonmember applying for full Membership will automatically be made an interim Associate Member, and must submit $115 with the application for the first year's dues. Election to full Membership may require six months or more for processing; dues as a full Member will be billed for subsequent years.

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CITY	STATE OR PROVINCE	ZIP OR POSTAL CODE	COUNTRY
NAME OF ORGANIZATION OR BUSINESS			
DEPARTMENT			
ORGANIZATION ADDRESS (STREET & NUMBER)			
CITY	STATE OR PROVINCE	ZIP OR POSTAL CODE	COUNTRY
BUSINESS TELEPHONE: AREA CODE/NUMBER	FAX: AREA CODE/NUMBER	HOME TELEPHONE: AREA CODE/NUMBER	
E-MAIL ADDRESS: (PRINT CLEARLY)	MOBILE PHONE: AREA CODE/NUMBER		
DATE AND PLACE OF BIRTH (Req'd for Awards and Emeritus Status)	SEX: ☐ Female ☐ Male ☐ Non-Binary ☐ Transgender ☐ Prefer not to answer ☐		
HIGHEST ACADEMIC DEGREE	DATE OF DEGREE	FIELD	INSTITUTION GRANTING DEGREE
OTHER DEGREE	MONTH/YEAR	FIELD	INSTITUTION GRANTING DEGREE

CHECK PERFERRED ADDRESS FOR MAIL: ☐ HOME ☐ ORGANIZATION

Part I Continued ➛

PART I CONTINUED: ACOUSTICAL AREAS OF INTEREST TO APPLICANT. Indicate your three main areas of interest below, using 1 for your main interest, 2 for your second, and 3 for your third interest. (DO NOT USE CHECK MARKS.)

- ☐ ACOUSTICAL OCEANOGRAPHY **M**
- ☐ ANIMAL BIOACOUSTICS **L**
- ☐ ARCHITECTURAL ACOUSTICS **A**
- ☐ BIOMEDICAL ACOUSTICS **K**
- ☐ COMPUTATIONAL ACOUSTICS **O**
- ☐ ENGINEERING ACOUSTICS **B**
- ☐ MUSICAL ACOUSTICS **C**
- ☐ NOISE & NOISE CONTROL **D**
- ☐ PHYSICAL ACOUSTICS **E**
- ☐ PSYCHOLOGICAL & PHYSIOLOGICAL ACOUSTICS **F**
- ☐ SIGNAL PROCESSING IN ACOUSTICS **N**
- ☐ SPEECH COMMUNICATION **H**
- ☐ STRUCTURAL ACOUSTICS & VIBRATION **G**
- ☐ UNDERWATER ACOUSTICS **J**

PART II: APPLICATION FOR STUDENT MEMBERSHIP

NAME AND ADDRESS OF COLLEGE OR UNIVERSITY WHERE PRESENTLY ENROLLED		
DEGREE EXPECTED	MONTH & YEAR DEGREE EXPECTED	NUMBER OF SEMESTER HOURS ATTENDED THIS SEMESTER
PRINT NAMES & E-MAIL ADDRESSES OF TWO FACULTY MEMBERS CERTIFYING THAT YOU ARE REGISTERED FOR AT LEAST ONE-HALF OF FULL TIME		
SIGNATURES OF THE TWO FACULTY MEMBERS LISTED ABOVE CERTIFYING THAT YOU ARE REGISTERED AT LEAST HALF TIME		
SIGNATURE OF APPLICANT		DATE

PART III: APPLICATION FOR ASSOCIATE MEMBERSHIP, CORRESPONDING ELECTRONIC ASSOCIATE MEMBERSHIP OR FULL MEMBERSHIP (and interim Associate Membership)

SUMMARIZE YOUR MAJOR PROFESSIONAL EXPERIENCE on the lines below: list employers, duties and position titles, and dates, beginning with your present position. Attach additional sheets if more space is required.

CONTRIBUTIONS TO ACOUSTICS: LIST MAIN PUBLICATIONS, PATENTS, ETC. Attach separate sheets if required.

SPONSORS AND REFERENCES: An application for full Membership requires the names, and email addresses of two references who must be **full Members or Fellows** of the Acoustical Society. Names and signatures are NOT required for Associate Membership, Corresponding Electronic Associate Membership or Student Membership applications.

PRINT NAME OF REFERENCE (required for Full Member applications only)	PRINT NAME OF SECOND REFERENCE (required for Full Member applications only)
EMAIL ADDRESS OF REFERENCE	EMAIL ADDRESS OF SECOND REFERENCE
SIGNATURE OF APPLICANT	DATE

MAIL THIS COMPLETED APPLICATION, WITH APPROPRIATE PAYMENT TO: ACOUSTICAL SOCIETY OF AMERICA, 1305 WALT WHITMAN ROAD, SUITE 110, MELVILLE, NY 11747-4300; FAX: 631-923-2875

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☐ Check or money order enclosed for $ ____________ (U.S. funds/drawn on U.S. bank)

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Expiration Date Mo. ☐☐ Yr. ☐☐

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Regional Chapters and Student Chapters

Anyone interested in becoming a member of a regional chapter or in learning if a meeting of the chapter will be held while he/she is in the local area of the chapter, either permanently or on travel, is welcome to contact the appropriate chapter representative. Contact information is listed below for each chapter representative.

Anyone interested in organizing a regional chapter in an area not covered by any of the chapters below is invited to contact the Cochairs of the Committee on Regional Chapters for information and assistance: Evelyn Hoglund, Ohio State University, hoglund1@osu.edu and Sandra Guzman, Shure, Inc., guzman_sandra@shure.com

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