# TECHNICAL PROGRAM CALENDAR
## 187th Meeting of the Acoustical Society of America
## 19-21-November 2024

**Please refer to the Itinerary Planner for Updated Information**

**Tuesday Morning, 19 November**

10:00 1aAA **Architectural Acoustics, Compuational Acoustics, and Noise:** Modeling Techniques and Computer Tools for Architectural Acoustics

10:00 1aAB **Animal Bioacoustics:** Spanning the Career Stages—A Conversation About Work in Bioacoustics

10:00 1aAO **Acoustical Oceanography, Animal Bioacoustics, and Underwater Acoustics:** Degree in Progress: Student Led Research

10:00 1aBA **Biomedical Acoustics:** Transitioning Technology from Idea to Industry

10:00 1aCA **Computational Acoustics, Education in Acoustics, and Musical Acoustics:** Artificial Intelligence vs. Human Listening to Music: Problems and Solutions

10:00 1aEA **Engineering Acoustics:** Show Your Work: Lab Visits and Demos

10:00 1aED **Education in Acoustics:** Teaching and Learning Acoustics with Jupyter Notebooks

10:00 1aMU **Musical Acoustics:** General Topics in Musical Acoustics I

10:00 1aNS **Noise:** Advanced Noise Control Design and its Benefits to Humans and Society

10:00 1aPA **Physical Acoustics and Engineering Acoustics:** Student Video Competition

10:40 1aPP **Psychological and Physiological Acoustics:** My Favorite Graph in Auditory Science

10:00 1aSA **Structural Acoustics and Vibration, Engineering Acoustics, and Physical Acoustics:** Acoustic Metamaterials

10:00 1aSC **Speech Communication:** Student Led Tutorials

10:00 1aSP **Signal Processing in Acoustics:** Signal Processing Potpourri I

10:00 1aUW **Underwater Acoustics, Acoustical Oceanography, and Engineering Acoustics:** Webinar on Python and Data Analysis

**Tuesday Afternoon, 19 November**

3:00 1pAA **Architectural Acoustics:** Acoustics of Sustainable Building Assemblies and More

3:00 1pAB **Animal Bioacoustics:** Acoustical DJ: Mixed & Matched Research Topics on Sound

3:00 1pAO **Acoustical Oceanography:** Acoustical Oceanography: Teaching Curriculum and In-class Demos

3:00 1pBA **Biomedical Acoustics:** Bubble-Based Therapies

3:00 1pEA **Engineering Acoustics:** Engineering Acoustics Lightning Round

3:00 1pED **Education in Acoustics:** What's That Sound? (Sounds of my Research)

3:00 1pMU **Musical Acoustics:** Evolving Technologies for Telematic Music Connections

3:00 1pNS **Noise and Practitioners and Industry:** Career Paths in Noise Control

3:00 1pPA **Physical Acoustics:** Current Trends in Physical Acoustics

3:00 1pPP **Psychological and Physiological Acoustics:** Psychological and Physiological Acoustics I

3:00 1pSA **Structural Acoustics and Vibration:** Lab Tours in Structural Acoustics and Vibrations

3:00 1pSCa **Speech Communication:** Student Choose-Your-Own-Adventure - Acquisition and Bilingualism

3:00 1pSCb **Speech Communication:** Student Choose-Your-Own-Adventure - Speech Articulation

3:00 1pSCc **Speech Communication:** Student Choose-Your-Own-Adventure - Speech Perception

3:00 1pSCd **Speech Communication:** Student Choose-Your-Own Adventure – Production

3:00 1pSCe **Speech Communication:** Student Choose-Your-Own Adventure – Suprasegmentals and Prosody

3:00 1pSCf **Speech Communication:** Student Choose-Your-Own-Adventure - Listening Environment, Instruments, and Tools

3:00 1pUW **Underwater Acoustics and Acoustical Oceanography:** Lightning Round Led by Students

**Wednesday Morning, 20 November**

10:00 2aAA **Architectural Acoustics, Noise, and ASA Committee on Standards:** Heating Ventilating and Air Conditioning (HVAC) Noise Challenges and Solutions

10:00 2aAB **Animal Bioacoustics:** Animal Bioacoustics Virtual Lab Tours

10:00 2aAO **Acoustical Oceanography, Animal Bioacoustics, and Underwater Acoustics:** Careers in Underwater Sound

10:00 2aBA **Biomedical Acoustics:** Debate: Nanobubbles—Can They Do Anything?

10:00 2aCA **Computational Acoustics, Education in Acoustics, Engineering Acoustics, Physical Acoustics, and Underwater Acoustics:** Interactive Computational Acoustics Demonstrations

10:00 2aEA **Engineering Acoustics:** General Topics in Engineering Acoustics

10:00 2aED **Education in Acoustics:** Tones, Tines, and Tings—Virtual Demonstration Show by David Cotton

10:00 2aMU **Musical Acoustics:** General Topics in Musical Acoustics II

10:00 2aNS **Noise:** International Aircraft Noise Regulation

10:00 2aPA **Physical Acoustics:** Hot Topics in Physical Acoustics

10:00 2aPP **Psychological and Physiological Acoustics:** Lightning Round Competition

10:00 2aSA **Structural Acoustics and Vibration:** Careers in Structural Acoustics and Vibrations

10:00 2aSC **Speech Communication:** In Honor of Ken Stevens' 100th Birthday

10:00 2aSP **Signal Processing in Acoustics and Computational Acoustics:** Tutorial on Machine Learning for Acoustics

10:00 2aUW **Underwater Acoustics, Acoustical Oceanography, and Engineering Acoustics:** Instrumentation/Lab Show and Tell

**Wednesday Afternoon, 20 November**

3:00 2pID **Interdisciplinary:** Keynote Lecture

**Thursday Morning, 21 November**

10:00 3aAA **Architectural Acoustics, Noise, and Structural Acoustics and Vibration:** The Unknown Unknowns: A Deep Look at Uncertainty and Precision in Architectural Acoustics

10:00 3aAB **Animal Bioacoustics and Student Council:** Students and Early Career Animal Bioacoustics Talks

10:00 3aAO **Acoustical Oceanography:** What's That Sound?

10:00 3aBA **Biomedical Acoustics:** Biomedical Ultrasound Tutorials: 1) Hydrophone Measurement Methods for Biomedical Ultrasound, and 2) Passive Cavitation Detection and Passive Acoustic Mapping Methods and Uses

10:00 3aCA **Computational Acoustics:** Innovations in Computational Acoustics I

10:00 3aED **Education in Acoustics and Student Council:** Student 5-minute Elevator Pitch

10:00 3aMU **Musical Acoustics:** Demonstrations of Measurement Techniques

10:00 3aNS **Noise, Architectural Acoustics, and ASA Committee on Standards:** Interventions in Soundscape

10:00 3aPA **Physical Acoustics and Engineering Acoustics:** Frugal Acoustics I

10:00 3aPP **Psychological and Physiological Acoustics:** Panel on Remote Testing

10:00 3aSCa **Speech Communication:** Choose-Your-Own-Adventure - Speech Production

10:00 3aSCb **Speech Communication:** Choose-Your-Own-Adventure - Speech Production and Technology

10:00 3aSCc **Speech Communication:** Choose-Your-Own-Adventure - Perceiving Speech

10:00 3aSCd **Speech Communication:** Choose-Your-Own-Adventure - Speech Potpourri I

10:00 3aSCe **Speech Communication:** In Honor of Ken Stevens - 100th Birthday

10:00 3aSCf **Speech Communication:** Choose-Your-Own-Adventure - Speech Potpourri II

10:00 3aSP **Signal Processing in Acoustics and Computational Acoustics:** Explainable Artificial Intelligence

10:00 3aUW **Underwater Acoustics and Acoustical Oceanography:** Invited Speaker Talk Outside the Field of Underwater Acoustics

**Thursday Afternoon, 21 November**

3:00 3pAA **Architectural Acoustics:** Potpourri Acoustics

3:00 3pAB **Animal Bioacoustics:** Climate Change and Animal Bioacoustics

3:00 3pAO **Acoustical Oceanography and Underwater Acoustics:** Program Managers Roundtable (Cosponsored by: UW)

3:00 3pBA **Biomedical Acoustics:** Show Me Your Lab Crib!

3:00 3pCA **Computational Acoustics:** Innovations in Computational Acoustics II

3:00 3pEA **Engineering Acoustics and Physical Acoustics:** Frugal Acoustics: Panel Discussion

3:00 3pED **Education in Acoustics:** Where Do You Get Your Inspiration?

3:00 3pMUa **Musical Acoustics:** General Panel Discussion on Music Acoustics Topics

7:00 3pMUb **Musical Acoustics:** VIMEO Livestream Concert: The Telematic Circle – Celebrating Free Music performed live over the internet

3:00 3pNS **Noise:** A World of Vehicle Noise

3:00 3pPP **Psychological and Physiological Acoustics:** Psychological and Physiological Acoustics II

3:00 3pSA **Structural Acoustics and Vibration:** General Topics in Structural Acoustics and Vibrations

3:00 3pSC **Speech Communication:** The State of the Art in Speech Communication

3:00 3pSP **Signal Processing in Acoustics:** Signal Processing Potpourri II

3:00 3pUW **Underwater Acoustics, Acoustical Oceanography, and Animal Bioacoustics:** Trivia Among AO/AB/UW

# Session 1aAA

## Architectural Acoustics: Modeling Techniques and Computer Tools for Architectural Acoustics

Laura C. Brill, Cochair
*Threshold Acoustics, 141 W Jackson Blvd., Suite 2080, Chicago, IL 60604*

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

### *Invited Papers*

**10:00**

**1aAA1. Using ray tracing modeling tools for room acoustics.** Ana M. Jaramillo (Olson Sound Design LLC, 8717 Humboldt Ave. N, Brooklyn Park, MN 55444, ana.jaramillo@afmg.eu) and Bruce Olson (Olson Sound Design LLC, Brooklyn Park, MN)

When designing a venue (new construction or remodel) with specific acoustics requirements, acousticians usually rely on room acoustics software for the prediction of specific parameters to match a set of predefined goals. This presentation will discuss the benefits and drawbacks of using geometrical acoustics software for room acoustics and best practices for defining appropriate settings and evaluating the reliability of predictions.

**10:25**

**1aAA2. Environmental noise computer modeling in CadnaA.** David Manley (DLR Group, 6457 Frances St., Omaha, NE 68106, dmanley@dlrgroup.com) and Steph Ahrens (DLR Group, Omaha, NE)

To calculate environmental noise levels over large sites with complex terrain and noise environments, such as mechanical equipment, generators, industrial, and traffic noise sources, it is sensible to implement an acoustic model as a part of the evaluation. This presentation includes a review of the CadnaA Noise Prediction Software from DataKustik and how it can be used to help visualize the noise environment of a site or onto a building façade. We will discuss field measurements of existing sites, sound power data of noise sources, checking the model for accuracy, engineering assumptions, and common breaking points for inaccuracies in CadnaA.

**10:50**

**1aAA3. Beginner and intermediate concepts in building systems noise predictions.** Brandon Cudequest (Threshold Acoustics, 141 W Jackson Blvd., Suite 2080, Chicago, IL 60604, bcudequest@thresholdacoustics.com) and Robert D. Miller (Threshold Acoustics, Chicago, IL)

Predictions of building system noise are a daily task for acoustical consultants. Though each consultant may rely on a different set of tools, these calculations generally trace back to guidelines and algorithms established by the American Society of Heating, Refrigerating, and Air-Conditioning Engineers (ASHRAE). Several software programs incorporate these algorithms, which can expedite the calculation process over look-up tables. This presentation will start with a survey of calculation approaches and the core assumptions on which they are based. Next, the presentation will shift to aspects of ductwork and fan design that can challenge these assumptions and how to model these intermediate concepts. Early career acousticians will come away equipped with a deeper understanding of building system noise calculation techniques.

**11:15–12:00 Discussion**

## Session 1aAB

# Animal Bioacoustics: Spanning the Career Stages–A Conversation About Work in Bioacoustics

Laura Kloepper, Chair
*Dept. of Biological Sciences, Univ. of New Hampshire, 230 Spaulding Hall, Durham, NH 03824*

**Chair's Introduction—10:00**

### *Invited Papers*

**10:05**

**1aAB1. Career roundtable–Mid career.** Laura Kloepper (Dept. of Biological Sciences, Univ. of New Hampshire, 230 Spaulding Hall, Durham, NH 03824, laura.kloepper@unh.edu)

This session will gather participants spanning students to late career professionals to have a conversation about work in bioacoustics. Each participant will give a brief introduction on their background and career path with ample time allotted for Q&A. Topics will include professional development, the importance of mentors, work/life balance, integrating family into careers, and more!

**10:10**

**1aAB2. From blind cave fish to anthropogenic sound: Fifty (plus) years of bioacoustics research.** Arthur N. Popper (Univ. of Maryland, Biology/Psychology Bldg., College Park, MD 20742, apopper@umd.edu)

This session will gather participants spanning students to late career professionals to have a conversation about work in bioacoustics. Each participant will give a brief introduction on their background and career path with ample time allotted for Q&A. Topics will include professional development, the importance of mentors, work/life balance, integrating family into careers, and more! I will briefly discuss how my work has "evolved," and the importance of evolution in one's scholarly career.

**10:15**

**1aAB3. An elder millennial in late early career stage.** Kerri D. Seger (Applied Ocean Sciences, 2127 1/2 Stewart St., Santa Monica, CA 90404, kerri.seger.d@gmail.com)

The "Spanning the Career Stages—A Conversation About Work in Bioacoustics" session will gather participants spanning students to late career professionals to have a conversation about work in bioacoustics. Each participant will give a brief introduction on their background and career path with ample time allotted for Q&A. As an elder millennial, I am currently in the phase between "early career" and "established professional." Topics will include professional development, the importance of mentors, work/life balance, integrating family into careers, and more. My experience, thus far, will allow me to speak about topics, such as morphing a music/bio undergraduate degree into a more physics-heavy research role, the transition from graduate school through post-doc to industry, mentoring students in my second language, advising as a side gig, and/or lessons learned about how business at a small research start-up differs from a FAANG-style work environment or academia.

**10:20**

**1aAB4. Participation in bioacoustics career discussion–Jennifer Miksis-Olds.** Jennifer Miksis-Olds (Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824, j.miksisolds@unh.edu)

This session will gather participants spanning students to late career professionals to have a conversation about work in bioacoustics. Each participant will give a brief introduction on their background and career path with ample time allotted for Q&A. Topics will include professional development, the importance of mentors, work/life balance, integrating family into careers, and more!

**10:25**

**1aAB5. Spanning the career stages: 3rd year Ph.D. student.** Valerie M. Eddington (Biological Science, Univ. of New Hampshire, 6 Stonecroft, Apt. 6, Portsmouth, NH 03801, valerie.eddington@unh.edu)

I am a 3rd year Ph.D. student at the University of New Hampshire in both the Ecological Acoustics and Behavior Lab and the Quantitative Marine Ecology Lab. This session will gather participants spanning students to late career professionals to have a conversation about work in bioacoustics. Each participant will give a brief introduction on their background and career path with ample time allotted for Q&A. Topics will include professional development, the importance of mentors, work/life balance, integrating family into careers, and more!

**10:30**

**1aAB6. The life and times of a comparative bioacoustician.** Micheal Dent (Univ. at Buffalo, SUNY, B76 Park Hall, Buffalo, NY 142260, mdent@buffalo.edu)

My Comparative Bioacoustics Laboratory investigates acoustic communication in animals. We conduct psychoacoustic studies of hearing using operant conditioning techniques, we perform preference tests in naïve animals, and we record sonic and ultrasonic vocalizations from our subjects in various contexts. Current lines of investigation in the mouse lab include simple measures of auditory processing such as time and intensity perception, the perception of ultrasonic vocalizations, the role of noise in masking of simple and complex signals, and the preference for certain vocalizations over others. We also measure hearing across the lifespan in different strains of mice, including genetic knockouts for Alzheimer's and Autism Spectrum Disorder. These studies are designed to better understand acoustic communication in animals across the lifespan and also serve as valuable tools to validate the mouse as a model for studies on aging and the environmental and genetic factors involved in hearing loss. [Work supported by NIH AG081747.]

**10:35**

**1aAB7. Spanning the career stages—A conversation about work in bioacoustics.** Marissa Garcia (Natural Resources and the Environment, Cornell Univ., 159 Sapsucker Woods Rd., Cornell Lab. of Ornithology - K. Lisa Yang Center for Conservation Bioacoustics, Ithaca, NY 14850, mg2377@cornell.edu)

This session will gather participants spanning students to late career professionals to have a conversation about work in bioacoustics. Each participant will give a brief introduction on their background and career path with ample time allotted for Q&A. Topics will include professional development, the importance of mentors, work/life balance, integrating family into careers, and more!

**10:40**

**1aAB8. Amaro Tuninetti.** Amaro Tuninetti (Dept. of Biological Sciences, Univ. of New Hampshire, Brown Univ., Box 1821, Providence, RI 02912-9067, amaro.tuninetti@unh.edu)

I am a postdoctoral researcher in Dr. Laura Kloepper's Ecological Acoustics and Behavior Lab at the University of New Hampshire. My educational background is in cognitive science and my research is on bat echolocation. This session will gather participants spanning students to late career professionals to have a conversation about work in bioacoustics. Each participant will give a brief introduction on their background and career path with ample time allotted for Q&A. Topics will include professional development, the importance of mentors, work/life balance, integrating family into careers, and more!

**10:45**

**1aAB9. Conversation with Xavier Mouy (Assistant Scientist, WHOI).** Xavier Mouy (Appl. Ocean Phys. & Eng. Dept., Woods Hole Oceanographic Inst., 3377 SW 28th Terrace, Miami, FL 33133, xavier.mouy@outlook.com)

This session will gather participants spanning students to late career professionals to have a conversation about work in bioacoustics. Each participant will give a brief introduction on their background and career path with ample time allotted for Q&A. Topics will include professional development, the importance of mentors, work/life balance, integrating family into careers, and more!

**10:50**

**1aAB10. Entrepreneurial experiences as a bioacoustician.** Benjamin N. Taft (Landmark Acoustics LLC, 1301 Cleveland Ave., Racine, WI 53405, ben.taft@landmarkacoustics.com)

This session will gather participants spanning students to late career professionals to have a conversation about work in bioacoustics. Each participant will give a brief introduction on their background and career path with ample time allotted for Q&A. Topics will include professional development, the importance of mentors, work/life balance, integrating family into careers, and more!

**10:55**

**1aAB11. Spanning the career stages–A conversation about work in bioacoustics.** Marie A. Roch (Dept. of Comput. Sci., San Diego State Univ., 5500 Campanile Dr., San Diego, CA 92182-7720, marie.roch@sdsu.edu)

This session gathers participants spanning students to late career professionals to have a conversation about work in bioacoustics. Each participant will give a brief introduction on their background and career path with ample time allotted for Q&A. Topics will include professional development, the importance of mentors, work/life balance, integrating family into careers, and more!

**11:00–12:00 Discussion**

# Session 1aAO

## Acoustical Oceanography, Underwater Acoustics, and Animal Bioacoustics: Degree in Progress: Student Led Research

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

### *Contributed Papers*

**10:00**

**1aAO1. Neural network for geoacoustic inversion of sub-bottom profiler data.** Justin Diamond (Mech. Eng., Univ. of Washington, 1013 NE 40th St., Room 173, Seattle, WA 98105, jdiam12@uw.edu), David Dall'Osto, and John Mower (Appl. Phys. Lab. at the Univ. of Washington, Seattle, WA)

Sub-bottom profilers are utilized to extract features pertaining to the sub-seafloor environment sediment stratification. Acquisition and analysis of sub-bottom profiles can provide insight into the sediment composition and acoustical properties. Typical analysis of profiles involves computationally expensive inversions, such as model-based or Bayesian techniques, which require large computational costs. Here, a neural network is developed to perform a geoacoustic inversion on simulated sub-bottom profiler data. The network is used to derive attenuation and acoustical impedance measurements corresponding to the layered media. Geoacoustic properties of the layered sediments are compared to values determined through a direct inversion of reflection coefficient, testing how well these techniques recover the ground truth values. The network, trained on simulated data, is tested on real sub bottom profiler data acquired over a well-studied area called the New England Mud Patch, roughly 80 km south of Nantucket. The simulated data-trained network is compared to a network trained on experimental data acquired by the R/V Tioga over the same test region.

**10:15**

**1aAO2. Water column observations and imaging: Revealing internal solitary waves and implications for sand wave dynamics in the Alor Strait.** I Wayan Sumardana E. Putra (Marine Sci., IPB Univ., IPB Darmaga Campus, Agatis R., Bogor, East Java 16128, Indonesia, sumardanai@apps.ipb.ac.id), Agus S. Atmadipoera, Henry Manik (Marine Sci. and Technol., IPB Univ., BogorWest Java, Indonesia), Gentio Harsono (Faculty of Sci. and Defence Technol., Republic Indonesia Defence Univ. (UNHAN), Bogor, Indonesia), Adi Purwandana (Research Center for Oceanography (RCO), National Research and Innovation Agency (BRIN), North Jakarta, Jakarta, Indonesia), I Wayan Gede A. Karang (Marine Sci., Udayana Univ., Badung, Bali, Indonesia), Budi Purwanto (Indonesian Naval Hydro-Oceanography Center (PUSHIDROSAL), North Jakarta, Jakarta, Indonesia), and Widodo S. Pranowo (Research Center For Climate and Atmosphere (RCCA), National Research and Innovation Agency (BRIN), Bandung, West Java, Indonesia)

This study presents the first comprehensive characterization of internal solitary waves (ISWs) in the Alor Strait, Indonesia, utilizing multi-modal observations from the Jala Citra Expedition 3-2023 "Flores." High-resolution water column imaging via echosounder, coupled with satellite synthetic aperture radar (SAR) imagery, revealed exceptional ISWs with amplitudes frequently exceeding 100 m, with scattering volume (SV) value ranges from −73 to −60 dB. The Korteweg-de Vries (KdV) equation was employed to analyze the subsurface wave structure, correlating with SAR-derived bright-dark patterns. In situ measurements confirmed bi-directional wave propagation towards the Flores and Sawu Seas. This research demonstrates that the Alor Strait, not just the Ombai Strait, is a significant generator of internal waves in the southern Indonesian archipelago. Time-series analysis indicated that intra-seasonal events (ENSO) significantly modulate ISW characteristics. Notably, the study uncovered a potential link between ISWs, bolus activity, and sand wave evolution, with observations suggesting ISW-induced formation of sand wave crests up to 20 m high, ~4 km long, and SV value ranges from −18.38 to −17.96 dB. These findings expand our understanding of internal wave generation in narrow Indonesian straits bordering the Indian Ocean and highlight the critical role of ISWs in shaping benthic morphodynamics, with implications for sediment transport and marine geohazards.

**10:30**

**1aAO3. Quantifying the accuracy of acoustic arrival-time predictions using high-resolution ocean circulation models.** Richard X. Touret (Ocean Sci. and Eng., Georgia Inst. of Technol., 771 Ferst Dr. NW, Office 131, Atlanta, GA 30332, rtouret@gatech.edu), Matthew McKinley (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA), Annalisa Bracco (Earth and Atmospheric Sci., Georgia Inst. of Technol., Atlanta, GA), and Karim G. Sabra (Mech. Eng., Georgia Inst. of Techn., Atlanta, GA)

Accurately modeling acoustic propagation in dynamic ocean environments exhibiting sub-mesoscale variability is challenging. Such environments require high-resolution ocean circulation model to accurately predict local sound speed variability at sub-mesoscales. This study focuses on high-resolution rendition of the De Soto canyon area, located in the Gulf of Mexico. Predicted sound speed profiles were found at times to exhibit very strong vertical gradients (e.g. with sharp "elbows") reflecting the small-scale vertical variations of ocean layer properties. If used as raw inputs for ray-tracing simulations, such sound speed profiles were found to generate inconsistent estimation of eigenray paths and spurious paths appearing intermittently caused by the inherent limitations of the ray-tracing methods. Nevertheless, we show that such artefacts can be mitigated by appropriately smoothing the input sound speed profile spatial variability especially if only low to mid-frequency (<1 kHz) propagation is predicted. To do so, the accuracy of arrival-times predictions using ray-tracing predictions with spatially smoothed sound speed profiles as a function of center frequency (<1 kHz) will be quantified against parabolic equations simulations used here as reference solution. We hypothesize that this smoothing approach for the input sound speed variations will enable to retain the numerical advantage of ray-tracing predictions while maintaining sufficient accuracy.

**10:45**

**1aAO4. Passive acoustic monitoring of a major seismic event at the Main Endeavour Hydrothermal Vent Field.** Brendan Smith (Oceanography, Dalhousie Univ., Halifax, Nova Scotia B3H 4R2, Canada, brendan.smith@dal.ca) and David R. Barclay (Oceanography, Dalhousie Univ., Halifax, Nova Scotia, Canada)

A major seismic event occurred in the region of the Main Endeavour Hydrothermal Vent Field (MEF) on March 5–6, 2024. During this 48-h period, a 4.1 magnitude event and up to 200 earthquakes per hour were recorded by Ocean Networks Canada's (ONC) cabled North-East Pacific Time-series Undersea Networked Experiments (NEPTUNE) observatory.

Low frequency waterborne acoustic signals associated with the earthquakes were detected by a four-element "box-type" hydrophone array located near multiple black smoker hydrothermal vents, with two of these elements being active during the event. Broadband signals up to 10 kHz were also measured by the hydrophones. Power spectral densities before, during, and after the seismic event show increases up to 50 dB relative to ambient levels below 100 Hz and increases up to 20 dB between 100 Hz and 10 kHz during the event. Power spectral densities between 100 Hz and 10 kHz remain elevated by approximately 5 dB more than one week following the seismic event. Power spectral density and complex coherence measurements, as well as cross-correlation with other sensors at MEF, suggest that these broadband signals may be associated with changes to the hydrothermal vent field resulting from the increased seismicity in this region.

**11:00**

**1aAO5. Quantifying three-dimensional effects on sound propagation near the New England Sea Mount Chain using ray-tracing.** Matthew McKinley (Mech. Eng., Georgia Inst. of Technol., 801 Ferst Dr. NW, Atlanta, GA 30318, mmckinley31@gatech.edu), Davis Rider (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA), Daoxun Sun, Annalisa Bracco (School of Earth and Atmospheric Sci., Georgia Inst. of Technol., Atlanta, GA), Laurent Grare, Evan Harris, Luc Lenain (Scripps Inst. of Oceanography, Univ. of California, San Diego, La Jolla, CA), and Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

This study investigates the impact of three-dimensional (3-D) bathymetry and sound speed variability on sound propagation, utilizing 3-D ray tracing compared to simple two-dimensional ray tracing (Nx2-D) within the New England seamount chain, focusing on the Atlantis II seamount area. The research has three main objectives. First, it examines the differences between 3-D and 2-D acoustic propagation modeling of bathymetric reflections. Second, it evaluates the impact of physical oceanographic effects on 3-D versus 2-D acoustic modeling, employing a high-resolution regional ocean circulation model with a horizontal resolution of 1 km and 100 terrain-following vertical layers. Third, it compares the modeled acoustic propagation to experimental data collected by a surface autonomous vehicle (wave glider) fully instruments for measuring local physical ocean variables (e.g., CTD data) and equipped with a compact tetrahedron hydrophone array (capable of profiling the water column from ∼10 to ∼150m) which recorded low and mid-frequency transmissions from moored acoustic sources deployed near Atlantis II seamount. The need for full 3-D ray-tracing modeling (instead of using simpler Nx2-D simulations) for accurately predicting shadow zones and understanding regional acoustic propagation paths in areas with rapid and diverse bathymetric changes, such as isolated seamounts, will be quantified.

**11:15**

**1aAO6. Concurrent physical and acoustical observations of the upper-ocean near Atlantis II seamount using Wave Gliders.** Davis Rider (Georgia Inst. of Technol., 771 Ferst Dr., Atlanta, GA 30332, drider3@gatech.edu), Luke Colosi (Scripps Inst. of Oceanography, La Jolla, CA), Laurent Grare, Evan Harris (Scripps Inst. of Oceanography, San Diego, CA), Karim G. Sabra (Georgia Inst. of Technol., Atlanta, GA), and Luc Lenain (Scripps Inst. of Oceanography, La Jolla, CA)

The rapid evolution of the physical properties of the upper ocean expected in regions with strong submesoscale features can lead to complex acoustic propagation effects. In spring 2023, an autonomous surface vehicle (LRI Wave Glider) was outfitted with a suite of oceanography and acoustic equipment and deployed in the Gulf Stream in a region located over the New England Seamounts, allowing an opportunity to study how the physical variability can affect the speed of sound and ambient soundscape. In August 2024, three more Wave Gliders were deployed in the same area. The vehicles, using GPS and time-synchronized equipment, were able to collect ambient acoustic data on a four-element hydrophone array, as well as local physical properties (e.g. CTD data, wind speed, and ADCP current data) that could affect the acoustic recordings during long term missions above and around the Atlantis II seamount. Two low frequency sources of opportunity were also able to be recorded over long distances, with noise level varying with vehicle position, allowing an understanding of acoustic propagation paths in this area. These long-term measurements demonstrate the unique capability of an autonomous surface vehicle for making wide-range acoustic surveys.

**11:30–12:00 Discussion**

# Session 1aBA

## Biomedical Acoustics: Transitioning Technology from Idea to Industry

Thomas Matula, Chair
*University of Washington, 1013 NE 40th St., Seattle, WA 98105*

**Chair's Introduction—10:00**

### *Invited Papers*

**10:05**

**1aBA1. The journey from bench to bedside: Converting the science of acoustics into medical technologies for the benefit of patients.** Constantin Coussios (Inst. of Biomed. Eng., Univ. of Oxford, 54 Franklin Rd., Headington, Oxford, Oxfordshire OX3 7SA, United Kingdom, constantin.coussios@eng.ox.ac.uk)

The science of acoustics has extraordinary potential to enable the development of new therapies ranging from non-invasive surgery to oncological drug delivery, neuromodulation, transdermal immunization, and immune-modulation. However, these relatively complex medical devices or combination products often challenge established clinical pathways and face considerable regulatory, usability, adoption, and reimbursement challenges. Using case studies drawn from the clinical translation of novel ultrasound technologies for drug delivery and tissue fractionation, as well as other medical technologies, some of the generic choices and considerations that should be identified and mitigated early on in the technological and commercial development process will be highlighted. For example, products involving ultrasound-responsive injectables require an in-depth understanding of their stability and stimulus-responsiveness, accompanied by the ability to confirm these characteristics post scale-up and sterilization. For all ultrasound technologies, pre-clinical testing also needs to be expanded beyond animal models to address scale and other challenges unique to humans in order to ensure preparedness for a successful clinical trial. Last but not least, the design of early feasibility trials needs to enable confirmation of the mechanism of action of the proposed new therapy and quantification of its potential added benefit relative to the most appropriate current or emerging standard-of-care.

**10:25**

**1aBA2. One lab to many: Commercialization of an automated 3D pre-clinical ultrasound imaging system.** Paul A. Dayton (The Joint Dept. of Biomed. Eng., The Univ. of North Carolina at Chapel Hill and North Carolina State Univ., CB 7575, Chapel Hill, NC 27599, padayton@email.unc.edu) and Ryan Gessner (Revvity, Inc., Durham, NC)

The Vega preclinical ultrasound system (Revvity, Inc.) is now a popular imaging tool used worldwide for automated volumetric imaging of rodents. However, like many modern scientific products, the birth of this technology started in an academic research lab. In this panel discussion, the birth and evolution of the automated small animal ultrasound system will be discussed, as well as the path this technology followed from lab, to startup company, through acquisition, and now worldwide sales. We will discuss funding, leadership, and intellectual property highlights and challenges over the technology's decade long journey.

**10:45**

**1aBA3. Vibrato Medical: Commercialization within academia without quitting your day job.** Babak Nazer (Univ. of Washington, 11103 NE 60th St., Kirkland, WA 98033, bnazer@uw.edu)

Pre-clinical studies have demonstrated that therapeutic ultrasound (TUS) increases perfusion and promotes new blood vessel growth in peripheral arterial disease (PAD) and coronary artery disease. Vibrato Medical was founded to translate these findings for PAD patients by developing a non-invasive, wearable array of TUS transducers (LimbSonic) engineered for the human calf muscles and tibial arteries. With initial support of Phase I and II NIH SBIR grants, Vibrato has grown and executed its first-in-human early feasibility study demonstrating a $180 \pm 34\%$ ($p<0.001$) increase in lower extremity perfusion (FlowMet; Medtronic) with TUS along with improvement in other limb perfusion and oxygenation endpoints, as well as patient-reported outcome measures. A sham-controlled randomized clinical trial in patients with advanced PAD is currently in progress. As part of the "Transitioning Technology from Idea to Industry" invited panel, Dr. Nazer will discuss his rubric to "commercialization within academia" (The Idea, The Unmet Need, The IP, Regulatory, Reimbursement, and Distribution) within the context of Vibrato. As an active academic physician-scientist, he will also share his perspective on how academic faculty and students can harness resources to commercialize their ideas without "quitting their day jobs" within academia.

**11:05**

**1aBA4. Five years' journey in turning sonobiopsy from concept to commercialization.** Hong Chen (Washington Univ. in St. Louis, 4511 Forest Park Ave., St Louis, MO 63108, hongchen@wustl.edu)

The standard method for diagnosing brain tumors has traditionally relied on magnetic resonance imaging (MRI). While MRI is informative, it lacks the ability to provide precise genetic information about tumors. Conversely, surgical biopsies can offer genetic insights but come with significant risks of morbidity and are not always feasible. Our team introduced sonobiopsy, a novel technique that uses ultrasound to safely and noninvasively release tumor-derived biomarkers from targeted brain locations into the bloodstream. This method allows for the molecular characterization of brain tumors through simple blood tests. In 2018, we published our proof-of-concept study in mice, followed by the first large animal study in healthy pigs in 2020. In 2022, we conducted our first study in a pig model of brain tumors. Concurrently, we developed neuronavigation-guided focused ultrasound devices for patient use. These milestones culminated in the IRB approval of our sonobiopsy clinical trial. We began recruiting our first patient for sonobiopsy in April 2022 and by September 2023, we published findings from our first-in-human prospective sonobiopsy trial in high-grade glioma patients. In the same year, we licensed our patents to Cordance Medical, which received FDA breakthrough device designation for the sonobiopsy devices. This talk will share the lessons learned throughout our journey from concept to commercialization.

**11:25**

**1aBA5. High throughput system for preparing samples for multi-omic assays.** Thomas Matula (Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, matula@uw.edu) and Karol Bomsztyk (Univ. of Washington, Seattle, WA)

Multi-omics considers the analysis of complex biological "omes" (genome, epigenome, proteome, metabolome, and others). Research in this systems biology approach to analyze associations between biological entities and to find biomarkers of disease, is growing exponentially. The basic workflow is divided into an upstream sample preparation step and a corresponding downstream assay. To prepare samples for these assays, cells must be disrupted, and/or DNA or chromatin sheared into fragments. Most often the sample preparation is done using cavitation. Current tools, such as ultrasonic horns, and even many commercial sample preparation instruments, can be highly inconsistent, leading to unreliable assays. A prototype device was built at the Univ. of Washington. Matchstick Technologies was formed to commercialize the prototype. It uses standard off-the-shelf microplates and treats 1–96 samples simultaneously using an array of transducers mounted below the microplate that focuses acoustic energy into each well of the microplate. Intense cavitation generated in each well\ results in consistent fragmentation that leads to consistent repeatable assays. [Work funded by NIH R33CA191135, R21GM111439, and R01DK103849.]

**11:45–12:00 Discussion**

TUESDAY MORNING, 19 NOVEMBER 2024 10:00 A.M. TO 11:15 A.M.

# Session 1aCA

## Computational Acoustics: Artificial Intelligence versus Human Listening to Music: Problems and Solutions

Andrea Calilhanna, Chair
*Faculty of Arts, Elder Conservatorium of Music, The Univ. of Adelaide, New South Wales, Sydney, 2126, Australia*

### *Invited Papers*

**10:00**

**1aCA1. Artificial intelligence versus human listening to music: Problems and solutions.** Andrea Calilhanna (Faculty of Arts, Elder Conservatorium of Music, The Unive. of Adelaide, Sydney, New South Wales 2126, Australia, andrea.calilhanna@adelaide.edu.au)

This session explores emerging innovations and challenges in artificial intelligence technology and practices. Panelists will present short papers on their work with AI and human interventions. A series of short talks about cutting-edge topics with insight into the research's benefits will be followed by a discussion. The audience will be encouraged to share their experience with AI and human listening. Does AI listen? How is human listening more advanced than AI? Where can AI and human listening intersect to improve outcomes? The session will provide opportunities for contributing to a broader conversation about solutions and progress in AI and human listening with topics from audio engineering, therapeutic applications, cultural heritage data, education, and AI-generated versus human music.

**10:05**

**1aCA2. Exploring the therapeutic effects of emotion equalization app during daily walking activities.** Man Hei Law (Comput. Sci. and Eng., Hong Kong Univ. of Sci. and Technol., Clear Water Bay, Kowloon, Hong Kong, mhlawaa@connect.ust.hk) and Andrew B. Horner (Comp. Sci, and Eng., Hong Kong Univ. of Sci. and Technol., Hong Kong, Hong Kong)

Mobile apps and games are discovered to be some forms of therapeutic tools for improving emotion. In this study, we explored the therapeutic effects of integrating music and digital devices into daily walking activities. We aimed to determine whether subjects experienced therapeutic effects when listening to proposed music during their walks in an uncontrolled setting. Through this study, we are able to analyze the emotion changes for the pre- and post-tests using Russell's model and identify the most effective method for generating a playlist that improves the emotions of the subjects. We recruited forty undergraduate students who used an app to listen to proposed music playlists during their walks. The results showed that the energy level and mood of the songs influenced the walking distance of the participants. Additionally, there were positive changes in the arousal and valence values of the subjects after listening to the music. The Angry quadrant was found to be the most powerful area for generating therapeutic music. These findings provide us an idea of the use of music and digital devices into daily activities for therapeutic purposes.

**10:15**

**1aCA3. Decomposing audio recordings with artificial intelligence.** Nicole Cosme-Clifford (Music, Yale Univ., 445 6th St., Brooklyn, NY 11215, nicole.cosme@yale.edu)

In the music world, artificial intelligence (AI) can now generate new sound clips, recommend music to listeners, and aid academic analysis of music recordings. Central to this advancement is a novel use of one-dimensional (1D) filter convolution, a decades-old technique broadly used in fields like sound engineering and telecommunications. Traditionally, 1D convolutional filters—small patches of signal that slide over larger signals to extract features—were manually predefined for specific purposes, such as muting frequencies above a threshold. But in audio-based AI systems, they are learned by the AI, making them both more opaque and versatile than their handcrafted counterparts. As AI music systems become more popular, the use of 1D filter convolution has expanded. Yet, the technique remains understudied in music scholarship. This talk shows how musical AI systems leverage 1D convolutional filters to decompose audio into short-term spectral features, suggesting potential new tools for sound analysis. While I will use musical examples, the implications extend beyond music. For example, AI-learned filters could enhance our ability to understand and manipulate environmental sound or improve sound processing in devices like hearing aids and smartphones. Ultimately, this talk aims to challenge existing methodologies and encourage innovative research in acoustics and related fields.

**10:25**

**1aCA4. Listeners consider AI-made music less expressive than near-identical human-made content.** Christopher W. White (Music and Dance, Univ. of Massachusetts Amherst, 151 Presidents Dr., Amherst, MA 01001, cwmwhite@umass.edu)

This presentation considers why and how listeners view musical content differently when they believe it be generated by an AI. Empirical studies (e.g., Shank *et al*. 2023) and historical commentary (e.g., Benjamin 1968) suggest that consumers find AI-made content less useful and expressive than human-made content. However, much of this research assumes noticeable differences between AI-made and human-made content, a less tenable assumption in the last few years. I describe two experiments that test listeners' reactions to technically comparable music made by humans and AI. In Experiment 1, half of participants were told a piece was composed by a human and half were told the same piece was by an AI. Participants found the piece less expressive when attributed to AI. In Experiment 2, 650 participants were given two excerpts and were asked which was created by a human versus by AI, justifying their choices using technical or expressive qualities. Half of participants was shown actual AI/human pairs, while the other half was presented with two excerpts both by AI or by a human. Analysis shows that even when there were no observable technical differences, listeners believed that the (supposedly) AI-generated content was less expressive. I argue that these sorts of distinctions will be increasingly important as AI-generated content becomes more sophisticated and that experiments like these can be used to advocate for good governance of AI companies.

**10:35**

**1aCA5. Abstract withdrawn.**

**10:45–11:15 Discussion**

# Session 1aEA

## Engineering Acoustics: Show Your Work: Lab Visits and Demos

Gary W. Elko, Chair
*mh acoustics, 25A Summit Ave., Summit, NJ 07901*

Michael R. Haberman, Chair
*Appl. Res. Lab., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758*

### *Invited Papers*

**10:00**

**1aEA1. IHTA Labs present.** Michael Vorlaender (IHTA, RWTH Aachen Univ., Kopernikusstr. 5, Aachen 52074, Germany, mvo@akustik.rwth-aachen.de), Janina Fels (IHTA, RWTH Aachen Univ., Aachen, Germany), and Lukas Aspoeck (IHTA, RWTH Aachen Univ., Aachen, NRW, Germany)

The IHTA at the RWTH Aachen University, Germany, opens its doors for a laboratory tour. It was founded in 1963 under the name ITA. The current directors of the institute are Michael Vorländer (since 1996) and Janina Fels (since 2012). In 2021, the institute was renamed the "Institute of Hearing Technology and Acoustics - IHTA" due to its expanded involvement in interdisciplinary networks, with a focus on hearing research. Since 1963, generations of students of electrical engineering and information technology as well as physicists and mechanical engineers have been trained in graduate courses. Anechoic chambers, virtual reality labs, and other labs focusing on measurement and perception research are presented.

**10:15**

**1aEA2. Sonos audio and acoustics lab tour.** Jerad Lewis (Sonos, 2 Ave. de Lafayette, Boston, MA 02111, jerad.lewis@sonos.com)

Sonos' office in downtown Boston is home to a variety of audio and acoustics labs where smart speakers, soundbars, subwoofers, headphones and installed speaker solutions are developed. This tour will show the 4-pi and 2-pi anechoic chambers, transducer development labs, and listening rooms. This is where the audio, microphone, and acoustical engineering teams develop, measure and listen to these products. In these labs, high-precision measurements are balanced with subjective listening evaluations to develop speaker products.

**10:30**

**1aEA3. Centre for immersive wave experimentation.** Henrik R. Thomsen (Inst. of Geophysics, ETH Zürich, Sonneggstrasse 5, Zurich 8092, Switzerland, henrik.thomsen@erdw.ethz.ch), Jonas Müller, Johannes Aichele, Christoph Bärlocher, Dirk-Jan van Manen, and Johan Robertsson (Inst. of Geophysics, ETH Zürich, Zurich, Switzerland)

The "Centre for Immersive Wave Experimentation" (CIWE), located at "Innovation Park Zürich," spans 200 $m^2$ and was designed to conduct ground-breaking research in the fields of acoustic and elastic wave propagation. In CIWE, we pioneer a novel approach to wave experimentation using immersive boundary conditions, seamlessly integrating the physical laboratory and a desired numerical domain. A cornerstone of CIWE is an advanced system of Field Programmable Gate Arrays (FPGAs), allowing the simultaneous acquisition and driving of 800 input and 800 output channels. This enables real-time interaction between a physical domain (e.g., the 40 $m^3$ water pool or the 2.25 $m^2$ 2-D waveguide in CIWE) and arbitrary virtual domains. Recent pioneering experiments in CIWE have demonstrated cloning and cloaking of acoustic scattering objects. Elastic wave experiments benefit from a robotized 3-D laser Doppler vibrometer providing high-resolution velocity vector measurements. The ability to conduct non-contact measurements facilitates research in the areas of metamaterials and small-scale modelling in real-world proxies like sand and clay. Here, such measurements provide ground truth, verifying and complimenting numerical simulations. Furthermore, CIWE also facilitates innovation in the area of non-destructive testing, developing ultrasound full-waveform inversion techniques for precise structural health monitoring of laminated composites.

**10:45–11:00 Discussion**

**11:00**

**1aEA4. A virtual lab tour at Bose Corporation.** Daniel Tengelsen (Bose Corporation, 100 The Mountain Rd, Framingham, MA 01701, daniel_tengelsen@bose.com) and David A. Dick (Research, Bose Corporation, Framingham, MA)

Have you ever wondered what the research and development processes look like outside of academia? In this talk, we'll provide one example by overviewing what the acoustics-related research and development work looks like at Bose Corporation. We'll provide a general walkthrough of the facilities used to create acoustic prototypes as well as the labs that an acoustical engineer would use to do, and check, their work. These unique and specialized labs are key enablers in creating products and experiences that include headphones, portable speakers, home theater systems, and automotive audio systems.

11:15

**1aEA5. Lab tour and demo of software application developed to measure spatial and acoustic linear systems responses.** Gary W. Elko (mh acoustics, 25A Summit Ave., Summit, NJ 07901, gwe@mhacoustics.com), Jens Meyer (mh acoustics, Fairfax, VT), Steven Backer (mh acoustics, Truckee, CA), Tomas Gaensler, Eric Diethorn, and Heinz Teutsch (mh acoustics, Summit, NJ)

Over many years, we have developed a software system that measures the frequency and spatial responses of microphones and loudspeakers. The software runs on a Mac or PC and uses standard multichannel sound card hardware and MATLAB for the UI. The UI calls a special processing library that records and averages the acoustic impulse responses for up to 64 channels simultaneously. The software application called SAMH, measures averaged multichannel impulse responses using a chirp signal. SAMH can also control a rotator that allows measurements to be made in with a desired angular resolution to measure the spatial responses of transducers. One underlying assumption is the chirp source signal is longer than the impulse response of the system being measured. We use a time editing feature of the measured impulse response functions to obtain the anechoic response of devices being measured. This approach is valid if the direct field impulse response decays significantly before the first reflections arrive. A video tour of our lab and the use of the SAMH software application to measure a 64-element spherical array will be shown.

11:30

**1aEA6. Design and printing micro-acousto-mechanical metamaterials.** Xiaoyu Zheng (Dept. of Mater. Sci. and Eng., Univ. of California, Berkeley, 210 Hearst Memorial Mining Building (HMMB), Berkeley, CA 94720, rayne23@berkeley.edu)

Additive manufacturing has shown the promise of freedom of designs, enabling parts customization and tailorable properties where superior structural performances can be achieved by a fraction of weight density compared to bulk material. However, it is presently difficult to combine multiple classes of materials (structural, dielectric, conducting, and ferroelectrics) and create features with hierarchical length-scales. Unlike biological systems where functions, including sensing, actuation, and control, are closely integrated, few materials have comparable system complexity. In this talk, I will present the design and printing of energy transduction devices comprised of heterogenous 3-D microarchitectures. The printed materials consist of a network of micro-unit cells enabling programmed sensitivity and directivity of external stimuli coming from different directions. I will lay out design and printing principles that enable a new class of materials capable of acoustics, tactile, and haptics generation. I will present the manufacturing and synthesis of these materials, as well as their mechanics and design methods underpinning their novel behaviors.

**11:45–12:00 Discussion**

TUESDAY MORNING, 19 NOVEMBER 2024 — 10:00 A.M. TO 11:45 A.M.

## Session 1aED

## Education in Acoustics: Teaching and Learning Acoustics with Jupyter Notebooks

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

**Chair's Introduction—10:00**

### *Invited Paper*

10:05

**1aED1. Coding in Acoustics Lessons.** Matthew Wright (ISVR, Univ. of Southampton, Tizard Building, Southampton SO17 1BJ, United Kingdom, mcmw@soton.ac.uk)

A picture is said to be worth a thousand words, and in acoustics, moving pictures can be even more valuable in helping students understand complicated and complex phenomena and relationships. My experience has been that the process of creating acoustic animations to use in my teaching has helped me to solidify my understanding, and this has led me to include coding activities in my acoustics lessons, in the hope that my students could get the same benefit. In this webinar, I will demonstrate the use of Jupyter notebooks to combine text, mathematics and coding and explore some of the possibilities, and possible pitfalls, of this approach. My examples will use the Python and Julia languages to explore how choice and use of a particular language can affect the learning process.

**11:05–11:45 Discussion**

# Session 1aMU

## Musical Acoustics: General Topics in Musical Acoustics I

Mark Rau, Cochair
*Music Res., McGill Univ., 550 Sherbrooke St. West, Suite 500, Montreal, QC H3A1B9, Canada*

Jonas Braasch, Cochair
*School of Architecture, Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180*

### *Contributed Papers*

**10:00**

**1aMU1. Visualization of three-dimensional acoustic radiation of Stradivari and Guarneri Violins using finite element analysis.** Hitoshi Asada (Inst. of Advanced Media Arts and Sci., 4-1-7 Kagano, Ogaki-shi, Gifu 503-0006, Japan, sonus_rectus2022@iamas.ac.jp)

Antonio Stradivari (1644-1737) and Guarneri del Gesù (1698-1744) are known for their remarkable violins.In this study, FE models were generated from computed tomography scan data of Willmotte Stradivari and Plowden Guarneri violins to visualize acoustic radiation into the air based on calculations of frequency response and mode shapes considering string tension preloading by numerical simulation. The results of the numerical simulation of the frequency response and mode shapes, mainly in the signature mode such as A0, CBR, B1-, and B1+, agreed with the measured data from the laser Doppler velocimeter.Furthermore, the acoustic radiation analysis results showed that the Stradivari and Guarneri results showed a difference in the tendency of sound radiation and suggested that the string tension preload may affect the directivity of the acoustic radiation. These results suggest the need for obtaining measured mode shapes from actual measurements and the importance of the influence of string tension preloading in considering geometry differences in Stradivari and Guarneri violins using FEA and acoustic numerical simulation.

**10:15**

**1aMU2. The emotional characteristics of the violin with different pitches, dynamics, and vibrato levels.** Wenyi SONG (Dept. of Comput. Sci. and Eng., The Hong Kong Univ. of Sci. and Technol., Hong Kong, Hong Kong), Anh Dung DINH (Dept. of Comput. Sci. and Eng., The Hong Kong Univ. of Sci. and Technol., Clear Water Bay, Kowloon, Hong Kong, addinh@connect.ust.hk), and Andrew B. Horner (Dept. of Comput. Sci. and Eng., The Hong Kong Univ. of Sci. and Technol., Hong Kong, Hong Kong)

Recent research has explored how the emotional characteristics of violin sounds are influenced by pitch and dynamics. This study extends previous work by considering the pitch, dynamics, and vibrato level effects on perceived valence, arousal, and 16 emotional categories. A listening test was conducted with 24 participants, who provided ratings on a 9-point Likert scale for both the valence and arousal of 32 single violin notes. Participants also indicated whether the 16 emotional labels described each sound. The results showed that pitch had the greatest impact, with higher pitches increasing arousal. Vibrato level was the second most influential factor - higher vibrato decreased valence but increased arousal. Dynamics had a relatively weaker effect, with louder dynamics increasing valence but surprisingly decreasing arousal. Interestingly, the associations between the 16 emotional categories were complex. Emotions within the same valence-arousal quadrant varied between quadrants. There was even stronger correlation between the low-valence-high-arousal and low-valence-low-arousal emotional categories than between low-valence-low-arousal and high-valence-low-arousal categories, where they might be expected. These findings provide detailed insight into how specific factors shape the perceived emotional characteristics of violin tones.

**10:30**

**1aMU3. Emotional characteristics of the erhu and violin: A comparative study of emotional intensity in musical excerpts.** Wenyi Song (Dept. of Comput. Sci. and Eng., The Hong Kong Univ. of Science and Technology, Clear Water Bay, Kowloon, Hong Kong, wsongak@cse.ust.hk) and Andrew B. Horner (Dept. of Comput. Sci. and Eng., The Hong Kong Univ. of Sci. and Technol., Hong Kong, Hong Kong)

This study investigates the emotional characteristics of the erhu and violin, focusing on the differences in emotional intensity. The experiment involved 25 participants categorizing the emotions and providing valence-arousal ratings for 14 musical excerpts. The results showed significant agreement between the two models, with the categorical emotions used to determine the emotional labels. A subsequent 9-point Likert comparison with 20 participants revealed that the instrument factor influenced emotional intensity. Overall, the violin conveyed more happiness, agitation, and calmness than the erhu, particularly for the Chinese-style excerpts. However, the sad emotional intensity was similar between the two instruments, contrary to the expectation that the erhu would convey sadder emotions. The performance factor had minimal impact on emotional intensity within the same instrument group, suggesting the inherent qualities of the erhu and violin were the primary drivers of the differences. Listeners more familiar with the instruments were likely to rate the violin higher in valence and arousal than the erhu. These findings contribute to understanding the expressive capabilities of the two instruments and highlight the instrument-specific emotional characteristics that shape the perceived emotional characteristics of music.

**10:45**

**1aMU4. Effects of goodness-of-fit on auditory segregation in musical backgrounds.** Jiwon Lee (None, Gwangjin-Gu Guui-dong, Seoul 05044, Korea (the Republic of), jwlee6153@gmail.com) and Eunmi Oh (Psychology, Yonsei Univ., Seoul, Korea (the Republic of))

This study examined the impact of "goodness-of-fit," a term by Krumhansl and Kessler that measures how well a tone fits within musical backgrounds on auditory segregation. The stimuli consisted of 40 consecutive sequences of 60 ms chords, totaling 2.4 sec. The figure tone was either C, D, E, F, G (high goodness-of-fit in C major), or their sharp equivalents (low goodness-of-fit in C major), and consistently repeated on each trial. The backgrounds were either consonant chords in C major or dissonant chords of random tones with each chord containing 8 tones ranging from 277 to 3729 Hz. Participants detected the figure from the background, using a staircase procedure to achieve over 70% accuracy, which set the figure volume for the main experiment. Results showed a significant accuracy decrease (up to 20%) only for the no-sharp figure in the consonant background. In contrast, the sharp figure's accuracy remained unaffected regardless of the background. These results indicate that figures with high goodness-of-fit

were harder to distinguish in consonant backgrounds, whereas figures with low goodness-of-fit were easier to detect in consonant and dissonant backgrounds. This implies that listeners may utilize lawful relationships among musical chords for auditory segregation.

**11:00**

**1aMU5. Acoustic profiling of Guqin music: Exploring the unique sound quality associated with an ancient Chinese musical instrument.** Jiarui Tian (Health Sci., Duquesne Univ., 1420 Centre Ave., Apt. 1213, Pittsburgh, PA 15219, tianj@duq.edu) and Manwa L. Ng (Speech and Hearing Sci., Univ. of Hong Kong, Pokfulam, Hong Kong)

Similar to speaking, production music using an instrument is also governed by the Source-Filter Theory—all sounds are produced as a product of source and filter. While Western musical instruments have been studied extensively, research on ancient Chinese musical instruments including guqin is scarce. With a history of over 3000 years, the quality of guqin music is characterized by a unique resonant timbre and subtle tonal variations. Its intricate playing techniques such as plucking and sliding also allow generation of sounds of distinct properties, contributing to its elegance, tranquility, and sublimate quality. To objectively describe and quantify guqin's timbre, the present study will attempt to objectively describe the timbre of guqin's music by analyzing its sounds using a number of time- and frequency-domain parameters including frequency, amplitude, perturbation, spectral contents, etc. It aims to highlight the guqin's distinctive acoustic characteristics in order to deepen our understanding of sound produced with this ancient Chinese musical instrument.

**11:15**

**1aMU6. Understanding melodic transitions: A neuro-acoustical study with Indian Classical Music.** Medha Basu (Jadavpur Univ., 188, Raja Subodh Chandra Mallick Rd., Sir C.V. Raman Centre for Physics and Music, Kolkata, West Bengal 700032, India, medhabasu1996@gmail.com), Kumardeb Banerjee, and Dipak Ghosh (Jadavpur University, Kolkata, West Bengal, India)

Different kinds of melodic movements and note-to-note transitions form one of the most important features of music. In this work, we have tried to analyze some of these transitions, within the domain of Instrumental Indian Classical Music. Two most widely used transitions in ICM renditions have been chosen for this pilot study- i). slow, gliding transitions, ii). fast, direct-note transitions Three pairs of melodic portions have been prepared from recorded Sitar renditions of Indian maestros, such that within each pair, one clips was predominantly characterized by slow, gliding inter-note transitions and the other by fast, direct ones. Time of transitions, pitch profiles and certain acoustic features like spectral centroid, entropy, kurtosis, skewness etc. were first compared for these two clip-categories. Next, Electroencephalogram (EEG) experiment was performed on 5 participants, using these two kinds of clips as input signals. Extracted neural signals were analyzed with help of robust non-linear scaling technique of 'Multifractal Detrended Fluctuation Analysis'. The neural responses corresponding to these two contrasting transitions were then compared with parameters like EEG-complexity and EEG-entropy. This pilot-study is a novel attempt to understand the acoustic attributes and characteristics of two contrasting melodic transitions of ICM, as well as their corresponding neural correlates.

**11:30**

**1aMU7. An automated mashup system with drum-bass stem swapping.** Xinyang Wu (Hong Kong Univ. of Sci. and Technol., HKUST, GGT, Sai Kung, Hong Kong, xwuch@connect.ust.hk) and Andrew B. Horner (Hong Kong Univ. of Sci. and Technol., Hong Kong, Hong Kong)

Music mashups have emerged as a compelling form of art, merging elements from various songs to create new auditory experiences. Our research automated the mashup generation process and analyzed listener preferences to understand what contributes to a mashup's popularity. Leveraging modern source separation technologies, we enabled the automatic creation of mashups from any two songs. Using three mixing approaches—drum swapping, bass swapping, and combined drum and bass swapping—we generated 270 mashups for evaluation. These approaches were tested with advanced beat tracking and harmonic alignment techniques to ensure musical coherence. Systematic listening tests were conducted with 50 survey participants to gauge preferences. The analysis revealed low correlations in preferences across the different approaches, indicating that each created a unique listening experience. Coupled with a comprehensive analysis of rhythm and harmony, our research sheds light on the qualities that make a mashup appealing and provides guidelines for developing more effective automated tools to assist music enthusiasts in creating popular and musically satisfying mashups.

**11:45**

**1aMU8. Effect of years of voice training on chest and head register tongue shape variability.** Jiu Song (Dept. of Linguistics, Univ. of British Columbia, Totem Field Studios, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, jiusongjd@gmail.com), Jaida Su, Jahurul Islam, and Bryan Gick (Dept. of Linguistics, Univ. of British Columbia, Vancouver, BC, Canada)

The transition between head and chest registers in operatic singing has been linked to adjustments in the larynx [Henrich, 2006, *LPV* 31], vocal tract length [Tokuda *et al*., 2010, *JASA* 127], and resonance frequencies [Echternach *et al*., 2011, *JASA* 129]. Research on supralaryngeal articulator differences, specifically midsagittal tongue shape differences, during this transition is limited. Our previous case study showed a higher tongue dorsum in head voice for low and mid vowels compared to chest voice [Bengtson *et al*., 2023, *CAA* 51]. The current study with ten participants (8 female, 1 non-binary, 1 male, aged 19–23) further explores this and the effect of the years of vocal training. Participants were recorded performing a chromatic scale through their register transition, followed by a whole-tone scale in each register, with both tasks repeated twice on each of seven vowels (/a, e, o, ɚ, i, u, y/). The hypothesis is that vowel-dependent tongue adjustments would be observed, with more experienced singers displaying smaller differences between registers. Tongue shapes were traced using DeepEdge, and target frames were extracted [Chen *et al*., 2020, *ISSP* 2020]. Preliminary analysis indicates that the tongue dorsum is lower for chest voice and for participants with more years of voice training.

## Session 1aNS

# Noise: Advanced Noise Control Design and its Benefits to Humans and Society

Tracy Y. Choy, Cochair
*Hong Kong Polytechnic Univ., Kowloon, Hong Kong*

Ning Xiang, Cochair
*School of Architecture, Rensselaer Polytechnic Inst., 110 Eighth St., Troy, NY 12180*

### *Invited Papers*

**10:00**

**1aNS1. Study on noise insulation using double layered acoustic gratings.** Liangfen Du (The Hong Kong Polytechnic Univ., China, Hong Kong, liangfen.du@polyu.edu.hk) and Zheng Fan (Nanyang Technological Univ., Singapore, Singapore)

Noise reduction technologies that enable natural ventilation have garnered significant attention, especially in densely populated cities like Hong Kong and Singapore. These technologies not only provide aural and thermal comfort but also serve as alternatives to air conditioning during cooling seasons, thereby reducing energy consumption in green buildings. Common passive solutions for noise reduction and natural ventilation include plenum windows, coiling-up-space structures, and Helmholtz resonator (HR)-based structures. However, achieving effective noise isolation, particularly at low frequencies, often involves trade-offs, such as larger spacing between glass panels in plenum windows or increased volumes in coiling-up-space and HR-based structures, which can make them bulky and limit their practical applications. To overcome these challenges, we propose an ultrathin solution: the double-layered acoustic grating (DLAG). This innovative structure features two layers of rigid panels with perforated slits. Unlike noise insulation technologies that rely on energy conversion or impedance mismatch, the DLAG redirects acoustic energy away from sensitive areas, creating a quiet zone where needed. Additionally, the slit configuration supports natural ventilation. This paper will examine the noise reduction performance of the DLAG, specifically investigating its potential for low-frequency noise isolation.

**10:10**

**1aNS2. Environmental noise reduction by sonic crystals barrier.** Xiaoru Qiao (Hong Kong Polytechnic Univ., 11 Yuk Choi Rd., Hong Kong, Hong Kong, xrfirst.qiao@connect.polyu.hk) and Yat Sze Choy (Hong Kong Polytechnic Univ., Kowloon, Hong Kong)

Noise barrier is widely applied as an effective approach to improve life quality of people by mitigating environmental noise that comes with booming transportation and urbanization. Compared with conventional noise barriers, sonic-crystal-based barriers stands out because of better noise mitigation performance at lower frequency while avoiding excessive airflow and visual obstruction. However, large thickness owing to multi-layer structure remains a challenge for practical application of sonic-crystal-based barriers. According to the generalized Snell's law, a compact barrier composed by the novel shadowed-C-shaped (SCS) unit cells is designed with a much smaller lattice constant. Furthermore, the SCS unit cell is optimized to achieve broader band gaps in the frequency range below 5kH based on the Bloch's theorem, thus improving the noise reduction performance. Simulation results show that the insertion loss of the optimal barrier is over 10 dB in broader band, especially around 800 and 1210 Hz. It is verified that the experimental insertion loss is basically consistent with the numerical results. The sonic-crystal-based barrier for traffic noise control in a compact thickness also shows a promising potential to be applied to various scenarios of noise control.

**10:20**

**1aNS3. On broadband non-synchronous measurements beamforming: A Laplacian-enhanced low-rank tensor completion approach.** Long Chen (Northwestern Polytechnical Univ., 127 West Youyi Rd., Beilin Dist., Xi'an 710072, China, chenmf767@foxmail.com), Jiaxin Hu (Northwestern Polytechnical Univ., Xi'an, China), Xiaoang Liu (Hebei Univ. of Technol., Tianjin, China), Weize Sun, and Lei Huang (Shenzhen Univ., Shenzhen, China)

Due to the physical limitations of microphone arrays, there are constraints on both the frequency range and the scale of the scanning area. While a microphone array can potentially cover a wider frequency range and provide a more comprehensive view, this requires a larger and denser array configuration, achievable through non-synchronous measurements beamforming. However, localizing multiple broadband sources remains challenging without prior information about the target signal's frequency. To address this, we propose a novel approach: a non-synchronous measurements method for broadband multiple-sound-source localization using Laplacian-enhanced low-rank tensor completion. Our method dissects the tensor data structure of broadband signals and employs an alternating direction method rooted in multiplier optimization. Simulations and experimental studies show that our Laplacian-enhanced algorithm accurately captures five distinct speech signal sources, offering a superior global view compared to other convex surrogates. Moreover, our method excels in low signal-to-noise ratio conditions, significantly reducing the side-lobe level in the sound map and outperforming previous optimization techniques. This advancement is promising for applications requiring precise localization of multiple broadband sound sources.

## *Contributed Papers*

**10:30**

**1aNS4. Measurement and mitigation for impulse noise from Pickleball Courts.** Braxton Boren (Audio Technol., American Univ., 4400 Massachusetts Ave. NW, Washington, D.C. 20016, boren@american.edu)

Pickleball is one of the fastest-growing sports in America, and many municipalities have converted tennis courts for pickleball in the past decade. However, the hard plastic ball and hard racquet yields a high-frequency impulsive noise with energy centered near 1200 Hz which is quite different from the low-frequency modal resonance of a tennis racquet. As a result, pickleball noise tends to result in much greater community annoyance than tennis noise from a similar distance since the human ear is more sensitive in that frequency range. However, the relatively short wavelength of pickleball noise also means that it is more easily obstructed by barriers and absorbed by the atmosphere. This study examines some case studies of pickleball noise mitigation and some general principles that are raised in the diffracted versus obstructed path. In addition, the question of the most appropriate metrics for pickleball sound are considered given the brief duration of pickleball hits.

**10:40**

**1aNS5. Study on the prevention and treatment of noise induced hearing loss in mice with saturated hydrogen physiological saline.** JinGe Tang (North Sichuan Medical College, Fuxing Rd., Haidian Dist., Beijing, Beijing 100039, China, 2353597062@qq.com)

Objective: This experiment explored the preventive effect of saturated hydrogen physiological saline by establishing a mouse model of noise induced hearing loss. Methods: 24 C57/6J male mice (8W) with SPF level and normal ABR threshold were randomly divided into three groups: normal control group, noise group, and saturated hydrogen physiological saline group. The saturated hydrogen physiological saline group received daily intraperitoneal injections of saturated hydrogen physiological saline (1 ml/100 g) three days before receiving steady-state noise exposure (110 dB SPL, 4h). Perform ABR audiometry before and 1, 3, and 7 days after noise exposure. After completion, the animals were euthanized and samples were collected for observation under scanning electron microscopy. Results: The ABR results showed that the hearing threshold of mice in the saturated hydrogen physiological saline group was lower than that of the noise group. Scanning electron microscopy observation showed that the cilia morphology of cochlear hair cells in the saturated hydrogen physiological saline group was better than that in the noise group. Conclusion: Daily intraperitoneal injection of saturated hydrogen physiological saline three days before noise exposure has a certain protective effect on hearing loss caused by steady-state noise, laying the foundation for further exploration of its specific mechanism at the pathological molecular level.

**10:50**

**1aNS6. A review of port noise management strategies to improve sonic cohabitation.** Negar Imani (School of Information Studies, McGill Univ., 3464, rue Hutchison, Montreal, QC H2X 2G7, Canada, negar.imani@mail.mcgill.ca), Cynthia Tarlao (School of Information Studies, McGill Univ., Montreal, QC, Canada), and Catherine Guastavino (CIRMMT & School of Information Studies, McGill Univ., Montreal, QC, Canada)

Sonic cohabitation between port areas and the communities living nearby is a major concern for many cities worldwide. To manage port noise, city governments, port authorities, and even academic researchers have formulated various strategies at different scales. Most of these strategies have been published in a variety of formats (e.g., academic articles, acoustic reports, and governance guidelines), and to our knowledge, have never been compiled and reviewed. The present paper is a review of available documentation and literature to identify trends and practices in port noise management. A Web of Science search and additional reference list checks yielded a total of 69 documents, out of which we identified 41 documents for full review. We found that three main types of strategies are suggested or implemented, namely (1) technical solutions (e.g., noise barriers), (2) design and planning considerations (e.g., traffic calming, buffer zones), and (3) processes and operations (e.g., communication with residents, hours of operation). Few of the suggested or implemented strategies to mitigate noise port are evaluated after implementation. Nearly all of the evaluations rely on acoustic measurements and overlook resident experiences. We discuss ways to better involve residents to improve sonic cohabitation.

**11:00**

**1aNS7. Managing the noise environment in the context of densification: Issues, expertise, methods, and limits.** Lucas Germain (École supérieure d'aménagement du territoire et développement régional, Université Laval, 2325 allée des Bibliothèques, bureau FAS-1616, Quebec, Quebec G1V 0A6, Canada, lucas.germain.1@ulaval.ca), Philippe Apparicio (Dépt. de géomatique appliqué, Univ. de Sherbrooke, Sherbrooke, QC, Canada), Frédéric Hubert (Dépt. des sciences géomatiques, Univ. Laval, Quebec City, QC, Canada), Tony Leroux (École d'Orthophonie et d'Audiologie, Univ. de Montréal, Montréal, QC, Canada), Jean-Philippe Migneron (Ecole d'architecture, Universite Laval, Quebec City, QC, Canada), and Johanne Brochu (École supérieure d'aménagement du territoire et développement régional, Université Laval, Québec, QC, Canada)

Urban densification is often proposed as a solution to ecological problems. Various densification models are studied in the literature, but noise aspects are often neglected. Based on a case study in Quebec City, environmental noise assessment methods (manual and automatic surveys, simulation, etc.) have been explored to identify the skills, expertise, and data needed to integrate the noise dimension into densification projects, and how to choose a model suited to the urban planning context. The availability and quality of data will also be assessed to identify any gaps. Using a combination of morphological analysis and sound simulations, the results will provide a better understanding of the influence of sound quality on densification choices. A discussion about the implications for sustainable urban planning and residents' quality of life, as well as the criteria to be considered in densification strategies to integrate the noise environment effectively. This project will provide answers on the impact of densification models on the noise environment and urban quality of life and allow the development of a framework for improved urban practices.

## *Invited Paper*

**11:10**

**1aNS8. A customized approach to hybrid active noise cancellation system design to address auditory hyperreactivity for children with autism spectrum disorder.** Tak Chun Kwong (Dept. of Mech. Eng., The Hong Kong Polytechnic Univ., Hung Hom, Kowloon, Hong Kong, tak-chun-daniel.kwong@connect.polyu.hk) and Yat Sze Choy (The Hong Kong Polytechnic Univ., Kowloon, Hong Kong)

Individuals with autism spectrum disorder often exhibit heightened sensitivity to certain sounds, which is known as auditory hyperreactivity. This can manifest in aversive behaviors when these children encounter particular noises in their daily lives. To address this challenge, noise-cancelling headphones are frequently employed. However, traditional headphone designs primarily focus on simply reducing overall noise levels, without accounting for the heterogeneous and nuanced aural perceptions exhibited by autistic children. To develop a more tailored noise-control solution, a series of behavioral and electroencephalography tests were conducted to analyze the auditory responses. Participants were exposed to sounds of various frequencies and amplitudes, and their subjective perceptions were analyzed and processed to establish appropriate noise-attenuation targets, based on a function that considered both the average aural perception ratings and the corresponding noise levels. Building upon these results, a hybrid active noise cancellation system that caters to the individual auditory needs of autistic children was designed and tested. When using the customized headsets, the children's aural perception responses showed marked improvement.

**11:20–12:00 Discussion**

TUESDAY MORNING, 19 NOVEMBER 2024 — 10:00 A.M. TO 12:00 NOON

# Session 1aPA

## Physical Acoustics: Student Video Competition

Joseph A. Turner, Chair
*Mechanical and Materials Eng., Univ. of Nebraska-Lincoln, Lincoln, NE 68588*

## *Contributed Papers*

**1aPA1. Measurements of acoustic properties of synthetic 3-D printed grass blades to predict sound attenuation of tall grass-covered ground.** Christian Di Nicolantonio (Mechanical Eng., The Catholic Univ. of America, 620 Michigan Ave. NE, Washington, D.C. 20064, dinicolantonio@cua.edu), Emily Bowman (Aerodyne Industries LLC, Washington, D.C.), Christopher Crognale, Ariel Wise, John Soria (Mechanical Eng., The Catholic Univ. of America, Washington, D.C.), Teresa J. Ryan (Engineering, East Carolina University, Greenville, NC), Joseph Vignola (Mechanical Eng., The Catholic Univ. of America, Washington, D.C.), Jason Davison (Civil and Environmental Eng., The Catholic Univ. of America, Washington, D.C.), and Diego Turo (Mechanical Eng., The Catholic Univ. of America, Washington, D.C.)

Tall grass covering the ground can greatly affect acoustic transmission loss over long propagation ranges. Understanding and predicting the sound attenuation introduced by this type of vegetation is of both naval and aerospace relevance. Tall grass fields are common near both shorelines and air- or spaceports. In this study, the effect of the grass blades is assumed to be decoupled from the absorption of the ground. A numerical simulation replaces the space occupied by the blades with an equivalent fluid volume that has an attenuation coefficient equal to that of synthetic grass blades. Grass blades are modelled with rectangular parallel stripes. A 5 mm thick disc frames the blades to form a sample that fits in a cylindrical impedance tube. Several samples are designed, and 3-D printed to have blades oriented at 0 or 45 degrees with respect to the propagation of the acoustic wave. The samples are stackable modular components that can be combined to create multilayered samples. Complex characteristic impedances and wavenumbers of different multilayered samples are measured through a series of impedance tube experiments. The measured wavenumber is then used as input in a parabolic equation solver to quantify the effect of a tall grassy field on acoustic transmission loss over a propagation range of 300 m.

**1aPA2. Enhancing Rayleigh-regime liquid jet breakup with MHz-order acoustic waves.** Kha Nguyen (Univ. of California San Diego, 9500 Gilman Dr., La Jolla, CA 92093, khn016@ucsd.edu) and James Friend (Mechanical and Aerospace Eng., Univ. of California San Diego, La Jolla, CA)

We present a new device that effectively reduces the breakup of cylindrical Rayleigh-regime liquid jets in quiescent air. The device is made of lithium niobate, a piezoeletric material, to convert input electrical power into oscillation. A 10x10 mm crystal with a through hole of 1 mm in diameter placed at the outlet of the liquid's nozzle, the device creates 7 MHz acoustic waves that disrupt the stability of the jet. We show its effectiveness over a range of different input powers and Weber and Ohnesorge numbers. Furthermore, we show modelling results in COMSOL and equations that describe this new instability mechanism. This opens new doors for droplet-on-demand applications.

**1aPA3. Cloud coverage estimation for improved atmospheric acoustic predictions.** Sophie Arruza (Engineering, East Carolina Univ., 1004 Haddon Hall Dr., Apex, NC 27502, arruzas22@students.ecu.edu), Jeff Foeller (Engineering, East Carolina Univ., Greenville, NC), Joseph Vignola (Mechanical Eng., The Catholic Univ. of America, Washington, D.C.), and Teresa J. Ryan (Engineering, East Carolina Univ., Greenville, NC)

Weather is an important factor in atmospheric acoustic transmission loss over moderate distances, 1–10 km. One critical issue is cloud cover. This work focuses on measuring the cloud coverage at a costal location that also has permanent temperature logging masts and a wind speed profiling LIDAR which characterizes the coastal meteorological conditions. Improving cloud coverage estimates improves surface heat flux estimates, which in turn supports acoustic propagation modeling. The cloud cover is determined from photos which are analyzed in MATLAB using RGB filtering. Photographic data collected on July 6, 15, and 25 of 2023 will be presented. These photos were taken at intervals of 2 min, over the course of a day, under changing weather conditions. The simple RGB filter was successful for cases without rain throughout most of the day with the exceptions primarily occurring near sunrise and sunset. [Work supported by the Robert W. Young Award for Undergraduate Student Research in Acoustics.]

**1aPA4. Empirical observations of significant wave height and atmospheric acoustic transmission loss in a near-shore environment.** Heath Faircloth (Dept. of Eng., East Carolina Univ., 1000 East Tenth St., Greenville, NC 27858, fairclothh22@students.ecu.edu), Matthew Stengrim (Dept. of Eng., East Carolina Univ., Greenville, NC), Jeff Foeller (Dept. of Eng., East Carolina Univ., Greenville, NC), Diego Turo, Joseph Vignola (Mechanical Eng., The Catholic Univ. of America, Washington, D.C.), and Teresa J. Ryan (Dept. of Eng., East Carolina Univ., Greenville, NC)

Sound is influenced by meteorological conditions as it travels, which can have profound effect on transmission loss over long ranges. Conditions, such as wind speed, wind direction, and temperature, are primary atmospheric variables affecting transmission loss. At longer ranges, the surface roughness of the water is also an important factor to consider. Significant wave height is a typical metric to characterize the water surface roughness. A June 2024 field measurement campaign was completed in the Pamlico Sound adjacent to the East Carolina University Outer Banks campus. Measurement of meteorological conditions and concurrent acoustic transmission losses were performed over water only at ranges from 250 m to 2 km. A buoy that records three-dimensional wave motion was incorporated into the measurement process and deployed between the acoustic source and receiver array. The record of wave dynamics allows for the calculation of significant wave height. The approach to synchronizing wave height measurements to the acoustic measurements is discussed. Results are presented to show how water surface roughness relates to measured transmission loss.

**1aPA5. Testing a spiral beamforming array for airborne acoustics.** Matthew Stengrim (Dept. of Eng., East Carolina Univ., 1000 East Fifth St., Greenville, NC 27858, stengrimm19@students.ecu.edu), Heath Faircloth, Brielle Wagner, Teresa J. Ryan (Dept. of Eng., East Carolina Univ., Greenville, NC), Joseph Vignola, Diego Turo (Mechanical Eng., The Catholic Univ. of America, Washington, D.C.), and Seth Hubbard (NSWC Carderock, W. Bethesda, MD)

Airborne acoustic signature characterization of watercraft is one of the interests of Naval Surface Warfare Center Carderock Division (NSWCCD). A twenty-element planar spiral array was recently designed for this purpose. In June 2024, the system was used in a collaborative measurement campaign with East Carolina University in Wanchese, NC. The array was fixed to an offshore platform and recorded pure tone bursts of 250, 500, 1000, and 2000 Hz from a Long Range Acoustic Device (LRAD) positioned at ranges from 250 m up to 2 km. Within a 10 min period, estimations of transmission loss and SNR in upward refraction cases varied by 10–20 dB for each frequency. Using a MATLAB-based time delay beamformer, SNRs typically increased by up to 5–10 dB. Broadside and off-angle results are discussed.

**1aPA6. Electric-acoustic force balance for sorting microparticles on a chip.** Yasaman Kazemipour (Colorado School of Mines/NIST, 1679 Ridge View Ct, Erie, CO 80516, ykazemipour@mines.edu), Bryan Bosworth, Nicholas R Jungwirth (National Institute of Standards and Technology, Boulder, CO), Lucas L Enright (Colorado School of Mines/NIST, Boulder, CO), Tomasz Karpisz (University of Colorado Boulder, Boulder, CO), Florian M Bergmann (National Institute of Standards and Technology, Boulder, CO), James C Booth (National Institute of Standards and Technology, Boulder, CO), Annalise Maughan (Colorado School of Mines, Golden, CO), Nathan Orloff (National Institute of Standards and Technology, Boulder, CO), and Angela C. Stelson (National Institute of Standards and Technology, Boulder, CO)

Electronic waste is the fastest growing solid waste stream in the world, accounting for 54 million tons of waste globally. Such waste contains valuable chemical elements, such as gold, palladium, copper, silver, tin, and nickel that could be recycled back into the economy. Here, we present a proof-of-concept device to test a new electric-acoustic force balance to separate particles based on their electrical and mechanical properties. The device applies an acoustic force and electric force simultaneously causing microparticles to move to a position where the forces are equal. We used interdigitated transducers to generate a standing acoustic wave in the channel, collimating the particles at the pressure nodes. We then applied an electric field by applying a voltage between two electrodes patterned on either side of the microfluidic channel. This electric force changed the position where the microparticles collimated. The vision is to test microparticles from new waste recovery processes and show how they separate in our force balance to inform large scale separation technology.

**1aPA7. Ultrasonic evaluation of electrically degraded multilayer ceramic capacitors.** Haley N. Jones (Materials Science and Engineering, Penn State Univ., N-225 Millenium Science Complex, State College, PA 16802, hnj5051@psu.edu), Andrea P. Arguelles (Engineering Science and Mechanics, Penn State Univ., University Park, PA), and Susan Trolier-McKinstry (Materials Science and Engineering, Penn State Univ., University Park, PA)

Multilayer ceramic capacitors (MLCCs) are a vital circuitry component, responsible for 30% of all circuit failures. A common degradation mechanism in MLCCs is insulation resistance degradation caused by electromigration of oxygen vacancies under DC bias which limits part lifetime. This work utilizes high frequency ultrasound (100 MHz) as a nondestructive characterization tool to spatially identify regions of degradation caused by oxygen vacancy migration. Ultrasonic attenuation, which results from absorption and changes in acoustic impedance that lead to scattering, was explored. Thermally stimulated depolarization current (TSDC) was utilized to degrade parts and electronically evaluate the presence of oxygen vacancies. A set of commercial 1812 BME X7R MLCCs (52 dielectric layers, 100 V rated, 0.560 $\mu$F) were characterized using a custom ultrasonic immersion setup in both a pristine state and after TSDC (130 °C, 140 °C, and 150 °C for 8 h and 10 deg/min heating rate). Each TSDC condition indicated oxygen vacancy migration across grain boundaries. Average ultrasonic attenuation increased (1%-22%) and changes in the spatial attenuation response were observed after TSDC. The most extreme poling condition resulted in the largest variation in the spatial attenuation response which could indicate sensitivity to structural or microstructural damage caused by electromigration of oxygen vacancies.

**1aPA8. Ultrasonic phased array testing to quantify material texture for metal additive manufacturing.** Geoffrey R. Soneson (Mechanical and Materials Engineering, Univ. of Nebraska - Lincoln, 2320 Y St., Apt. 1, Lincoln, NE 68503, gsoneson2@unl.edu), Nathanial Matz, and Joseph A. Turner (Mechanical and Materials Engineering, Univ. of Nebraska-Lincoln, Lincoln, NE)

Metal additive manufacturing (AM) processes result in complex microstructures, porosity, material texture, and residual stresses. These properties may vary throughout AM parts. Ultrasonic nondestructive evaluation (NDE) is essential for understanding the processes leading to these variations. However, the use of single-element transducers to scan AM parts with complex

geometries can be challenging. Such measurements may require multiple scans to level a surface or excite various wave types within a part. On the contrary, phased array ultrasonic transducers (PAUTs) can be much more efficient for these studies. In this presentation, custom AM sample designs are discussed which allow multiple ultrasonic wave speeds to be measured using a PAUT. The manufactured samples were assessed with the sectorial scan feature of the PAUT to scan multiple positions and directions quickly without moving the probe. Data from these scans were then used to estimate material wave speeds, material texture, and residual stress along multiple directions within the samples. The results from these scans were also compared with experiments that used single-element ultrasonic transducers. This presentation will describe the sample designs, the experiments, and the analysis in order to highlight the usefulness of PAUTs for scanning AM parts.

TUESDAY MORNING, 19 NOVEMBER 2024 — 10:40 A.M. TO 12:00 NOON

## Session 1aPP

## Psychological and Physiological Acoustics: My Favorite Graph in Auditory Science

Sunil Puria, Cochair
*Mass Eye and Ear, Harvard Medical School, 243 Charles St., Boston, MA 02111*

Daniel Tollin, Cochair
*Univ. of Colorado, School of Medicine, 12800 E 19th Ave., Aurora, CO 80045*

**Chair's Introduction—10:40**

### *Invited Paper*

**10:45**

**1aPP1. Modeling binaural hearing: Stimulus into ear j (j = 1, 2).** Barbara Shinn-Cunningham (Carnegie Mellon Univ., 5000 Forbes Ave., Psychology, BH 254G, Pittsburgh, PA 15213, bgsc@andrew.cmu.edu)

The year before the presenter was born, Nat Durlach published "Equalization and Cancellation Theory of Binaural Masking-Level Differences." The paper's elegant mathematics yielded quantitative predictions for how binaural differences can influence the ability to detect a pure tone embedded in noise. The resulting formulae accounted well for existing psychophysical data coming from a number of different binaural hearing studies (conducted by various researchers). This seminal paper still exemplifies best practices when developing models of perceptual outcomes and neural processing: clearly spelling out assumptions to produce testable, verifiable, and quantitative predictions.

### *Contributed Paper*

**10:55**

**1aPP2. My favorite graph: Individual differences in cochlear implant outcomes.** Aaron C. Moberly (Otolaryngology, Vanderbilt Univ. Medical Ctr., 1215 21st Ave. South, Suite 7209, Nashville, TN 37232, aaron.c.moberly@vumc.org), Terrin N. Tamati (Otolaryngology, Vanderbilt Univ. Medical Ctr., Nashville, TN), and David Pisoni (Psychological and Brain Sci., Indiana Univ., Bloomington, IN)

The Childhood Development after Cochlear Implantation (CDaCI) study was a pivotal study of language development in children receiving cochlear implants (CIs) in the early 2000s. The main findings were that children receiving CIs developed better receptive and expressive language skills than would be predicted if they had not undergone implantation, presented in Niparko *et al*. (2010). Figure 1 from that publication showed the trajectories of language improvements after cochlear implantation, represented as spaghetti plots for individual CI children, relative to a group of normal-hearing (NH) peers. Figure 1 demonstrated enormous individual differences in receptive and expressive language scores for the CI children, spanning from zero (the minimal performance possible) to the maximal performance demonstrated by NH peers. Most research has focused on the significant group effect findings (CI versus NH), while neglecting a deep dive into the factors underlying the individual differences across participants. Yet, Figure 1 from Niparko *et al*. (2010) remains a valuable figure that has motivated work from our research groups to explain and predict CI outcomes, the longitudinal trajectories of outcomes following implantation, and sources of variability, which will be discussed at a high level and related back to Figure 1 by Niparko *et al*. (2010).

## *Invited Paper*

**11:05**

**1aPP3. The binaural interaction component (BIC) of the auditory brainstem response (ABR) reveals brainstem origin of listening difficulties in children.** Daniel Tollin (Dept. of Physiol. & Biophysics, Univ. of Colorado, School of Med., 12800 E 19th Ave., Aurora, CO 80045, Daniel.Tollin@CUAnschutz.edu)

Children experiencing conductive hearing loss (CHL) often display binaural hearing impairments that persist after CHL resolution. The neurophysiological correlates of the deficits and where problems begin to emerge are unknown. The long-time dogma was that maladaptive changes occur in cortex because while cortex remains plastic the brainstem circuits were not thought amenable to plasticity. Figures 5 and 6 of Gunnarson A. D. and Finitzo T. (1991, “Conductive hearing loss during infancy: Effects on later auditory brain stem electrophysiology,” J Speech, Lang. Hear. Res. 34, 1207–1215) demonstrated that the brainstem could be a plausible site by revealing that the amplitude of the BIC was eliminated in children with CHL histories. The BIC is computed by subtracting the sum of left and right ear monaural ABRs from binaural-evoked ABRs. The amplitude of the most prominent BIC peak, DN1, scales with interaural time differences and correlates with performance on binaural tasks. The latency of DN1 corresponds to the binaural nuclei of the auditory brainstem that initially extract the binaural cues to sound location, the lateral and medial superior olive. The study established that listening difficulties due to CHL emerge in the brainstem, a result with implications for how to diagnose and develop treatments for listening difficulties.

## *Contributed Papers*

**11:15**

**1aPP4. A single reflective surface reveals contextual cue-weighting in spatial hearing.** G. Christopher Stecker (Ctr. for Hearing Res., Boys Town National Research Hospital, 555 N 30th St., Omaha, NE 68131, cstecker@spatialhearing.org)

Prominent among the factors supporting accurate sound localizatoin in reverberant scenes is the dominance of sound onsets, which carry binaural cues matching the direct sound rather than its echoes. Aspects of onset dominance can be demonstrated synthetically via localization of paired sounds separated by a brief delay (the “precedence effect”)–a rather extreme simplification of reverberation but a useful one for dissecting the roles of single reflections. In 1985, Brad Rakerd and Bill Hartmann published a precedence-like study [*JASA* 78(2), 524–533] using actual acoustic reflections induced by a single reflective surface positioned within an otherwise-anechoic space. Figure 3 of that paper shows that the effects on localization depend in a complex way on the direction and proximity of the sound source and its reflections. The impacts on binaural acoustic cues are similarly complex. Localization could be predicted by combinations of these cues only if each cue was weighted by its plausibility (i.e., likelihood given the sensory context). With this 1985 study, Rakerd and Hartmann, thus, revealed a second critical factor for accurate spatial hearing in reverberation: the context-dependent weighting of multiple imperfect sensory cues.

**11:25**

**1aPP5. A key figure in cochlear nonlinearity.** Elizabeth A. Strickland (SLHS, Purdue Univ., 715 Clinic Drive, West Lafayette, IN 47907, estrick@purdue.edu)

A figure that has had a huge influence on my research, and that I still use often, is Fig. 7 from a paper entitled “Basilar-membrane responses to tones at the base of the chinchilla cochlea” (Ruggero *et al*., 1997, *JASA* 101(4), 2151–2163). This figure shows that the velocity of the response of the basilar membrane grows compressively for a tone at the characteristic frequency and linearly for tones below the characteristic frequency. This was not the first study to show this effect, but because of the technique the experimenters used, they were able to cover a larger range of input levels than some previous studies. The figure is also very clear because of the plotting style used. This type of figure has been important in psychoacoustics in explaining how masking by a masker well below a signal frequency can be used to measure compression in the cochlea.

## *Invited Paper*

**11:35**

**1aPP6. The upper frequency limit of hearing is negatively correlated with head size in mammals.** Sunil Puria (Mass Eye and Ear, Harvard Med. School, 243 Charles St., Boston, MA 02111, sunil_puria@meei.harvard.edu)

My favorite graph in auditory science shows that the upper frequency limit of hearing is negatively correlated with the head size in mammals (defined as the functional interaural distance). It shows that sound localization is such an important requirement for survival that it was probably the driving factor that led to all the mammalian ear specializations required for the sensitivity and discrimination of high-frequency sounds. This was shown across 60 terrestrial mammals in several publications by Heffner and Heffner (e.g., 2016 Acoust Today). The wavelength of sound is inversely proportional to frequency. As frequency increases and the wavelength becomes shorter than a given head size, there is a level difference between the two ears. As head size decreases for smaller animals like mice, higher frequency hearing was required to exploit interaural spectral intensity differences needed to localize sounds. In order support such high frequency capabilities, the pinna, the middle ear, cochlea, and central mechanisms evolved together. Heffner and Heffner (1992) proposed that another important function of sound localization is to direct the eyes to the sound source and show a beautiful relationship between auditory and visual acuity. Hearing aids are limited to 3–4 kHz and, thus, do not make use of these high frequency capabilities. I have successfully argued to investors the importance of making wideband hearing aids and this culminated in the Earlens light driven wideband hearing aid (Levy *et al*. 2015 and Puria *et al*. 2016).

**11:45–12:00 Discussion**

## Session 1aSA

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

Christina Naify, Cochair
*Applied Res. Lab., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758*

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

Alexey Titovich, Cochair
*Naval Surface Warfare Center, Carderock Division, West Bethesda, MD*

Bhisham Sharma, Cochair
*Mech. Eng. - Eng. Mech., Michigan Tech Univ., 1400 Townsend Dr., Houghton, MI 49931*

### *Contributed Papers*

**10:00**

**1aSA1. Amplitude-dependent group velocity transformations in nonlinear lattices with nonlocal coupling.** Matthew D. Fronk (Physics, United States Naval Academy, 572C Holloway Rd., Annapolis, MD 21402, fronk@usna.edu), Caleb F. Sieck (Code 7160, U.S. Naval Research Laboratory, Washington, D.C.), Alec K. Ikei (Code 7160, U.S. Naval Research Laboratory, Washington, D.C.), and Matthew D. Guild (Code 7160, U.S. Naval Research Laboratory, Washington, D.C.)

Mass-spring lattices with weak stiffness nonlinearity undergo shifts in their band structure as a function of wave amplitude. Prior studies have characterized these shifts using perturbation techniques to derive small corrections to linear band structures. While lattice systems with nearest neighbor interactions are typically studied, growing attention is surrounding the effect of non-local elastic coupling. In this work, we report dramatic amplitude-dependent dispersion shifting in nonlinear lattices with nonlocal coupling. We demonstrate that nonlocal interactions with weak cubic stiffness have a profound impact on the amplitude-dependent group velocity of plane waves. Using a multiple scales perturbation technique, we obtain closed-form corrections to band structures that exhibit significant changes in sign and magnitude of group velocity compared to their linear counterparts. Analytical results are validated through direct numerical integration of the lattice equations of motion. [Work supported by the Office of Naval Research.]

**10:15**

**1aSA2. Exploring interesting effects of symmetric functionally graded unit cells for the design of nonlinear mechanical metamaterials.** Pravinkumar R. Ghodake (Dept. of Mech. Eng., Indian Inst. of Technol., Bombay, Mumbai, Maharashtra 400076, India, mech7pkumar@gmail.com)

Elastic waves at high frequencies (MHz) interact with microscopic imperfections, showing harmonic generation—where frequencies are multiples of the original wave. Elastic wave scattering by material imperfections was studied using theoretical models of circular and 1D strip-like nonlinear inclusions. Researchers (Tang *et al.*, 2012; Kube, 2017–18; and Achenbach & Wang, 2017–18) aimed to understand how the properties of these imperfections influence the scattered waves. Expanding on this understanding, Ghodake (2023) devised a novel unit cell comprising three nonlinear layers. This metamaterial mimics nonlinear light scattering, showing 'Raman Scattering' effects and exhibits tunable wave manipulation. Expanding on these findings, a functionally graded nonlinear unit cell is defined by establishing a spatially continuous distribution of the nonlinear parameter in the first and last layers, symmetrically centered around the central layer. Our novel approach manipulates the spatial distribution of nonlinear parameters, central layer intensity, and unit cell width to achieve unprecedented control over the harmonic response of scattered waves. Numerical experiments reveal non-intuitive behaviors in forward and backscattered harmonics. These harmonics can mysteriously emerge or vanish entirely due to complex interactions between the scattered waves. The proposed unit cell design empowers the inverse design of nonlinear metamaterials for tailored wave harmonic control.

**10:30**

**1aSA3. Impact of design variations of a micro-perforated panel on psychoacoustic metrics of transmitted sound.** Jiahua Zhang (Test Div., Siemens Digital Industry Software, 110 8th St., Graduate Program in Architectural Acoustics, Troy, NY 12180, jiahua.zhang@siemens.com), Jacques Cuenca (Test Division, Siemens Digital Industry Software, Leuven, Belgium), Laurent De Ryck (Test Div., Siemens Digital Industry Software, Leuven, Belgium), Lucas Van Belle (Dept. of Mech. Eng., KU Leuven, Heverlee, Belgium), and Elke Deckers (Dept. of Mech. Eng., KU Leuven Campus Diepenbeek, Leuven, Belgium)

Recent work reveals that sound travelling through metamaterials can lead to changes in sound perception [L. De Ryck *et al.*, Proceedings of ISMA and USD, pp. 1147–1162 (2018)]. As a starting point unravelling the impact of intricate vibro-acoustic design on perception, this study investigates the relationship between the intrinsic physical properties of micro-perforated panels (MPP) and several psychoacoustic metrics (loudness, sharpness, prominence ratio, and speech intelligibility index) in sound transmission conditions. A Sobol global sensitivity analysis is performed to quantify the impact of variations in MPP design parameters—panel thickness, perforation rate, hole diameter, and air cavity depth between panels—on these sound quality metrics [I. M. Sobol, MATCOM 55, 271–280 (2001)]. The analysis is conducted using both broadband and narrowband noise stimuli. The data trends suggest that the psychoacoustic effects of MPP acoustic treatments exhibit a non-trivial dependence on specific parameters of the panel setup. This also provides a framework towards innovative material design, where desired psychoacoustic metrics can help guide the manufacturing of metamaterials in order to meet specific sound quality targets. [The European Commission is gratefully acknowledged for its support of the Marie Sklodowska Curie program through the Horizon Europe DN METAVISION project (GA 101072415).]

### 10:45

**1aSA4. Enhancing sound absorption through geometric symmetry alterations in additively manufactured triply periodic minimal surface structures.** Janith Godakawela (Mech. Eng.-Eng. Mech., Michigan Technological Univ., 1400 Townsend Dr., Houghton, MI 49931, pgodakaw@mtu.edu) and Bhisham Sharma (Mech. Eng. - Eng. Mech., Michigan Tech Univ., Houghton, MI)

Cellular structures are essential for sound attenuation in various industrial and commercial applications. The stochastic nature of traditional foams makes it challenging to tailor their properties for specific applications. Additively manufactured Triply Periodic Minimal Surface (TPMS) structures have emerged as a promising alternative for sound absorption applications. These structures offer customizable sound absorption capabilities due to their unique geometric properties and porosity. While traditional methods such as altering porosity and introducing gradients have been used to modify their acoustic properties, they often lead to increased weight and fabrication difficulties. We propose leveraging their mathematical formulation to alter the inherent symmetry of TPMS structures as a novel approach to manipulate their acoustic properties. This allows property enhancement and customization without the weight penalties or fabrication complications associated with traditional methods. In this study, the samples are fabricated using the stereolithography additive manufacturing method and are tested for their sound absorption properties using the two-microphone normal incidence method. Current results demonstrate that specific alterations in geometric symmetry can significantly improve sound absorption properties, making it possible to tailor absorption characteristics in designing acoustic liners for specific applications.

### 11:00

**1aSA5. Characterizing porous materials using the two-cavity method: Numerical considerations and predictive accuracy.** Anthony Ciletti (Mech. Eng. - Eng. Mech., Michigan Technological Univ., 990 Hamilton Dr., Lucas, TX 75002, avcilett@mtu.edu), Janith Godakawela (Mech. Eng. - Eng. Mech., Michigan Technological Univ., Houghton, MI), Martha C. Brown (Aeroacoustics Branch, NASA Langley Research Center, Hampton, VA), and Bhisham Sharma (Mech. Eng. - Eng. Mech., Michigan Technological Univ., Houghton, MI)

The characteristic impedance and propagation constant are intrinsic parameters for understanding the acoustic absorption performance of a porous material. These parameters can be characterized by testing a bulk absorber in a normal incidence impedance tube via the two-thickness or two-cavity method. The two-thickness method requires two samples backed by a hard wall. The two-cavity method requires only one sample to acquire these intrinsic parameters–a useful feature for characterizing materials expensive or difficult to manufacture. This study uses the two-cavity method to predict the surface impedance and absorption coefficient of various bulk materials. The samples tested include open-cell metallic and non-metallic foam, and additively manufactured diamond-type Triply Periodic Minimal Surface (TPMS) samples. The predictions derived from the two-cavity method are compared to the predictions derived from the two-thickness method and the measured results. Certain numerical conditions of impedance in the two-cavity calculations, dependent on selected cavity depths, impacted local predictive accuracy. Cavity depth sizing guidelines to avoid these numerical conditions and improve predictive accuracy are provided.

### 11:15

**1aSA6. Acoustic black hole effect due to variation in duct wall impedance.** Jerry W. Rouse (Analytical Structural Dynamics, Sandia National Lab., Albuquerque, NM 87185, jwrouse@sandia.gov), Cameron A. McCormick (Simulation Model. Sci., Sandia National Lab., Albuquerque, NM), and Benjamin C. Treweek (Comput. Solid Mech. & Structural Dynamics, Sandia National Lab., Albuquerque, NM)

This presentation explores the acoustic black hole (ABH) effect achieved through variation of duct wall impedance. Unlike the method of Mironov and Pislyakov, which utilizes tapered annular rings within the duct to achieve the ABH effect, this approach leverages mechanical impedance variation to create the desired acoustic behavior. By engineering the impedance profile along the duct wall, an axially decreasing local phase speed results that mimics the event horizon of a black hole, causing sound waves to be significantly attenuated and trapped. Governing equations are derived and solved using both the WKB approximation and numerical methods. The solutions reveal that specific impedance profiles can effectively decelerate and absorb acoustic waves, resulting in substantial reduction in sound transmission. Verification is performed using the fully coupled structural acoustic finite element algorithm Sierra/SD. This research further paves the way for the development of innovative acoustic materials and devices which utilize ABH principles to achieve enhanced noise control and acoustic wave manipulation. [Sandia National Laboratories is a multimission laboratory managed and operated by National Technology and Engineering Solutions of Sandia, LLC, a wholly owned subsidiary of Honeywell International, Inc., for the U.S. Department of Energy's National Nuclear Security Administration under Contract DE-NA-0003525.]

### 11:30

**1aSA7. Estimating effective elastic properties of 3-D printed specimens using resonant ultrasound spectroscopy.** Abdullah Al Masud (Dept. of Mech. Eng., Texas Tech Univ., Box 41021, Lubbock, TX 79409-1021, Abdullah-Al.Masud@ttu.edu), Jingfei Liu, Paul F. Egan (Mech. Eng., Texas Tech Univ., Lubbock, TX), and Karl A. Fisher (Materials Eng., Lawrence Livermore Natl. Lab., Liveremore, CA)

This work continues the investigation of resonant ultrasound spectroscopy (RUS) as a method to characterize the elastic properties of 3-D printed lattice structures. Here, we are focused on lattices fabricated using stereolithography (SLA) with a polyurethane-like resin (Amber) and featuring beam-based unit cells. The RUS analysis leverages the assumption that the diagnostic ultrasound wavelength exceeds the characteristic dimensions of the unit cell. This allows us to treat the 3-D printed structure as a transversely isotropic continuum with effective material properties reflecting the underlying unit cell symmetry. The RUS technique retrieves the five elastic constants for these structures, which are subsequently employed to calculate the elastic moduli. We further compare the elastic coefficients obtained via RUS from both numerical simulations (finite element analysis) and experimental measurements of the eigenmodes. Initial results demonstrate good agreement between experiments and models, highlighting the potential of RUS as a valuable tool for characterizing the elastic behavior of resin-based lattices.

### 11:45

**1aSA8. Unveiling nonlinear phi-bit dynamics in elastic systems: Advancing quantum-inspired computing.** Abrar Nur E. Faiaz (Mech. Eng., Wayne State Univ., 4206 Saint Antoine, Detroit, MI 48201, hr1642@wayne.edu), Akinsanmi S. Ige (Univ. of Arizona, Tucson, AZ), Kazi Tahsin Mahmood (Mech. Eng., Wayne State Univ., Detroit, MI), M. Afridi Hasan (Univ. of Arizona, Tucson, AZ), M. Arif Hasan (Mech. Eng., Wayne State Univ., Detroit, MI), Pierre A. Deymier (Mater. Sci. and Eng., Univ. of Arizona, Tucson, AZ), Keith Runge (Univ. of Arizona, Tucson, AZ), and Joshua A. Levine (Univ. of Arizona, Tucson, AZ)

Phi-bits, the classical mechanical analogs of qubits, play a pivotal role in the development of quantum-analog computing systems. Understanding the nonlinear processes governing control and interconnectivity among phi-bits is imperative for their advancement. These phi-bits, existing as acoustic waves within arrays of interconnected waveguides, exhibit a remarkable ability to maintain coherent superpositions of two states under external nonlinear driving forces. Manipulating the frequency, amplitude, and phase of external drivers allows precise control over phi-bit states. To analyze and predict the nonlinear response of phi-bits to external stimuli, we have developed a discrete element model. This model comprehensively captures various types, strengths, and orders of nonlinearities stemming from intrinsic medium coupling between waveguides and external factors like signal generators, transducers, and ultrasonic couplant assemblies. Our study unveils significant insight, highlighting how nonlinearity type, strength, order, and damping impact the complex amplitudes' modulus and phases in the coherent superposition of phi-bit states, with a notable impact on their predictability and stability, particularly at high damping levels. This investigation explores the controlled creation of phi-bits to observe the superposition of states, essential for advancing phi-bit-based quantum analogue information processing platforms.

## Session 1aSC

## Speech Communication: Student Led Tutorials

Benjamin V. Tucker, Chair
*Commun. Sci. and Disorders, Northern Arizona Univ., 208 E. Pine Knoll Dr., P.O. Box: 15045, Flagstaff, AZ 86011*

### *Invited Papers*

**10:00**

**1aSC1. Analyzing acoustic data with Bayesian finite mixture models: A practical introduction.** Scott J. Perry (Translation and Language Sciences, Universitat Pompeu Fabra, Roc Boronat, 138, Barcelona, Spain, sperry1@ualberta.ca)

Acoustic correlates of speech can present empirical distributions that create issues for the linear models we often use to analyze them. In some such cases, we have reason to believe that there are multiple distributions responsible for producing the data. One example is the distribution of voice onset time (VOT) in the phonologically voiced stop series /b,d,g/ in North American English. While more often produced as a short lag stop with a VOT near zero, it can also be produced with prevoicing. This leads to a bimodal distribution of VOT values for these segments. The first study of VOT by Lisker and Abramson (1964) wisely decided to report descriptive statistics for each of these pronunciation variants separately, recognizing that listing a single mean and range would have been a poor characterization of the data and potentially misleading to readers. This tutorial introduces Bayesian finite mixture models, which can model a dependent variable using a combination of two or more distributions. This is useful in situations like the one described above for English VOT, as having two distributions allows us to model both short-lag and prevoiced VOT values together in the same model. The tutorial begins with a brief conceptual explanation of these models, followed by a walkthrough of an analysis of simulated English VOT data using the package "brms" (Bürkner, 2017). An emphasis is placed on setting priors, especially setting informative priors for the intercepts of the two distributions.

**10:40**

**1aSC2. What is possible with speecollectr.** Abbey L. Thomas (Brain and Behav. Sci., The Univ. of Texas at Dallas, 800 W Campbell Road, Richardson, TX 75080, abbey.thomas@utdallas.edu)

The need for remote speech data collection during the Covid-19 pandemic prompted the development of the open-source software speechcollectr, an R package for building speech experiment interfaces. This package relies on Shiny, an R package for web development. Experiments built with speechcollectr can be run from a local machine or distributed over the Internet for web-based protocols. Designed primarily for researchers familiar with R, this tutorial will demonstrate some foundational concepts of reactive programming in R. This type of programming allows researchers to construct interfaces that respond to participant inputs. Such interfaces include surveys or audio recording and playback interfaces, all of which may be created and displayed to participants with speechcollectr functions. A key feature of the package is its suite of functions for recording uncompressed WAV files from the participant's browser. Recordings created with speechcollectr are uploaded automatically to the experimenter's server and may be subjected to the automated acoustic evaluation performed by speechcollectr's evalWav function. This tutorial will include a brief walkthrough of a speech production task to demonstrate the speechcollectr protocol for audio recording and, more broadly, to show how speech data may be collected using R.

**11:20**

**1aSC3. Reconsidering the formant correction formula in VoiceSauce.** Yuan Chai (Dept. of Linguistics, Univ. of Washington, Guggenheim Hall, 4th Floor Box 352425, Seattle, WA 98105, yuanchai@uw.edu) and Patricia A. Keating (Linguistics, UCLA, Los Angeles, CA)

VoiceSauce (Shue *et al.*, 2011) is an acoustic signal analysis software widely used for analyzing voice quality for various purposes (e.g., linguistic phonation acoustics: Keating *et al.*, 2023; articulatory correlates: Wu & Zhang, 2023; speech disorder: Asiaee *et al.*, 2022; singing: Meireles, 2016). One of the voice quality measurements provided by VoiceSauce is the amplitude difference between the first and second harmonics (H1–H2), a measurement widely used as a proxy for the degree of constriction of the vocal folds. The built-in formant correction formula (Iseli *et al.*, 2007) in VoiceSauce enables users to remove the amplifying effect of vowel formants on harmonic energy, and consequently compare the harmonic energy across different vowel qualities. However, we found that the formant correction formula is unsuitable for correcting harmonics for formants much lower than the harmonic frequency. Such a correction will result in a large boost in the corrected harmonic energy. In this tutorial, we will demonstrate the rationale and problems of the formant correction formula used in VoiceSauce, how to modify the formant correction formula in VoiceSauce, and how to add additional harmonic parameters to VoiceSauce.

**11:40–12:00 Discussion**

# Session 1aSP

## Signal Processing in Acoustics: Signal Processing Potpourri I

Kendal Leftwich, Cochair
*Physics, Univ. of New Orleans, 1021 Science Building, New Orleans, LA 70148*

Natalia Sidorovskaia, Cochair
*Physics, Univ. of Louisiana at Lafayette, P.O. Box 44210, Lafayette, LA 70504-4210*

**Chair's Introduction—10:00**

### *Contributed Papers*

**10:05**

**1aSP1. Classification of morphologically separated synthetic aperture sonar imagery via dual-representation convolutional neural networks.** Geoff Goehle (Pennsylvania State Univ., 225 Science Park Rd., State College, PA 16803, goehle@psu.edu), Benjamin Cowen, J. Daniel Park, and Daniel C. Brown (Pennsylvania State Univ., State College, PA)

Morphological component analysis (MCA) involves the separation of time series into components based on morphological factors, such as duration or envelope type. Acoustic returns from sonar systems can be separated into short-duration and long-duration components using MCA, and the resulting time series formed into traditional synthetic aperture sonar (SAS) images. We present an application of this approach to classification of data from hollow and solid spherical targets collected using a linear in-air SAS system (AirSAS). AirSAS time series were separated into morphological components and, utilizing online image formation, the resulting imagery processed through a multi-branch convolutional neural network (CNN) architecture to classify targets by type. Classification results are included for a variety of scene backgrounds and representation modalities. This is a challenging classification problem since the targets have the same exterior geometry and are distinguished by non-specular acoustic phenomenology.

**10:15**

**1aSP2. Target-direction sound extraction using a hybrid DSP/deep learning approach.** Jingya Yang (National Tsing Hua Univ., No.101, Sect. 2, Kuang-Fu Rd., Hsinchu, Taiwan 300044, R.O.C., Hsinchu 300, Taiwan, jing.ya161@gmail.com), Yicheng Hsu, Jiaxuan Fan (National Tsing Hua Univ., Hsinchu, Taiwan), Jiahsin Lin, Hsuanyu Shih, Zongen Wu, Tingyu Liu (LITEON Technol., Taipei, Taiwan), and MingSian Bai (National Tsing Hua Univ., Hsinchu, Taiwan)

The goal of Target Sound Extraction (TSE) is to isolate sound signals of interest from a mixture of overlapping sounds. In learning-based TSE approaches, the cues, such as sound embedding, which provide information about the sound characteristics, are used as auxiliary features for the Deep Neural Networks (DNNs) to extract the target sound signals. However, in the presence of strong interference, the extraction performance is reduced when the model relies solely on the sound embedding. To this end, this study proposes an efficient Target-Direction Sound Extraction (TDSE) system by integrating DSP-based spatial feature extraction and a DNN backend. In spatial feature extraction, the spatial blocking technique is used to analyze the signals out of the target direction. The DNN backend is based on the Waveformer backbone, which has shown promising results and real-time capability on single-channel TSE tasks. Experimental results show that the proposed TDSE system can achieve 2–3 dB improvement over the Waveformer baseline. In addition, the proposed system performs well on a variety of array setups, including those not included in the training set.

**10:25**

**1aSP3. A multichannel audio tagging and localization system for home surveillance.** Yuhsin Lai (National Tsing Hua Univ., No.101, Section 2, Kuang-Fu Rd., Hsinchu, Taiwan 300044, R.O.C., Hsinchu 300044, Taiwan, yuhsinlai@gapp.nthu.edu.tw), Yicheng Hsu, Jiaxuan Fan (National Tsing Hua Univ., Hsinchu, Taiwan), Jiahsin Lin, Hsuanyu Shih, Zongen Wu, Tingyu Liu (LITEON Technol., Taipei, Taiwan), and MingSian Bai (National Tsing Hua Univ., Hsinchu, Taiwan)

Audio Tagging (AT) is a critical technique in smart home applications, enabling continuous monitoring of specific sound events for subsequent surveillance systems. Although Deep Neural Networks (DNNs) can provide promising tagging results, the performance degrades significantly in adverse acoustic environments. In addition, if the AT system can provide both the tagging and the location results, it can greatly enhance the system's ability to capture sound events. To this end, a multichannel audio tagging system based on Convolutional Neural Network (CNN) is proposed to simultaneously tag and localize the sound event. Interchannel Phase Differences (IPDs) between each pair of microphones are used as the spatial feature for the input to the model. Instead of predicting the angle or zone index, the proposed system outputs a unit vector pointing to the sound event for localization. A novel loss function is introduced that computes the square of the cosine similarity between the ground truth and the estimated unit vector, allowing for more accurate localization performance. Experimental results show that the proposed system outperforms the single-channel baseline under strong interference, making it well suited for real-world applications. In addition, the model trained with the proposed loss function can greatly improve the localization accuracy.

**10:35**

**1aSP4. Time-frequency spectral optimization of underwater sonar acoustic returns for the purpose of characterization using spectral descriptors.** Wendy N. Newcomb (GTRI, Georgia Tech, 3142 Chase Ct SW, Marietta, GA 30008, wendy.newcomb@gtri.gatech.edu), Saaketh Reddy (Mech. Eng., Georgia Tech, Atlanta, GA), Aprameya Satish, and Alessio Medda (Georgia Tech Research Inst., Smyrna, GA)

Underwater sonar acoustic returns of objects on the seafloor have many time varying components including the specular geometric backscatter, scattering from vibrational dynamics, helical, circumferential, and meridional Rayleigh waves, guided circumferential Lamb waves, and multipath from wave field reflections from both the object of interest and scene. These features of the acoustic return can reveal information about an object's shape, structure, material composition, and relationship to the background scene. Analysis of the acoustic return in the time-frequency domain is, therefore, useful for mine counter measure objectives.Optimizing the performance of various time-frequency transforms supports better understanding of the

spectral behavior related to the time varying components of the acoustic return.This talk will examine the accuracy of several time-frequency representations including the short-time Fourier transform, the smoothed pseudo Wigner-Ville distribution, the empirical wavelet transform, and the chirplet transform.This work will take a detailed look at the relationships between measures of concentration, such as Renyi entropy, Stankovic concentration, frequency specificity, and resolution. Techniques are devised to improve the fidelity of the signal's spectral content for each respective transform and to compare performance across transforms. The impact of this process is then assessed against a collection of time-frequency spectral descriptors.

**10:45**

**1aSP5. Generation and use of very wideband chip signals for coded underwater applications.** Yasin Kumru (Electric. and Electron. Eng., Bilkent Univ., Ankara 06800, Turkey, yasin@ee.bilkent.edu.tr), A. Sinan Taşdelen (None, Ankara, Turkey), and Hayrettin Köymen (None, Ankara, Turkey)

We present the generation and use of coded signals with very wideband chip signals for underwater applications. We performed measurements using a parametric array in a large water tank. We fabricated a transducer consisting of a 3 mm thick and 75 mm-by-75 mm square-shaped PZT ceramic plate, which is matched to water media at the radiating face and terminated by a very low impedance at the back. We used the square-root amplitude modulation and complementary Golay sequences to code the driving signals centered around 855 kHz primary frequency. We employed a matched filter at reception. Our measurement results showed that generating coded signals with varying code and chip signal lengths at 10–80 kHz difference frequency range is possible. We measured that the normalized matched filter output at 40 kHz for a 2-chip coded signal with the widest bandwidth chip signal length of 2-cycle/chip is 0.94, which indicates a strong correlation and is consistent with those measured at other difference frequencies. As a result, we obtained well-defined coded signals with a chip length of 2-cycle/chip at a 10–80 kHz frequency range, although we also observed coded signals with shorter chip signals.

**10:55–11:00 Break**

**11:00**

**1aSP6. Strategies for preprocessing speech to enhance neural model efficiency in speech-to-text applications.** Andrzej Czyzewski (ETI Faculty, Multimedia Systems Dept., Gdansk Univ. of Technol., Narutowicza 11/12, Gdansk 80-233, Poland, andczyz@gmail.com)

A comprehensive study on the impact of advanced speech preprocessing strategies on the performance of speech-to-text models is presented. Our approach incorporates noise augmentation, speech rate descreasing, and anonymization to create a more robust training dataset. Additionally, we utilized voice cloning techniques to generate thousands of supplementary recordings, significantly expanding our dataset. These preprocessing strategies aim to improve the accuracy and efficiency of speech recognition systems. Our experiments demonstrate a notable reduction in Word Error Rate (WER) by an average of 3.5% and Character Error Rate (CER) by 4.5%. These results indicate that the applied preprocessing methods effectively enhance the model ability to generalize across diverse and challenging speech inputs. We provide insights into the preprocessing techniques, their implementation, and the subsequent impact on model performance, offering valuable guidance for future advancements in speech-to-text technologies. [The Polish National Center for Research and Development (NCBR) supported the project "ADMEDVOICE" (INFOSTRATEG4/0003/2022), focusing on voice processing applications in the healthcare sector.]

**11:10**

**1aSP7. Nonlinear classification and regression in underwater sonar applications using convex-complex neural networks.** Andrew J. Christensen (Elec. and Comput. Eng., Univ. of Iowa, 3100 Seamans Ctr., Iowa City, IA 52242, andrew-christensen@uiowa.edu), Ananya Sen Gupta (Dept. of Elec. and Comput. Eng., Univ. of Iowa, Iowa City, IA), and Ivars Kirsteins (Naval Undersea Warfare Center, Newport, RI)

Recently, the sonar signal processing community has seen increased interest in neural networks for nonlinear classification and regression. While innovative neural network architectures have achieved remarkable results in other fields such as computer vision and natural language processing, their application to sonar data presents challenges. The multiple nonlinearities in these networks make it difficult to interpret results. Recent works such as Grad-CAM and Layer-wise Relevance Propagation attempt to empirically visualize what features are being selected by the network, but there has been little theoretical work to explain why neural networks perform so well. However, recent work by Pilanci *et al*. has shown that two-layer neural networks with a ReLU activation function can be reformulated as a convex optimization that can be solved optimally in polynomial time. Their results rely on the observation that ReLU neural networks are piece-wise linear models that lift the original data into a higher dimensional feature space. Using a valid activation function, we show that this Convex Neural Network framework can be extended to applications where the input data is complex-valued. We evaluate our algorithm on simulated sonar data. [This talk will present research funded by DoD Navy (NEEC) Grant number N001742010016 and the ONR grant numbers N000142112420 and N000142312503.]

**11:20**

**1aSP8. Representation and temporal prediction of ocean states with fully complex autoencoders.** Timothy Linhardt (College of Elec. and Comput. Eng., Univ. of Iowa, 103 South Capitol St., Iowa City, IA 52240, tlinhardt@uiowa.edu), Nicholas Durofchalk (Dept. of Physics, Naval Postgraduate School, Monterey, CA), Ananya Sen Gupta (Dept. of Elec. and Comput. Eng., Univ. of Iowa, Iowa City, IA), Kay L. Gemba (Dept. of Physics, Naval Postgraduate School, Monterey, CA), and Ivars Kirsteins (USN NUWC, Newport, RI)

The one-directional oceanic channel is very dynamic and complicated. Changing temperatures affect sound speed, the tidal effects of the lunar cycle change depth and pressure, and the physical aperturing from undersea terrain filters the ever-changing multipath scattering rays. All of these effect the impulse response of the channel. This is a time-variant system with a very high number of degrees of freedom, and thus difficult to model and make predictions for. Autoencoders allow for the projection of the space of the varying channel transfer functions to a much lower dimensional "encoding space" from which physical interpretations may be gleaned, and constructing an autoencoder with fully complex neuron layers allows a better representation of the complex-valued transfer function space. Within the encoding space, temporal trends can be observed at scales of minutes or days, and ongoing research explores the predictive capacity of trends at these scales. [Work is done in collaboration with the Naval Postgraduate School and funded by US Department of Defense NEEC grant N00174-20-1-0016.]

**11:30**

**1aSP9. Lessons learned in developing a distributed in-air ensonification and sensing lab test bed.** Jayden Smith (GTRI, 7220 Richardson Rd. Building 1, Smyrna, GA 30080, jayden.smith@gtri.gatech.edu), Wendy N. Newcomb, Natalya Gage, Mike Baden, Aprameya Satish, and Daniel Cook

In recent years, the use of commercial off-the-shelf audio acoustic hardware has been leveraged in the sonar community to emulate in-water sonar acquisitions in a controlled lab environment to benefit research and student education. In-air sonar test beds offer the opportunity to quickly create experimental data which would be costly and time consuming if done in-water. Tests may include different source receiver configurations (monostatic, bistatic, and multistatic) or experimentation with diverse materials and waveforms. Accurate positioning measurements result in the ability to create synthetic aperture imagery. The Georgia Tech Research Institute (GTRI) has developed the Distributed In-air Ensonification & Sensing Lab (DIESEL) testbed to contribute to collaboration and dataset generation for the mutual benefit of dataset sharing in the sonar community. There are several critical aspects of test bed development that must be evaluated and verified to bridge the gap between in-air acoustics and in-water sonar simulation and ensure proper system functioning. To further community collaboration and education in this area, this talk will discuss some of the lessons learned from challenges in setting up hardware, debugging aliasing, mitigating noise, and room artifacts and modifying signal processing for sonar dataset simulation in monostatic, multistatic, and bistatic acoustic configurations.

11:40

**1aSP10. Addressing non-linearity in tomography using ray theory.** Paul Hursky (Applied Ocean Sciences LLC, 4825 Fairport Way, San Diego, CA 92130, paul.hursky@gmail.com), Alison B. Laferriere (Applied Ocean Sciences LLC, La Jolla, CA), and Emanuel F. Coelho (Science, Applied Ocean Sciences, LLC, Springfield, VA)

Ocean acoustic tomography has been formulated as a linearized inversion of a forward model based on ray tracing. In this formulation, we are solving for refinements of a zero-order sound speed model that reproduce measured travel times, travel times that are also being modeled by the ray tracer. The linear approximation is based on calculating travel time perturbations by integrating the sound speed refinements over the ray paths, assuming the ray paths themselves are not impacted by the sound speed refinements. Clearly, the offset between the presumed (zero-order) sound speed and the actual sound speed may be so great that the ray paths need to be modified, not just the sound speeds being integrated along these paths to reproduce the travel time perturbations. There has been a lot of work on using full wave propagation models to predict the sensitivity of travel times to sound speed perturbation throughout the medium. We will present the use of ray theory, leveraging dynamic ray tracing to gain practical insights into how to alert a user and to overcome non-linearities that arise in tomography.

TUESDAY MORNING, 19 NOVEMBER 2024 10:00 A.M. TO 12:00 NOON

## Session 1aUW

## Underwater Acoustics: Webinar on Python and Data Analysis

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

Tyler J. Flynn, Cochair
*Acoustics & Electromagnetics, Johns Hopkins Applied Physics Lab., 1231 Beal Ave., Ann Arbor, MI 48109*

### *Invited Paper*

10:00

**1aUW1. UW data analysis with Python webinar.** Tracianne B. Neilsen (Physics and Astronomy, Brigham Young Univ., N251 ESC, Provo, UT 84602, tbn@byu.edu) and Tyler J. Flynn (JHU/APL, Laurel, MD)

The collection and analysis of data are central to all areas of acoustics. While data analysis has been performed extensively with several coding languages in the past, Python has been gaining prominence. This webinar will highlight some common applications of data analysis in Python using examples from underwater acoustics. The webinar will consist of a series of presentations by underwater acoustics experts. The first presentation will be a brief overview of some openly available datasets. Examples of spectral analyses on a few types of data will then be presented. Following the spectral analyses, a few array processing methods will be demonstrated and applications to data presented. The link to a GitHub repo for the codes and data samples used in this special session webinar will be provided to participants at the beginning of the session. Those familiar with Python and Jupyter Notebooks will be able to clone the repo and follow along during the webinar. The webinar will end with a discussion regarding additional freely available resources for data analysis in ocean acoustics.

## Session 1pAA

# Architectural Acoustics: Acoustics of Sustainable Building Assemblies and More

Arthur W. van der Harten, Cochair
*Acoustics, Acoustic Distinctions/Open Research in Acoustical Science and Education, 400 Main St. Ste. 600, Stamford, CT 06901*

Jonathan Broyles, Cochair
*University of Colorado, Boulder, CO*

Kristen Murphy, Cochair
*Acentech Inc, Cambridge, MA 01238*

Adam Bettcher, Cochair
*Seattle, WA*

**Chair's Introduction—3:00**

### *Contributed Paper*

**3:05**

**1pAA1. Achieving LEED acoustics credits: Three project case studies.** Josh Thede (Acoustical Consulting, BrightTree Studios, 301 Brush Creek Rd., Warrendale, PA 15086, jthede@brighttreestudios.com)

Leadership in Energy and Environmental Design (LEED) is a widely recognized green building standard. This presentation will highlight three completed LEED buildings that successfully earned acoustics credits or prerequisites as part of their certification. Each project will discuss the strategies, solutions, and documentation provided to meet the acoustics requirements. Discussion includes an overview of various LEED scorecards including building design and construction (BD+C), interior design and construction (ID+C), LEED for schools, and LEED for Healthcare. Acoustics requirements from previous LEED versions v3 and v4.1 will be summarized. Future trends and the proposed LEEDv5 changes will be discussed. Together we'll explore the relationship between acoustics and sustainability and ponder the appropriate noise control and acoustics requirements for high performance green buildings.

### *Invited Papers*

**3:25**

**1pAA1. The building material renaissance: A review of innovative low-carbon, healthy, and acoustically viable building materials.** Jonathan M. Broyles (Civil, Environmental and Architectural Eng., The Univ. of Colorado Boulder, Engineering Ctr., ECOT 441, Boulder, CO 80305, Jonathan.Broyles@colorado.edu)

With the pressing need to curb carbon emissions in buildings, architects and designers are increasingly determining low-carbon mitigation strategies across all building disciplines. One solution is the selection of novel low-carbon construction materials, often composed of recycled materials, bio-based products, or wastes. These new building materials are commonly manufactured without harmful chemicals, thereby making the building safer for its occupants. Furthermore, many of the building materials have acoustic benefits, such as improved sound absorption at mid-frequencies. While this new era of building materials encourages low-carbon and healthy buildings, many designers are unaware of them and their advantages. This presentation aims to educate acousticians, designers, and other practitioners by providing a review of innovative building materials (e.g., Pliteq's GenieMat, fSorb's Acoustic Panels) that both aid decarbonization efforts and improve building acoustics. An additional outcome of this work is an early version of an open-access dataset of building materials and products for their carbon emissions and acoustic benefit. Overall, this presentation continues the theme of reducing the carbon emissions in the building environment while also considering occupant health and acoustic performance.

**3:45**

**1pAA3. Decarbonizing architectural acoustics: Assessing the $CO_2$ emissions of acoustic systems in an office building in Milan.**
Andrea Giglio (Architecture, Built Environment and Construction Eng. Dept., Politecnico di Milano, Via Giuseppe Ponzio, 31, Milan 20133, Italy, andrea.giglio@polimi.it), Marcella Benini (Deerns Italia, Milan, Italy), Jonathan Broyles (Dept. of Civil, Environmental and Architectural Eng., Univ. of Colorado, Boulder, Boulder, CO), and Ingrid Paoletti (Architecture, Built Environment and Construction Eng. Dept., Politecnico di Milano, Milan, Italy)

Nowadays, the Architectural-Engineering-Construction (AEC) market requests buildings with both low $CO_2$ emissions and high performance across all building disciplines, including architectural acoustic systems, to satisfy design regulations and protocols. However, not all acoustic material systems guarantee both of them. Previous research has suggested that design trade-offs exist when reducing $CO_2$ emissions and achieving high-performing room acoustics but a study on a case study is absent. In response, this paper analyses the carbon data set and material quantities of the acoustic systems of a 7-story office building in Milan, Italy. The data are gathered by outsourcing them from the LCA and further analyzed by considering the technical data sheets of the as-built products. The analysis highlights the important role of acoustics in the total amount of $CO_2$ emissions of the building. In particular, horizontal and vertical partitions have the highest impact due to the large material intensities. In conclusion, this research demonstrates that a performance-based LCA analysis is needed to better evaluate the relationship between acoustic performance and $CO_2$ emissions, with future research needed to understand these complex design trade-offs further.

**4:05**

**1pAA4. Designing beyond the acoustical requirements of sustainable design.** Jessica S. Clements (Acoustics, SmithGroup, 999 Peachtree St. NE, Ste. 400, Atlanta, GA 30309, jessica.clements@smithgroup.com) and Danna Lopez Richey (Sustainability Studio, Newcomb & Boyd, LLP, Atlanta, GA)

LEED, WELL, GreenGlobes, and The Living Building Challenge. There are many sustainable design guidelines now being used and developed that include goals related to architectural acoustics and noise control. How can we as consultants bring value to our clients and design partners? Beyond meeting the specifically outlined requirements to achieve points, there are ways we can help our design partners improve the sustainability of the projects. This presentation will review ways to get started on a sustainable design, to start discussions on meeting the specific acoustical goals, and ways to look beyond meeting STC, RT, or NC requirements and bring more to the table.

**4:25**

**1pAA5. Investigating embodied carbon of common floor ceiling assemblies and low carbon buildups to meet new requirements.**
Aedan Callaghan (Pliteq Inc., 4211 Yonge St., Ste. 400, Ste. 404, Toronto, Ontario M2P 2A9, Canada, acallaghan@pliteq.com)

The buildings and construction sector is known to be the largest global emitter of greenhouse gasses, representing 37% of all emissions. International building regulations are poised to incorporate embodied carbon targets for new building construction, with restrictions expected to become stricter over time. California became the first state to add embodied carbon emission control as part of the California Green Building Standards Code. Some cities are also enacting policies like Toronto where new city owned buildings must demonstrate less than 350kg/CO2/m2. With this increased focus and requirements, whole building life cycle assessments will become an important tool in the architecture, engineering and construction (AEC) design process. These evolving environmental regulations demand a deeper understanding of the relationship between the acoustic performance and embodied carbon of common assemblies. This work compares this relationship in several construction types, including concrete, mass timber, and wood frame assemblies. This relationship is evaluated by using environmental product declarations (EPDs) and acoustic testing of various assembly types to offer a holistic comparison of the performance metrics. The findings will support development of best practices for selecting materials and acoustic design approaches to balance low embodied carbon with high acoustic performance and support the construction industry to meet future environmental standards.

## Session 1pAB

## Animal Bioacoustics: Acoustical DJ: Mixed & Matched Research Topics on Sound

Kerri D. Seger, Chair
*Applied Ocean Sciences, 2127 1/2 Stewart St, Santa Monica, CA 90404*

### *Invited Paper*

"Acoustical DJ: Mixed & Matched Research Topics on Sound" will be like other societies' "PowerPoint Roulette" sessions. Speakers do not need to submit abstracts to participate. They merely need to arrive in the virtual meeting room and say they want to participate. Each player will be given an unknown-to-them set of slides to present in an off-the-cuff style. Ahead of time, the session chair will collect past talks from ASA members and collate the slides randomly to form all new, never-before-seen, "research" presentations. Each "talk" will be 3 slides long. At the beginning of the session, a role of willing players will be taken and placed into an arbitrary order. Each presenter will have 15 min to present their slides. A likely way to split those 15 min could be (1) taking 3 min to prepare thoughts once given the first slide, (2) discussing each slide for 3 min, and (3) leaving 3 min for fun Q&A. Speakers should think on their feet, interpret graphs whether or not the axes are labelled, combine seemingly unrelated topics into an attempt at coherency, and bring their funny bones and best impressions of whatever animal they may be saddled to sound like.

**3:15–5:00 PowrPoint Roulette**

TUESDAY AFTERNOON, 19 NOVEMBER 2024 3:00 P.M. TO 4:15 P.M.

## Session 1pAO

## Acoustical Oceanography: Teaching Curriculum and In-class Demos

David R. Barclay, Chair
*Dept. of Oceanography, Dalhousie Univ., PO Box 15000, Halifax B3H 4R2, Canada*

### *Contributed Paper*

**3:00**

**1pAO1. Acoustical oceanography meets bioacoustics: Teaching ocean acoustics to biologists and physicists at the University of Victoria.** Stan Dosso (School of Earth and Ocean Sciences, Univ. of Victoria, Victoria, BC V8W 2Y2, Canada, sdosso@uvic.ca) and William D. Halliday (Wildlife Conservation Society Canada, Whitehorse, Yukon, Canada)

The graduate-level Ocean Acoustics course at the University of Victoria is targeted at students in different departments with diverse backgrounds, ranging from physics to biology. We lead the students through a series of lectures, discussions, and assignments to provide fundamental knowledge of acoustical oceanography and marine bioacoustics, with subjects ranging from acoustic propagation modelling and geophysical inverse theory to soundscape ecology and the impacts of underwater noise on marine life. Given the varied background of the students, we must balance the material so that the course covers physical concepts without being too "math heavy" for the biologists, and so the physicists can understand and appreciate the biology. We frame assignments around skills that the students can use in their graduate research, and also give the students opportunities to review scientific papers and give presentations to the class. We have taught this course for two years now, and students have come out with a deeper understanding of the physics of sound and how it is affects marine life, providing a good foundation for their graduate research in various aspects of underwater acoustics.

## *Invited Papers*

**3:15**

**1pAO2. Ocean acoustics education in the MIT-WHOI Joint Program.** Julien Bonnel (Woods Hole Oceanographic Inst., 266 Woods Hole Rd., MS# 11, Woods Hole, MA 02543-1050, jbonnel@whoi.edu), Andone C. Lavery (AOPE, Woods Hole Oceanographic Inst., Woods Hole, MA), and John A. Colosi (Oceanography, Naval Postgraduate School, Monterey, CA)

The Massachusetts Institute of Technology (MIT) - Woods Hole Oceanographic Institution (WHOI) Joint Program (JP) is a five-year doctoral program conferring a PhD degree awarded by both institutions. It is organized in five disciplines: Applied Ocean Science and Engineering (AOSE), Biological Oceanography, Chemical Oceanography, Marine Geology and Geophysics, and Physical Oceanography. Because the MIT-WHOI JP is so broad in scope, it is an ideal place for advanced studies in acoustical oceanography. The MIT-WHOI acoustic curriculum falls mostly under the AOSE umbrella. Students can take classes both at WHOI and MIT and can do research on a variety of topics covering acoustic propagation and scattering, undersea navigation/communication, remote sensing and environmental inversion, bioacoustics, robotics, and high latitude acoustics. JP students have access to courses, programs and resources at one of the top oceanographic research institutions in the world (WHOI) and one of the top research universities in the world (MIT). By benefiting from the best of these two institutions theoretical and sea going strengths, JP students can become part of the next generation of ocean science leaders, allowing them to pursue careers all over the world in diverse areas including universities, research institutions, government, private industry or NGOs.

**3:30**

**1pAO3. Acoustical oceanography curriculum at The University of Texas at Austin.** Megan Ballard (Applied Research Lab., The Univ. of Texas at Austin, Austin, TX), Preston S. Wilson (Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, 204 East Dean Keeton St., Mail Stop: C2200, Austin, TX 78712-0292, pswilson@mail.utexas.edu), and Mark F. Hamilton (Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX)

The acoustical oceanography (AO) curriculum at The University of Texas at Austin (UT) consists a number of courses and thesis research, which often includes field work. The core course is EE/ME 384N-5, Underwater Acoustics, which covers acoustic properties of the ocean, propagation, reflection, reverberation, scattering and target strength, ocean noise, array and signal processing and basic sonar design. The course is offered in alternate years and is cross-listed as both an electrical and mechanical engineering course. Prior to this, students usually take two semesters of physical acoustics: EE/ME 384N-1 and 2, Acoustics I and II, which covers plane waves in fluids, transient and steady-state reflection and transmission, lumped elements, refraction, ray acoustics, absorption and dispersion, spherical and cylindrical waves, radiation and scattering, multipole expansions, Green's functions, waveguides, Fourier acoustics, and Kirchhoff theory of diffraction. Both are offered every year. Another course commonly taken by AO students is EE/ME 384N-3: Electromechanical Transducers, which covers basic modeling, analysis and design of acoustics and vibration transducers, including calibration. Recent student thesis topics have included marine acoustic ecology, the investigation of methane seeps, acoustic seagrass monitoring, and the assessment of glacial processes. Appropriate courses in UT's natural and earth sciences departments supplement the acoustics courses.

**3:45**

**1pAO4. Acoustics for all at UNH.** Anthony P. Lyons (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, Durham, NH 03824, anthony.lyons@ccom.unh.edu), Jennifer Miksis-Olds (Ctr. for Acoust. Res. and Education, Univ. of New Hampshire, Durham, NH), and Laura Kloepper (Dept. of Biol. Sci., Univ. of New Hampshire, Durham, NH)

Specialized education in acoustics is critical for students at all levels, from undergraduates to those already graduated and doing basic research in academia or more applied work in academia, state and federal agencies, or industry. A three-component plan has been designed and is being implemented to improve acoustics education and training at all levels by the University of New Hampshire (UNH). The three components are specifically designed to serve (1) residential university students seeking formal degrees and graduate certificates, (2) non-residential students seeking formal degrees and graduate certificates, and (3) non-traditional students seeking continuing education and professional development opportunities in ocean acoustics. For non-residential and continuing education students, graduate level courses are being transitioned to blended formats in support of distance education efforts. These courses also allow for new degree options for in-residence students, such as a graduate certificate in acoustics or for students in an undergraduate minor in acoustics that is in development. Continuing education and professional development opportunities consist of specialized short courses, micro-credentials, and digital badges, which provide flexibility to non-traditional students or those seeking continuing education or professional development opportunities. This talk will discuss in detail the ongoing development of programs at UNH to address all levels of education in acoustics.

**4:00**

**1pAO5. Graduate programs in acoustics in the Physics and Oceanography Departments at the Naval Postgraduate School.** Kay L. Gemba (Physics, Naval Postgraduate School, Monterey, CA), Nicholas Durofchalk (Physics, Naval Postgraduate School, Monterey, CA), Derek Olson (Oceanography, Naval Postgraduate School, Spanagel Hall, 833 Dyer Rd., Monterey, CA 93943, dolson@nps.edu), and Oleg A. Godin (Physics, Naval Postgraduate School, Monterey, CA)

The Departments of Physics and Oceanography at the Naval Postgraduate School offer graduate programs leading to MS and PhD degrees. The acoustic track within the Department of Physics offers degrees in applied physics and engineering acoustics. Engineering acoustics degrees can be completed in either resident or distance learning modes. The department also offers stand-alone academic certificate programs in fundamentals of engineering acoustics, underwater acoustics, and sonar system applications, with a set of three certificates leading to a MS degree in engineering acoustics. MS and PhD programs are interdisciplinary, with coursework, traditional

lectures, and laboratory exercises drawn primarily from the fields of physics, electrical engineering, and fundamental acoustics. The Oceanography Department offers degrees in both Physical Oceanography and a joint degree with Meteorology that are interdisciplinary. Both oceanography degrees contain a fundamentals of ocean acoustics course which is tightly coupled to the curriculum through dispersion relations. Connections between ocean processes and detection theory are made through a course on tactical oceanography. Subjects covered in both departments include fundamentals of acoustics; the generation, propagation, and reception of sound in the ocean; acoustic signal processing; and detection fundamentals.

TUESDAY AFTERNOON, 19 NOVEMBER 2024 — 3:00 P.M. TO 4:50 P.M.

# Session 1pBA

## Biomedical Acoustics: Bubble-Based Therapies

Mitra Aliabouzar, Cochair
*Univ. of Michigan, 3218-02 Med Sci I, 1301 Cathreine St., Ann Arbor, MI 48109*

Eric Rokni, Cochair
*Acoustics, Penn State, 511 W. Kenilworth Cir., Mequon, WI 53092*

**Chair's Introduction—3:00**

### *Invited Papers*

**3:05**

**1pBA1. Comparing the delivery of free and liposomal-encapsulated doxorubicin across the blood-brain barrier with microbubble-mediated focused ultrasound.** Stecia-Marie Fletcher (Radiology, Brigham and Women's Hospital/Harvard Medical School, 221 Longwood Ave., EBRC 515b, Boston, MA 02115, sfletcher4@bwh.harvard.edu), Yongzhi Zhang (Radiology, Brigham and Women's Hospital/Harvard Medical School, Boston, MA), Ma'Kaya Mashburn (Dana Farber Cancer Inst., Boston, MA), Sofia Martinez (Dana Farber Cancer Inst., Boston, MA), Amanda Chisholm (Radiology, Brigham and Women's Hospital/Harvard Medical School, Boston, MA), and Nathan J. McDannold (Radiology, Brigham and Women's Hospital/Harvard Medical School, Boston, MA)

This work investigates the effect of drug formulation (free, small molecule drug or liposomal-encapsulated nanoparticle) on Focused Ultrasound (FUS) and microbubble-mediated delivery of doxorubicin (Dox) across the blood-brain barrier (BBB) in healthy and F98 tumor-bearing rats. Dox is auto-fluorescent (Ex: 470 nm; Em: 560 nm). 1, 4, and 24 h after FUS and the administration of Dox, brain samples were collected and homogenized, and the concentration of Dox was measured using a fluorometer. In healthy brains, at 1, 4 and 24 h, FUS significantly increased the delivery of both formulations ($p<0.05$). At 1 h, the mean concentration of Dox was 47% more in the striatum ($p = 0.010$) and 51% more in the hippocampus (n.s.; $p = 0.072$) compared to Lipo Dox. No differences in Dox versus Lipo Dox delivery were observed at 4 and 24 h. In F98 tumors, at 1 and 24 h, FUS significantly increased the concentrations of both formulations ($p<0.05$). No significant difference was observed in the concentrations of Dox and Lipo Dox at 1 h ($p = 0.291$). However, at 24 h, 96% higher drug concentrations were measured in tumors in animals that received Lipo Dox compared to Dox ($p = 0.045$). The results of this study may have important implications for drug selection in clinical applications of FUS-mediated BBB opening.

**3:10**

**1pBA2. Harnessing bioeffects of ultrasound triggered microbubble cavitation in medicine.** Lauren J. Delaney (Radiology, Thomas Jefferson Univ., 132 S 10th St., Main 763K, Philadelphia, PA 19107, lauren.delaney@jefferson.edu)

The use of gas-filled microbubbles (1-10 mm in diameter) for ultrasound imaging is well established. Such contrast agents are used worldwide to improve the diagnostic capabilities of contrast enhanced ultrasound imaging (CEUS). At higher acoustic pressures (albeit well within United States Food and Drug Administration limits), the microbubbles undergo stable cavitation followed by destruction via gas diffusion and inertial cavitation, which produces numerous, localized bioeffects. Among these bioeffects, microbubble cavitation can improve drug delivery and sensitize solid tumors to radiation. This talk will discuss the variety of applications in which our laboratory investigates these bioeffects. Our pre-clinical and clinical trials to investigate the safety and therapeutic benefits of combining radioembolization and ultrasound-triggered microbubble destruction (UTMD) in patients with hepatocellular carcinoma (HCC) as well as the possibility of improving antibiotic treatment of joint infections with UTMD will be discussed. The current status of our ongoing trial of

sonoporation in patients with pancreatic ductal adenocarcinoma (PDAC) will be presented. Moreover, we have developed a noninvasive pressure measurement technique, known as subharmonic-aided pressure estimation (SHAPE), which was used to acquire intra-tumoral pressure measurements. The use of SHAPE and other quantitative ultrasound techniques as biomarkers for predicting treatment response will also be presented.

## *Contributed Papers*

**3:15**

**1pBA3. Ultrasound-responsive collagen scaffolds for tissue engineering.** Veronica Lucian (Inst. of Biomed. Eng., Univ. of Oxford, Inst. of Biomed. Eng., Marcela Botnar Wing, Oxford OX3 7LD, United Kingdom, veronica.lucian@eng.ox.ac.uk), Brian Lyons, Michael Gray, Constantin Coussios, and Malavika Nair (Inst. of Biomed. Eng., Univ. of Oxford, Oxford, United Kingdom)

Homogeneous nutrient distribution throughout three-dimensional (3-D) scaffolds remains a key challenge in tissue engineering. The buildup of cells on scaffold edges and rapid nutrient uptake along the periphery often cause large regions of the centre to be left unoccupied by cells. Microstreaming associated with acoustic cavitation has been exploited to enhance mass transport in oncological drug delivery and transdermal vaccination, making it an attractive mechanism for promoting nutrient and oxygen distribution in tissue engineering scaffolds. In this work, we seek to use protein cavitation nuclei to synthesize ultrasound-responsive collagen scaffolds. Cavitation nuclei were embedded into the scaffold during fabrication and then exposed to 0.5 MHz focused ultrasound at peak negative pressures ranging from 0.5 to 2.7 MPa to induce inertial cavitation. Acoustic data was collected using passive cavitation detection (PCD) and post processed to isolate harmonics and broadband emissions. We compare the benefits and disadvantages of including cavitation nuclei in the scaffold fabrication process versus adding them to surrounding media during ultrasound exposure, discussing potential use of embedded protein nuclei to induce cell migration and differentiation. Additionally, we examine the effects of cavitation, exposure time, and peak negative pressure on the microstructure of collagen scaffolds, namely, pore size, interconnectivity, and percolation diameter.

**3:20**

**1pBA4. Ultrasound-guided activation of the stimulator of interferon genes pathway for cancer immunotherapy.** Sina Khorsandi (Radiology, UT Southwestern Medical Center, Dallas, TX), Yifan Wang (Radiation Oncology, MD Anderson Cancer Center, Houston, TX), Kristin Huntoon (Neurosurgery, Emory School of Med., Atlanta, GA), Adam Woodward, Connor Endlsey and Nazia Hafeez (Radiology, UT Southwestern Medical Ctr., Dallas, TX), Wen Jiang (Radiation Oncology, MD Anderson Cancer Ctr., Houston, TX), and Jacques Lux (Radiology, UT Southwestern Medical Ctr., 5323 Harry Hines Blvd., Dallas, TX 75390-8514, jacques.lux@utsouthwestern.edu)

Recent studies have shown that the activation of the innate immune sensor cyclic GMP–AMP synthase–stimulator of interferon genes (cGAS-STING) promotes antigen presentation and T-cell activation, thus transforming cold tumors to immunologically responsive tumors. However, because STING is a cytosolic protein and a key mediator of inflammation, activation of STING needs to be specific to antigen-presenting cells (APCs) in tumor tissues. To address this technical challenge and meet the clinical need, we developed a technology termed MUSIC (Microbubble-assisted Ultrasound-guided Immunotherapy of Cancer) that utilizes gas-filled microbubbles (MBs) conjugated with APC-targeting antibodies, and loaded with the STING activator cGAMP. Our results show that, upon exposure to US, MUSIC produces robust STING activation and type I interferon responses in APCs and more efficiently primes antigen-specific CD4+ and CD8+ T cells *in vitro*. These immune stimulatory effects of MUSIC directly translated into antitumor responses *in vivo*, where we showed that the MUSIC was able to generate antitumor effects against syngeneic orthotopic primary (EO771) and metastatic (4T1) murine breast cancer models as well as against two syngeneic murine melanoma models (B16-F10 and D4M). We also confirmed the establishment of systemic immune memory following MUSIC treatments as mice rejected tumor cells upon re-challenge.

**3:25**

**1pBA5. Effect of shell on the stability, radial dynamics, and acoustic characteristics of phase-shift droplets.** Sugandha Chaudhary (Univ. of Michigan, Ann Arbor, MI, sugandhachaudhary93@gmail.com), Bachir A. Abeid, Mario L. Fabiilli, and Mitra Aliabouzar (Univ. of Michigan, Ann Arbor, MI)

Shell-stabilized, phase-shift droplets of a perfluorocarbon liquid have been investigated for a wide range of applications. The shell plays a key role in maintaining droplet stability and impacting the acoustic responses of generated bubbles via acoustic droplet vaporization (ADV). While shell behavior under ultrasound exposure is well-studied with contrast microbubbles, it is not fully explored for phase-shift droplets during and post ADV. By integrating ultra-high-speed microscopy and time-lapse confocal imaging, we investigated the radial dynamics during ADV and the growth rate post-ADV for monodisperse, micron-sized perfluorohexane droplets with three different shells: lipid, polymer, and albumin. Droplet size distribution was not significantly different for each formulation for up to three weeks. The expansion ratio during ADV was significantly lower ($\sim$26%) for lipid-coated droplets compared to the other shells under similar acoustic parameters (2.5 MHz and 5 MPa). The growth rate post-ADV, recorded up to 2 min, showed an initial fast rate constant followed by a slow rate constant for all ADV-bubbles, which were significantly different among the droplets, highlighting the effect of shell. We will also discuss the acoustic characteristics of droplets with varying shell formulation. The findings will help optimize shell composition, benefiting both diagnostic and therapeutic applications of ADV.

**3:30**

**1pBA6. An *ex vivo* investigation on the potential of catheter-assisted pulsed focused ultrasound ablation for arterial plaques.** Abhirup Samaddar (Dept. of Mech. Eng., Univ. of Kansas, 1530, West 15th St., Learned Hall, 5114, Lawrence, KS 66045, abhirupmech@ku.edu), Rohit Singh (Dept. of Mech. Eng., Univ. of Kansas, Lawrence, KS), Koji C Ebersole (Dept. of Neurosurgery, Univ. of Kansas Medical Ctr., Kansas City, KS), M.Laird Forrest (Pharmaceutical Chem., Univ. of Kansas, Lawrence, KS), and Xinmai Yang (Dept. of Mech. Eng., Univ. of Kansas, Lawrence, KS)

Existing medications and interventional procedures for removing arterial plaques might cause vessel injuries or hemorrhages, leading to post-treatment complications. To reduce these risk factors, this study developed a minimally invasive technique by combining focused ultrasound (FUS) with an acoustically reflective hard surface. FUS with 500 kHz frequency was focused on the interface between a Nitinol catheter and soft tissue. Cavitation threshold was determined by monitoring the peak amplitude values of the broadband signals in experiments with agar phantoms and pork belly fat samples using a 10 MHz passive cavitation detector. It was observed that an early onset of cavitation event occurred when FUS was targeted at the tissue-catheter interface. Treatment was performed on pork belly fat and carotid artery plaque samples attached with and without a 3 mm diameter catheter. FUS bursts of 2500-cycle and 10% duty-cycle with peak-negative-pressure of 2 and 4 MPa were applied on pork belly fat and plaque samples, respectively, for 3 min. Catheter-assisted FUS treatment resulted in the enhanced fragmentation of lipid contents from pork belly and plaque samples. In contrast, FUS-only treatment showed negligible amount of plaque depletion. This study demonstrated the potential of catheter-assisted FUS therapy for plaque ablation.

3:35

**1pBA7. Assessment of an image-guided histotripsy system for renal cell carcinoma.** Muskan Singh (Dept. of Radiology, Univ. of Chicago, 5841 South Maryland Ave., Chicago, IL 60637, Muskan.Singh@bsd.uchicago.edu), George R. Schade (Urology, Univ. of Washington, Seattle, WA), Himanshu Shekhar (Electrical Engineering, Indian Inst. of Technol. Gandhinagar, Gandhinagar, Gujarat, India), Adam D. Maxwell (Biomedical Engineering and Mechanics, Virginia Polytechnic Insti. and State Univ., Blacksburg, VA), and Kenneth B. Bader (Radiology, Univ. of Chicago, Chicago, IL)

Renal cell carcinoma cancer (RCC) is the sixth most common cancer in the United States. Although surgery remains the primary choice for the initial treatment, the number of patients who qualify decreases annually due to an aging population with multiple comorbidities. Histotripsy, a non-thermal focused ultrasound technique that ablates tissue through the generation of bubble clouds, has emerged as a promising non-invasive treatment for RCC. In this study, an image-guided histotripsy system for targeting RCC was build and characterized. A focused histotripsy transducer was developed with an elliptical geometry (12/9.6-cm major/minor dimensions), 8-cm focal length, and 1-MHz fundamental frequency. The acoustic field of the transducer was assessed in the linear and nonlinear regimes. Histotripsy pulses with a 10-cycle duration were capable of generating bubbles for peak negative pressures greater than 20 MPa. The transducer was equipped with a port for a curvilinear imaging probe to facilitate therapy guidance. Bubble detection was performed in a scattering phantom with ultrafast imaging. An image processing pipeline was developed that achieved a bubble-to-tissue ratio greater than 19 dB. Overall, this system has the capacity to serve as a prototype tool for both preclinical and clinical investigations into histotripsy treatment of RCC.

**3:40–3:55 Discussion**

## *Invited Papers*

3:55

**1pBA8. Abstract withdrawn.**

4:00

**1pBA9. A comprehensive numerical investigation of the potential influence of the bubble size and ultrasound focal pressure in drug delivery enhancement.** Amin Jafarisojahrood (Toronto Metropolitan Univ., 77 Harbour Square, Apt. 2103, Toronto, Nova Scotia M5J 2S2, Canada, amin.jafarisojahrood@ryerson.ca)

Stable Bubble oscillations contribute to a range of bio-effects such as enhanced drug delivery and blood-brain barrier opening. The treatment outcome depends on the bubble oscillation characteristics and the number of bubbles present at the target. Higher concentrations result in more bubbles per unit length of the vessels and, therefore, more uniform effects. At higher bubble concentrations, however, the pre-focal attenuation increases. Moreover, above a concentration threshold, bubble-bubble interactions suppress bubble oscillations. To enhance the treatment outcome, thus, not only the bubble concentration and activity should be optimized, but also the problem of pre-focal attenuation should be tackled. Numerical results show that, using the pressure gradient of focused ultrasound transducers and by taking advantage of the pressure dependent attenuation of size isolated bubbles, pre-focal attenuation can be minimized. The optimal bubble size for maximum propagation depends on the focal pressure. When volume is matched, and at lower pressures, bigger bubbles exhibit stronger radial oscillations, scattered pressure, and microstreaming (RSM). Above a pressure threshold that depend on the bubble size, smaller bubbles exhibit stronger RSM compared to their bigger counterparts. The treatment outcome may be enhanced using an optimal set of size and pressures. These results are in qualitative agreement with experimental case studies using size isolated bubbles.

4:05

**1pBA10. Evaluation of nonlinear imaging strategies for histotripsy monitoring.** Himanshu Shekhar (Elec. Eng., Indian Inst. of Technol. Gandhinagar, AB6/207, IIT Gandhinagar Campus, Near Palaj, Gandhinagar, Gujarat 382355, India, himanshu.shekhar@iitgn.ac.in), Vishwas Trivedi (Elec. Eng., Indian Inst. of Technol. Gandhinagar, Gandhinagar, India), Emily Wallach, Katia Flores Basterrechea, and Kenneth B. Bader (Radiology, Univ. of Chicago, Chicago, IL)

Real-time monitoring is necessary for histotripsy to ensure accurate targeting of tumors. Delineating the bubble cloud can be challenging when the tumors are deep-seated. In this talk, I will discuss collaborative work on harnessing nonlinear signal processing to visualize the histotripsy-induced bubble cloud. Specifically, we employed chirp-coded excitation, Volterra filtering, and the subharmonic component of the scattered signal to visualize the bubble cloud relative to the background. Histotripsy exposures were performed on agarose and red-blood cell-doped phantoms. The bubble cloud was visualized at ultrafast frame rates and tracked over time. Our results showed the ability of the nonlinear imaging approach to improve bubble cloud visualization relative to the background. Further, the detected cloud sizes were consistent with the treatment zone. These findings could enable the use of histotripsy for patients where the bubble cloud is not clearly visible in standard B-mode ultrasound images.

4:10

**1pBA11. Histotripsy dose selection influences immune response and tumor-free survival in an orthotopic liver tumor model.** Tejaswi Worlikar (Univ. of Michigan, 2200 Bonisteel Boulevard, Ann Arbor, MI 48109, wtejaswi@umich.edu), Hanna Kim, Timothy L. Hall, Clifford S. Cho, Mishal Mendiratta-Lala, Zhen Xu, and Man Zhang (University of Michigan, Ann Arbor, MI)

Histotripsy has recently gained FDA approval for the treatment of liver cancer. However, there is limited clinical understanding of how different histotripsy doses can generate varying levels of anti-tumor response, thereby impacting patient survival. This work evaluates the impact of histotripsy dosing on immune response and tumor-free survival outcomes in an orthotopic rodent liver model. Orthotopic McA-RH7777 liver tumors generated in immunocompetent Sprague Dawley rats were ablated using different histotripsy doses: dose 1 (50Hz PRF, 50 pulses per location[pp]), dose 2 (100Hz PRF, 100 ppl), and dose 3 (200Hz PRF, 200 ppl). Control animals

received no treatment. In the survival cohort, animals were survived until 12 weeks or when tumors reached 2.5 cm. In the immune cohort, immunohistochemistry staining was performed on day 7 post-histotripsy. Complete homogenization of the targeted region was observed on acute histology. Doses 1 and 2 had more favorable survival outcomes and immune response compared to dose 3. There was significant difference in survival outcomes of histotripsy-treated animals at different doses vs. untreated controls (log-rank test, $p < 0.001$). These trends suggest that higher doses may not translate to improved therapeutic outcomes after complete homogenization of the target region is achieved. These results demonstrate the importance of selecting optimal histotripsy doses for improving treatment outcomes and will have an immediate clinical impact.

## *Contributed Papers*

**4:15**

**1pBA12. Multiparametric ultrasound image-derived analysis for monitoring breast cancer response to chemotherapy.** Swapnil Dolui (Biomedical Eng., Texas A&M Univ., College Station, TX), Basak Dogan (Univ. of Texas Southwestern Medical Ctr., Dallas, TX), Corinne E. Wessner (Thomas Jefferson Univ., Philadelphia, PA), Jessica Porembka (Univ. of Texas Southwestern Med. Ctr., Dallas, TX), Priscilla Machado (Thomas Jefferson Univ., Philadelphia, PA), Berat Bersu Ozcan (Univ. of Texas Southwestern Med. Ctr., Dallas, TX), Nisha Unni (Univ. of Texas Southwestern Med. Ctr., Dallas, TX), Maysa Khalaf, Flemming Forsberg, Kibo Nam (Thomas Jefferson Univ., Philadelphia, PA), and Kenneth Hoyt (Biomedical Eng., Texas A&M Univ., 800 W Campbell Rd. BSB 13.929, Richardson, TX 75080, kenneth.hoyt@utdallas.edu)

A major challenge in monitoring neoadjuvant chemotherapy (NAC) is how to detect and accurately quantify tumor response at an early stage of therapy. Based on the principle of ultrasound (US) scattering from different classes of tissue, the H-scan analysis has evolved. Preclinical results have demonstrated the ability of H-scan US imaging to detect early tumor response to chemotherapy in animal models. In this research, we compared the role of both B-scan and H-scan US imaging integrated with texture analysis, envelope statistics, and tumor shape markers for assessing breast cancer response to NAC in human subjects. Twenty-two patients underwent US scans before and after 10 % NAC using a clinical US scanner equipped with a 9L-D transducer. After data acquisition, a series of Gaussian filters were employed on the US data, to determine a close match to the localized tissue scattering function, and then the matched filter order were color-coded to form H-scan US images. The filters were centered equally between 3 and 9 MHz. Other parameters, such as Nakagami, texture, and tumor convexity features, were also assessed. All possible combinations of US parameters underwent dimensionality reduction using principal component analysis and then first component was used for predictive analysis through support vector machines, and an area under the curve (AUC) was reported. The data classification indicated that using optimized US combination yielded an higher AUC when H-scan US images utilized rather than B-scan US images (0.87 vs. 0.76).

**4:20**

**1pBA13. Effective distinction of lipid microbubbles in pediatric bone phantom using a machine learning approach.** Mark Farid (Bioengineering, Univ. of Texas at Dallas, 800 West Campbell Rd, Richardson, TX, mark.farid@utdallas.edu), Katherine Brown (Bioengineering, Univ. of Texas at Dallas, Richardson, TX), Michaelann Tartis (Chem. Eng., New Mexico Inst. of Mining and Technol., Socorro, NM), Sonia Hernandez (Univ. of Chicago Medical Ctr., Chicago, IL), and Shashank Sirsi (Bioengineering, Univ. of Texas at Dallas, Richardson, TX)

This study investigates the ability to detect microbubble attenuation signals across a bone phantom (meant to mimic a pediatric femur) using ultrasound and machine learning. Lipid microbubbles were fabricated and injected through a hollow capillary fiber (200 $\mu$m) in a water tank attached to a syringe pump. Focused ultrasound (1.1 MHz, 1–10 MPa, 500 pulses) was applied to the flowing bubbles in the presence or absence of a custom 3-D-printed bone phantom. A co-aligned passive cavitation detector (PCD) captured the reflected ultrasound signals. Ultrasound signals were compared using Fast Fourier Transform (FFT) to process the data. Eleven spectral features were extracted and used as the basis for machine learning algorithms. The results confirm that despite strong signal interference from the bone, machine learning can effectively distinguish the presence of lipid microbubbles. The study signifies the potential of lipid microbubbles for targeted drug delivery to bone marrow using PCD, a method often deemed challenging due to the hard barrier posed by bone structures. This method has important implications for medical treatments such as chemotherapy or gene therapy, which often struggle to reach the bone marrow due to its protective nature.

**4:25**

**1pBA14. Miniature histotripsy device to treat human pathologies.** Connor Centner (Med. & Bioeng., Univ. of Louisville, 580 S Preston St., Louisville, KY 40202, connor.centner@louisville.edu), Matthew Mallay, Jeremy Brown (Biomedical and Electrical, Dalhousie Univ., Halifax, Nova Scotia, Canada), and Jonathan A. Kopechek (Bioengineering, Univ. of Louisville, Louisville, KY)

A high-frequency (6 MHz), miniature histotripsy device was developed to treat human pathologies using bubble-based focused ultrasound therapy to mechanically ablate tissue. The goal of this study was to determine the effect of pulses per treatment point on bubble formation and subsequent tissue ablation area using tissue-mimicking phantoms. These phantoms were designed to closely simulate the mechanical and acoustic properties of human tissues. Fibrinogen and thrombin, along with 5% fetal bovine serum, were added to form a fibrin gel incubated at 37 °C for 1 h to ensure fibrin formation. Subsequently, human MC38 cells were co-cultured with the fibrin gel for at least 1 day prior to treatment. Histotripsy pulses above the cavitation threshold were applied ranging from 500 to 5000 pulses. Bubble formation and dynamics were monitored in real-time using ultrasound imaging to detect the extent of ablation and was compared to *post hoc* optic imaging. The results indicated that the number of pulses applied, and ultrasound pressure influenced bubble dynamics, which in turn affected the ablation area. Increased number of pulses applied was correlated with larger ablation areas, highlighting the importance of optimizing bubble detection under real-time imaging to maximize histotripsy treatment efficacy.

**4:30**

**1pBA15. Feasibility study of histotripsy with the clinical device FocalOne®.** Virgil Accary (LabTAU - INSERM U1032, 151 Cours Albert Thomas, Lyon, Rhône 69003, France, virgil.accary@inserm.fr), Thomas Payen (Innovation, EDAP TMS, Vaulx en Velin, Rhône, France), Maxime Lafond (LabTAU - INSERM U1032, Lyon, France), Jérémy Vincenot (Innovation, EDAP TMS, Vaulx en Velin, Rhône, France), and Cyril Lafon (LabTAU - INSERM U1032, Lyon, France)

This study investigates the feasibility of generating histotripsy lesions in biological tissues *ex vivo* with the HIFU clinical device FocalOne® (EDAP-TMS). The Focal One® probe, composed of a 192-element imaging array and a 16-element therapy transducer operating at 3 MHz, was used with an upgraded amplifier (ADECE, 500W, 200 kHz-6MHz ) and a Verasonics Vantage system for passive cavitation and elastography monitoring. To ensure reproducibility, efficiency, and safety, efforts were directed into the electrical hardware stability and power optimization. Pressure fields, nonlinear simulations, radiation force balance, and electrical instant power measurements were performed to validate that this setup can produce boiling histotripsy which is characterized in literature by a 10–18 MPa negative

pressure and a shock amplitude >70 MPa. Typical treatment exposure was 10 ms repeated 60 times at 1 Hz pulse repetition frequency. Experiments in gel phantoms and fresh degassed bovine liver were performed. The ablations were detected with ultrasound B-mode at the focal spot. The millimetric lesions were confirmed by hyposignals on 3-D MRI scans and visual observation of cavities at the targeted spot after dissection. Further studies will be required to optimize treatment parameters and improve reproducibility of histotripsy using the Focal One® probe.

**4:35–4:50 Discussion**

TUESDAY AFTERNOON, 19 NOVEMBER 2024 3:00 P.M. TO 4:05 P.M.

# Session 1pEA

## Engineering Acoustics: Engineering Acoustics Lightning Round

Michael R. Haberman, Chair
*Applied Res. Lab., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758*

**Chair's Introduction—3:00**

### *Contributed Papers*

**3:05**

**1pEA1. Tunable phase-based liquid acoustic lenses for spherical and line focused beamforming.** Sina Rostami (Physics, Univ. of Mississippi, 108 Lewis Hall, P.O. Office Box 1848, University, MS 38677, srostami@go.olemiss.edu) and Joel Mobley (Physics and Astronomy, Univ. of Mississippi, University, MS)

Phase-based tunable liquid lenses provide for adjustable beamforming in a planar configuration. This phase-based lens is distinct from refractive liquid lens designs as no shape changes are required. Previously we reported on the beamforming capabilities of lenses utilizing four channels which performed at a level comparable to quasi-planar stepped lenses (QSLs). In this talk, we present field measurements for a lens utilizing more channels and examine the effects of these additions on forming spherical and line-focused fields. Furthermore, we compare their performance with numerical simulations and discuss the potential of these systems for nearfield droplet manipulation.

**3:15**

**1pEA2. Parallel addressing of biosensing ultrasonic implants using plane wave imaging.** Anam Bhatti (LabTAU - INSERM U1032, 151 Cours Albert Thomas, Lyon, Autre 69003, France, anam.bhatti@inserm.fr), Hugo Guillot (Grenoble Inst. of Neuroscience, Grenoble, France), Ouafae Firdaouissi (LabTAU - INSERM U1032, Lyon, France), Mohammed Irar, Clément Hébert (Grenoble Inst. of Neuroscience, Grenoble, France), and Maxime Lafond (LabTAU - INSERM U1032, Lyon, France)

Ultrasonic backscatter communication allows wireless and battery-free biosensing and has been proposed to monitor various physiologic signals. These sub-mm implants serve as antennae deeply seated in tissue, and are typically addressed using external single-element transducers, preventing parallel sensing of multiple implants. Their basic principle is that their acoustic reflectivity is modulated by the electrical impedance of a transistor tuned on a specific physiologic signal. Here, we investigate plane wave imaging to simultaneously and selectively detect the modulation of multiple implants. An L11-5v ultrasound array operating at 4.4 MHz was connected to a Verasonics programmable scanner and used to observe 350-$\mu$m piezoceramic cubes connected to a variable impedance shunt load. We also observed the variation of the amplitude of the radiofrequency signals picked up by the implants and radiated acoustically (an active signal from the passive implants) after beamforming of the received signals. Accurate detection of a change in the amplitude of this emitted wave was assessed for two piezoceramic transducers separated by 5 mm in water both with a shunt load and a transistor sensitive to neural activation, and below 15 mm turkey breast. Successful detection of the signal variation validates this approach for simultaneously addressing multiple neural implants.

**3:25**

**1pEA3. Investigating commercially available force sensors for bone conduction hearing device evaluation.** Kiersten A. Reeser (Applied Research Associates, Inc., 7921 Shaffer Parkway, Littleton, CO 80127, kreeser@ara.com), Alexandria Podolski (Applied Research Associates, Inc., Littleton, CO), William Gray, Andrew D. Brown (Speech and Hearing Sciences, Univ. of Washington, Seattle, WA), and Theodore F. Argo (Applied Research Associates, Inc., Littleton, CO)

Bone conduction devices (BCDs) stimulate the auditory system by transmitting audio-frequency vibrations into the head. BCDs have long been used to diagnose and treat hearing loss, but increased interest in BCDs for consumer and communications applications has led to a proliferation of transducer designs. BCD calibration has historically been performed using an artificial mastoid, an expensive device that does not readily accommodate all BCD styles or account for impacts of anatomical variability on BCD coupling. Recent developments in force sensing technology could offer an inexpensive alternative to artificial mastoids for use with human subjects. Here, three commercial off-the-shelf (COTS) sensors were identified that met size, flexibility, and dynamic range requirements necessary for evaluating human BCD performance. Sensor accuracy and repeatability of both static and dynamic force measurements were compared against artificial mastoid measurements. A capacitance-based sensor successfully measured static force coupling but was incapable of capturing dynamic force output due to sensor bandwidth limitations. A piezoresistive sensor had a suitable bandwidth, but required significant signal conditioning to amplify signals and

improve accuracy and precision. An alternative piezoresistive sensor utilized integrated circuitry to amplify signals, increasing sensor durability and accuracy. Future research will evaluate additional low-cost solutions for BCD evaluation.

**3:35**

**1pEA4. Demonstrating wave and vibrational behavior in spatiotemporally modulated systems.** Kayla Cecil (Applied Res. Lab., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, kncecil97@utexas.edu), Benjamin M. Goldsberry, and Michael R. Haberman (Applied Res. Lab., The Univ. of Texas at Austin, Austin, TX)

Elastic materials with time and space varying properties are interesting candidates to control bulk and guided waves in unprecedented ways which include nonreciprocal transmission, unidirectional mode conversion, mode coupling, and frequency conversion. Previous work on elastic waves in spatiotemporally modulated (STM) media have focused on nonreciprocal vibrations of finite domains with modulated stiffness. Less attention has been given to the modulation of impedance boundary conditions of finite structures or at interfaces between domains. This work investigates elastic wave behavior at boundaries with modulated impedances. The boundary impedance is represented using a lumped parameter mass spring damper system with a spring having a temporally modulated stiffness. The effective input impedance is modeled using direct numerical simulation and a semi-analytical Fourier series expansion model. The models are compared to experiments on a base-excited cantilevered Euler-Bernoulli beam with modulated effective stiffness via piezoelectric patches with temporally varying electric shunting circuits. [Work supported by the ARL:UT Chester M. McKinney Graduate Fellowship in Acoustics and by the Office of Naval Research under Award N00014-23-1-2660.]

**3:45–4:05 Discussion**

TUESDAY AFTERNOON, 19 NOVEMBER 2024 — 3:00 P.M. TO 5:00 P.M.

## Session 1pED

## Education in Acoustics: What's That Sound? (Sounds of my Research)

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

Keeta Jones, Cochair
*Acoust. Society of America, 1305 Walt Whitman Rd., Ste. 300, Melville, NY 11747-4300*

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

**Chair's Introduction—3:00**

### *Contributed Papers*

**3:05**

**1pED1. The sound of culture: Exploring acoustical features of sounds perceived in free-choice learning environments.** Donnelley Hayde (Ctr. for Research and Evaluation, COSI, 333 W. Broad St., Columbus, OH 43215, dhayde@cosi.org), Nickolay Hristov (ERC, TERC, Cambridge, MA), Justin Reeves Meyer, Laura Weiss (Ctr. for Res. and Eval., COSI, Columbus, OH), Elise Levin-Güracar (ERC, TERC, Cambridge, MA), Joe Heimlich (Ctr. for Res. and Eval., COSI, Columbus, OH), Kim Kawczinski, Daniel Shanahan (Music Theory & Cognition, Northwestern Univ., Evanston, IL), and Martha Merson (ERC, TERC, Cambridge, MA)

The Sound Travels research team will share a recording that exemplifies affective associations made with specific sounds by visitors to free-choice learning environments (a science museum, a park, a zoo, and a botanical garden). This recording reflects direct collaboration with visitors and demonstrates the variation in how people make sense of sound, both in identifying its sources and in describing its effects on their emotional and cognitive states. Our US-based, federally funded project explores the impacts of ambient and designed sound on STEM learning and leisure experiences. Beyond addressing our research questions, we embrace the larger goals of seeking meaningful input from professionals in and visitors to these spaces and directly informing educational design practice. Our methods include multiple stationary ambient recordings within spaces of interest, a post-experience visitor questionnaire, and a "sound search" instrument in which visitors record video clips during their experience to represent sounds that make them feel curious, energized, uneasy, and peaceful. Together, the resulting data reveal not only how visitors are affected by sound but also how visitors experience and notice sound in context, and in what ways a person's embodied and culturally informed associations with sound relate to their experiences of learning and leisure.

3:10

**1pED2. An advanced physics laboratory based on the compound string.** Philip L. Ratner (Physics, Whitman College, 280 Boyer Ave., Walla Walla, WA 99362, ratnerp@whitman.edu) and Kurt R. Hoffman (Physics, Whitman College, Walla Walla, WA)

The compound string is a simple one dimensional system that we are developing as an advanced undergraduate laboratory experiment. Experimentally, the winding of an overwound guitar string is unwrapped to create a discontinuity of the mass per unit length. When plucked at different positions along the string, the anharmonic frequencies of the normal modes create an unusual tone. The experiments to characterize the compound string expose students to high resolution FFT measurements, careful data analysis, and a non-trivial system to practice solving differential equations.

**3:15–5:00 Impromptu Contributions**

TUESDAY AFTERNOON, 19 NOVEMBER 2024 3:00 P.M. TO 5:00 P.M.

## Session 1pMU

## Musical Acoustics: Evolving Technologies for Telematic Music Connections

Jonas Braasch, Cochair
*School of Architecture, Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180*

Samuel Chabot, Cochair
*EMPAC, Rensselaer Polytechnic Inst. 110 8th St, Troy, NY 12180*

### *Invited Papers*

3:00

**1pMU1. Telematic connection of two co-located venues using virtual microphone control.** Jonas Braasch (School of Architecture/EMPAC, Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, braasj@rpi.edu), Samuel Chabot (EMPAC, Rensselaer Polytechnic Inst., Troy, NY), and Mincong Huang (School of Architecture/EMPAC, Rensselaer Polytechnic Inst., Troy, NY)

In this project, a spatially accurate, long-distance audio connection was developed between two collaborative spaces, Rensselaer's CRAIVE-Lab and the Panorama Screen System at EMPAC. This talk focuses on the special requirements for music collaborations that result from simultaneously producing sounds at both remote ends. Unlike with the back-and-forth occurring speech, the latency between both parties has to be kept at a minimum to avoid temporal disruptions. We use a low-latency transmission software (Jacktrip) and a low-latency, time-based sound spatialization software package (Virtual Microphone Control, ViMiC) to mitigate this. ViMiC is based on the virtual simulation of traditional microphone techniques, where each virtual microphone in a multichannel array can be adjusted in directivity and 3-D orientation using delay lines and amplitude adjustments for auralization. A spherical microphone is used at each end to track the musical instruments and voices. The latency requirement for the tracking system is less strict because the additional latency only affects the spatial but not the temporal properties of the sounds. The two sites use immersive loudspeaker systems capable of accurately reproducing spatialized audio from the remote site. [Work supported by NSF IIS-1909229.]

3:20

**1pMU2. Blending physical and cyber infrastructures to enable seamless telematic and in-person presentations.** Margaret Schedel (Music/IACS, Stony Brook Univ., 1 wels Lane, Setauket- East Setauket, NY 11733, gem@schedel.net)

Connecting laptops to projectors and PAs during conferences often frustrates presenters and organizers, especially for hybrid events. Issues such as incorrect input sources, loose connections, resolution mismatches, incompatible adapters, damaged cables, outdated drivers, projector settings, HDCP incompatibility, projectors hijacking audio, and operating system discrepancies can arise. These problems are worsened by tight schedules, with back-to-back presentations involving different personal computers with unique issues. Some conferences use an on-site computer to mitigate these issues, but this approach has its own challenges. These include inconsistent presentation slide behavior (fonts, images, transitions), the absence of necessary software or plugins for interactive demonstrations, and the need to upload slides well in advance. This paper presents a system that uses standard remote conferencing software (e.g., Zoom), a simple mixer, a PA, and an Internet connection—all typically available in presentation settings—to facilitate seamless presentations without physical connections between computers and projectors. First implemented at the 2022 IRCAM Forum at NYU, this system has since been successfully deployed at the 2023 WFAE and 2024 Web Audio Conferences, proving its effectiveness in ensuring smooth and uninterrupted presentations.

**3:40**

**1pMU3. Distributed acoustical meshes on the Internet.** Chris Chafe (Music, Stanford Univ., CCRMA/Music, Stanford, CA 94305, cc@ccrma.stanford.edu)

An acoustical mesh distributed over the Internet is constructed of scattering junction nodes with bidirectional audio streaming between them. Components of common network music performance (NMP) applications like JackTrip can be adapted for experimentation with the concept. In the usual context of ensemble NMP, an audio hub server accepts bidirectional audio connections from multiple clients. The clients are located apart from each other physically and the hub server handles audio to and from the ensemble of sites, typically comprising a band or choir. The usual spoke and wheel topology (single hub server/mulitple hub clients) requires some modifications to run as a mesh of interconnected scattering juntions. The resulting waveguide mesh topology is explored for its properties, some of which resemble extensions of waveguide mesh physical model simulations of musical instruments and rooms. The acoustical medium that this creates is a live, vibrating acoustic mesh across Internet space. Unique properties, such as non-uniform distribution of nodes, are described in an experiment that explores analogies to plates and membranes and is characterized by the inherently anisotropic propagation of the network.

## *Contributed Papers*

**4:00**

**1pMU4. Integrated digital twin of networked immersive rooms for distributed interactive telepresence in shared virtual worlds.** 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)

We present an integrated digital twin for room-oriented immersive systems (ROIS) that facilitates spatially distributed multi-user remote telepresence. This work, built upon prior development of game-based virtual collocation systems in immersive rooms, further addresses synchronous communication and spatially distributed interactivity between remote participants through a distributed audio spatialization system. This digital twin allows its users in physically disjoint locations to navigate through the virtual worlds with virtual sound source information transmitted through remote procedure calls (RPC). Precisioned user location tracking data is broadcast directly through in-game sessions. For each virtual sound source, generated procedurally or through non-invasive microphone inputs, real-time reverberation is synthesized and encoded based on proximity-modulated perceptual parameters. These virtual sound sources are rendered in each ROIS facility through Open Sound Control (OSC). This approach offloads the otherwise expensive computational resources for physically based acoustic rendering techniques unsuited for networked applications. We demonstrate this by evaluating the system using latency metrics informed by round-trip-time measures. Taken as a whole, the integrated digital twin in this work also considers broader scalability so that researchers and content creators can deploy it for any immersive rooms alike in the future. [Work supported by NSF IIS-1909229 & CNS-1229391.]

**4:15**

**1pMU5. Sound art, no input mixing, and the spencer-brown modulator.** Divyamaan Sahoo (Dept. of Elec. & Comput. Eng., Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., North Dartmouth, MA 02747, divyamaansahoo@gmail.com)

This presentation pays homage to sound artists in their quest to make the inaudible audible using piezoceramic discs, inductive coil pickups, microphones, speakers, and wires, to name a few. The burgeoning practice of contemporary sound art identifies itself in feedback – the sound that creates itself. Such an approach is exemplified through practices like no input mixing, where signals from a mixer are fed back into the mixer itself, generating complex musical textures. This presentation contextualizes such sound art practices by introducing the theory of "modulators" or "reductors," re-entering NOR circuits, which manipulate oscillating inputs to create distinctive and predictable rhythmic patterns and frequencies. By tracing the origins of the square wave to the humble inverter and referencing a variety of handmade electronic music practices, this presentation challenges traditional definitions of musical acoustics and sound art, reevaluating what constitutes musical expression.

**4:30–5:00 Demonstrations and Discussion**

## Session 1pNS

# Noise: Career Paths in Noise Control

James E. Phillips, Chair
*Intertek, 4703 Tidewater Ave., Ste. E, Oakland, CA 94601*

### *Invited Papers*

**3:00**

**1pNS1. From playing in a high school band room to designing them.** Jessica S. Clements (Acoustics, SmithGroup, 999 Peachtree St. NE, Ste. 400, Atlanta, GA 30309, jessica.clements@smithgroup.com)

Jessica Clements discovered the field of acoustics as a high school senior when a square concrete box was constructed for their new band room. Watching the acousticians moving panels around the room to get the sound the band director wanted illustrated what she'd been looking for in a career – one that captured both music and math. She started her consulting career at Merck & Hill Consultants, with the very acousticians who fixed that HS band room. Over 10 years at Merck & Hill, she performed numerous field measurements and noise control investigations as well as architectural acoustics designs. In 2013, she moved to Newcomb & Boyd, LLP where she spent most of her initial time on mechanical system noise and vibration control design, eventually moving into larger noise control projects related to transportation noise, vibration control, and building acoustic occupant comfort in addition to architectural acoustic design, project management, and business development. She is currently a Principal and the Acoustics Discipline Lead for The Smith Group. This presentation will briefly discuss the trajectory of her career and a few of the key lessons she's learned.

**3:10**

**1pNS2. Sound, computers, and all the rest—A career in acoustics.** Michael Vorlaender (IHTA, RWTH Aachen Univ., Kopernikusstr. 5, Aachen 52074, Germany, mvo@akustik.rwth-aachen.de)

After studying physics and completing his doctorate in 1989 on room acoustic computer simulations, Michael Vorlaender worked from 1989 to 1996 in various areas of acoustics, such as psychoacoustics, electroacoustics and building acoustics at the PTB in Braunschweig. He moved from the position of research officer back to his academic home ("alma mater") RWTH Aachen University in 1996 and still works there today as Professor of Acoustics. His research focuses on auralization and acoustic virtual reality for perception research and application, particularly in architectural acoustics and noise control. Michael Vorlaender has been involved in several organizations as board member and president. The presentation will give a brief overview of the points at which the career steps included planned or unexpected turning points.

**3:20**

**1pNS3. Thankful for a unique career path and trying to pay it forward.** K. Anthony Hoover (McKay Conant Hoover, 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362, thoover@mchinc.com)

It is so interesting to learn how someone found their career, especially a career in acoustics. Often, as was the author's case, that path is a winding road filled with challenges and surprises. Especially unforeseen and fulfilling is the vast amount of teaching, with a number of students changing their own career paths toward grad school and consulting in acoustics.

**3:30**

**1pNS4. Experiences and mindsets along the pathway of a early-career faculty in United States.** Yangfan Liu (Purdue Univ., Ray W. Herrick Lab., 177 S, Russell St., West Lafayette, IN 47907, yangfan@purdue.edu)

Yangfan Liu received his Mechanical Engineering bachelor's degree in China in 2009 and joined Purdue University in United States for Master and PhD degrees in Acoustics and Noise Control right after that. His PhD degree was earned in 2016, he continued to work as a post-doc researcher for two years at the same lab, Ray W. Herrick Lab, at Purdue. Dr. Liu was hired as an Assistant Professor by Purdue School of Mechanical Engineering in 2018 after his Post-doc. His research was focused on sound field reconstruction and room acoustics simulation during his graduate school study, he then established many other research areas later on, such as active noise control, multi-physics simulation, sound perception, vibroacoustic signal based fault diagnosis, etc. Dr. Liu is also very active in academic organizations, such as the Acoustical Society of America (ASA) and the Institute of Noise Control Engineering (INCE) USA, with multiple serving and leading roles. In this presentation, Dr. Liu will share his experiences in his career pathway, lessons he learned as well as opinions on the key skills and mindsets for an academic career preparation and development based on his personal experiences.

**3:40–3:55 Discussion**

**3:55**

**1pNS5. From concert halls to air conditioners.** Derrick P. Knight (Trane Technologies, 824 Olympic Dr., Onalaska, WI 54650, derrick.knight@irco.com)

This talk will describe a circuitous career path in noise control, describing the journey from concert halls to air conditioners through consulting, teaching, self-employment, and engineering positions held over the past 24 years.

**4:05**

**1pNS6. Noise control at the receiver: Promoting hearing conservation in society.** William J. Murphy (Stephenson and Stephenson, Research and Consulting, 5706 State Rte. 132, Batavia, OH 45103, wmurphy@sasrac.com)

Noise Control Engineering traditionally employs a Source/Path/Receiver model to mitigate the generation of noise in all sorts of activities. The industrial hygiene hierarchy of controls seeks to eliminate noisy processes, to identify substitute processes, or to develop engineering solutions that reduce noise at the Source or along the Path. Administrative controls and personal protective equipment (PPE) are the last resort to reduce noise at the receiver, our ears. The first three decades of my career were as an officer of the Commissioned Corps of the United States Public Health Service and I was assigned to the Centers for Disease Control, National Institute for Occupational Safety and Health. My research focused on developing rating and testing methods for hearing protection devices, standards to evaluate occupational noise exposures, and methods to measure impulse noise exposures. This talk will describe how I transitioned from graduate student in physics to a noise researcher focused on the protection of the receiver's ears.

**4:15**

**1pNS7. Making rocks sing: A career path.** Evelyn Way (Maxxon Corp., 920 Hamel Rd., Hamel, MN 55340, eway@maxxon.com)

From voice student to acoustics consultant to running an acoustics lab, it's been an eventful career. The internal and external forces that drive a career will be discussed, as well as the joys and challenges of different paths.

**4:25**

**1pNS8. A nonlinear career path for acoustic metamaterial research.** Christina Naify (Applied Research Lab., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, christina.naify@arlut.utexas.edu)

Acoustic metamaterials are designed materials which have specific functionality to control acoustic or elastic waves. Over the last 20 or so years, metamaterials have proven to be a promising approach for noise control such as acoustic absorption control of acoustic fields such as holography. My technical research has focused on acoustic metamaterials for over 15 years and that work has taken me on a very nonlinear path through 4 institutions over the course of my career. When I was leaving school, like many students or those early in their career in research or industry, I expected that my professional path would be fairly linear but the reality has taken unexpected twists and turns. In this talk, I will describe my own career path researching acoustic metamaterials. This path has included multiple cross-country moves with noise control using acoustic metamaterials as a common theme.

**4:35**

**1pNS9. Planes, rockets, trains, and buildings—A career that included them all, and more.** James E. Phillips (Intertek, 4703 Tidewater Ave., Ste. E, Oakland, CA 94601, james.phillips@intertek.com)

James Phillips started his career conducting aircraft cabin acoustics measurements in the laboratory and in-flight at McDonnell Douglas while completing his Master of Science in Acoustics degree from the Pennsylvania State University in 1989. From there, he worked briefly at the Jet Propulsion Laboratory before moving on to the Aerospace Corporation where he analyzed launch vehicle and payload acoustic, vibration, and shock environments for the then United States Air Force Space and Missile Systems Center. After that, he changed his career path to work for an acoustical consulting firm where he spent much of his time analyzing sound and ground-borne vibration from rail transit vehicles, as well as many other noise and vibration related issues in other fields. He currently works in the Intertek Building Science Solutions - Noise, Acoustics, and Vibration group measuring and analyzing noise and vibration within buildings and propagated to communities. This presentation will briefly discuss how Mr. Phillips used the knowledge gained from each step in his career to move on to the next.

**4:45–5:00 Discussion**

# Session 1pPA

## Physical Acoustics: Current Trends in Physical Acoustics

Christopher M. Kube, Chair
*Eng. Sci. and Mech., The Pennsylvania State Univ., 212 Earth and Engineering Sciences Bldg., University Park, PA 16802*

**Chair's Introduction—3:00**

### *Contributed Papers*

**3:05**

**1pPA1. Pressure wave propagation in high viscous magma containing crystals and bubbles.** Daichi Kurata (Graduate School, Univ. of Tsukuba, 1-1-1 Tennodai, Tsukuba 305-8573, Japan, krt-daichi@outlook.jp), Takahiro Ayukai (Graduate School, Univ. of Tsukuba, Tsukuba, Japan), and Tetsuya Kanagawa (Eng. Mech. and Energy, Univ. of Tsukuba, Tsukuba, Ibaraki, Japan)

The presence of bubbles and crystals in highly viscous liquid (e.g., glass, magma) has been important for the purpose of both basic and practical viewpoints. In this presentation, especially, we focus on a magma, and investigate theoretically P-wave (i.e., pressure wave) propagation in the magma containing many bubbles. Based on an averaged theory of dispersed multiphase flow, we can regard the mixture of crystals (i.e., solid phase) and melt (i.e., liquid phase) as a single liquid phase, i.e., single non-Newtonian liquid, which is approximated to a Newtonian liquid by using the concept of an effective viscosity composed of various parameters (especially crystals). The system of basic equations for multiphase flow containing the effective viscosity, can be reduced to the KdV-Burgers equation as a nonlinear evolution equation via theoretical approximation based on singular perturbation method. As a result, the effect of fraction of crystals and bubbles on the acoustic characteristics (e.g., waveform evolution, nonlinearity, attenuation) is discussed.

**3:15**

**1pPA2. Acoustically driven rapid flow through microporous media.** Sujith Jayakumar (Mechanical and Aerospace Engineering, Univ. of California San Diego, 3869 Miramar St., PO 4232, La Jolla, CA 92092, sujayakumar@ucsd.edu), Ofer Manor (Chemical Eng., Technion - Israel Inst. of Technol., Haifa, Israel), and James Friend (Mech. and Aerospace Eng., Univ. of California San Diego, La Jolla, CA)

Porous media provides numerous useful capabilities, including absorption, filtration, separation, and large surface areas for reactions and diagnostics. Acoustofluidic techniques have been tentatively explored to enhance flow in such media, but the majority of such work has problems, including flows that bypass the medium, irregular pores, slow or opposing flows, and non-directional random flow. Of particular interest is the porous material's significant acoustic losses; therefore, attenuation is greater when pore dimensions are small and non-directional. In this work, we drive fluids through randomly oriented pores as small as 12 micrometers. Acoustic streaming phenomena generated by the attenuation of traveling surface acoustic waves in the medium are used to pump the fluid. The media is carefully laminated with the sides sealed to facilitate flow solely through the media. The flow rates generated by a unique floating electrode unidirectional transducer (FEUDT) are compared with a typical interdigital transducer. Because the FEUDT is specifically designed to continuously generate unidirectional acoustic waves, we show that the flow is 3 times faster than the diffusion rate. Furthermore, we reveal new evidence of homogeneous mixing caused by acoustic streaming inside the porous media.

**3:25**

**1pPA3. Linear analysis of sound velocity and attenuation in bubbly liquids.** Akihiro Nakamura (Dept. of Eng. Mech. and Energy, Univ. of Tsukuba, 1-1-1, Tennodai, Tsukuba, Ibaraki 305-8573, Japan, s2320876@u.tsukuba.ac.jp) and Tetsuya Kanagawa (Eng. Mech. and Energy, Univ. of Tsukuba, Tsukuba, Ibaraki, Japan)

Owing to a dispersion effect induced by bubble oscillations, the change of phase velocity and attenuation of waves in bubbly liquids is important. Based on a set of volumetric averaged equations in a two-fluid model with bubble dynamics equation and temperature gradient models, we theoretically investigate linear propagation of pressure (or ultrasonic) waves in mono- and poly-disperse bubbly liquids. By incorporating the dissipation effects, the stop band (i.e., frequency range where the waves cannot propagate) appears a moderate high frequency region in a linear dispersion relation. From the comparison of sound velocity and attenuation with previous experimental data, the effect of acoustic radiation and thermal conduction is discussed. We found that, when using a two-fluid model, even if dissipation effects are considered, the inconvenience of infinite divergence of wave numbers at resonant frequency cannot be solved. However, based on the classical dispersion relation, we discuss how to solve this problem. Furthermore, thermal conduction and acoustic radiation should be appropriately set up to accurately predict the sound velocity and attenuation except in the high-frequency region, the sound velocity in the resonant frequency region, and the attenuation in the high-frequency region.

**3:35**

**1pPA4. Theoretical comparison of some models for thermal process inside oscillating multiple bubbles.** Reon Matsui (College of engineering systems, University of Tsukuba, Tennodai, 1-1-1, Tsukuba, Ibaraki 3058577, Japan, matsui.reon00@gmail.com), Quoc N. Nguyen (Dept. of Eng. Mech. and Energy, Univ. of Tsukuba, Tsukuba, Ibaraki, Japan), and Tetsuya Kanagawa (Eng, Mech. and Energy, Univ. of Tsukuba, Tsukuba, Ibaraki, Japan)

Several models have been proposed to describe thermal processes at gas pressure inside bubble. Recently, the comparison of three typical models was performed for the case of single bubble (Sojahrood *et al.*, 2020): Model A approximates the effect of thermal damping by adding an artificial viscosity to the viscosity of liquid surrounding the bubble and is widely used for the case of shell-coated bubbles. Model B is a model that includes the average temperature rise within the bubble and the heat loss at the bubble boundary; this makes it more accurate by taking into account the pressure dependence that was ignored in Model A. Futhermore, Model C is based on the assumption of polytropic process. In this study, we extend the findings of Models A, B, and C from the case of a single bubble to the case of multiple bubbles (i.e., bubbly flow). The nonlinear propagation of ultrasound in bubbly flow is theoretically investigated based on singular perturbation method up to a second order of approximation. As a result, a nonlinear evolution equation is obtained; the differences among Models A, B, and C are

summarized in the coefficients. The details of the differences will be explained in a presentation.

**3:45**

**1pPA5. Fluid flow measurements in nanoslits using holographic microscopy.** Siyang Yu (Mech. and Aerospace Eng., Univ. of California San Diego, 3535 Lebon Drive, San Diego, CA 92122, siy011@ucsd.edu) and James Friend (Mech. and Aerospace Eng., Univ. of California San Diego, La Jolla, CA)

To understand the mechanisms driving fluid flow behavior in nanofluidics so that they may be used for on-chip biomedical and chemical applications, the fluid's motion itself needs to be observable and measurable, a difficult challenge at these small scales. We present a new method for measuring both slow and fast flows in nanofluidics using high-speed digital holographic microscopy. We measure the evaporation-driven flow in 25- and 7-nm tall nanoslit channels, showing that the consequent flow speed is about 15 times slower than open atmospheric evaporation due to the confinement of the nanoslit channel. We also measured the surface acoustic wave-driven flow in the 25-nm channel, showing flow at a speed of 0.12∼m/s from acoustic wave propagation at 39.7 MHz interacting with the fluid in the channel. A process to eliminate the many sources of noise to produce these results are provided, showing that—in particular—spatial averaging is useful to determine the fluid flow and the dewetting of the fluid in the nanoslit channel over time.

**3:55**

**1pPA6. Acoustic streaming.** James Friend (Mech. and Aerospace Eng., Univ. of California San Diego, 9500 Gilman Dr. MC0411, MADLab SME344K, La Jolla, CA 92093, jfriend@ucsd.edu)

In recent years, the need for accurate analysis of acoustic streaming has become urgent, as classical methods fail to quantify flows generated by acoustic waves beyond a few hundred kHz. This problem, identified since Lighthill's seminal 1978 namesake publication, remains unresolved. Researchers, including myself, have relied on computational analysis and approximations to address some of these challenges. Few have examined the dynamic behavior of acoustic streaming—such as delayed onset, response to inharmonic excitation, and flows sustained after the acoustic wave stops—despite their importance in experiments. The difficulty lies in analyzing these flows. However, after extensive work on an alternative approach, we now have a method for closed-form analysis of one-dimensional acoustic streaming that includes transient behavior and accounts for nonlinear compressibility and viscous effects. We offer a coherent and straightforward plan for analyzing simple acoustic streaming cases where classical theories fail. This method can model both steady-state and transient acoustic streaming, potentially eliminating the need for computational analysis and providing enough physical insight for design purposes. Our goal is to equip listeners with the tools to perform this analysis themselves without need for tedious computational methods.

**4:05**

**1pPA7. Combined effect of acoustic radiation force and acoustic streaming for focused beams to trap cells in three dimensions.** Shiyu Li (Shanghai Jiao Tong Univ., Dongchuan Rd. 800, Shanghai 200240, China, shiyu.li@sjtu.edu.cn) and Zhixiong Gong (Shanghai Jiao Tong Univ., Shanghai, China)

Selectively trapping of a single cell has potential applications such as reproductive cell selection, cell mechanics measurement, *in-vivo* rheology probes, etc. A widely-used technique called the micropipette aspiration has been used [Mitchison and Swann, J. Exp. Biol.,31, 443 (1954).] However, this contact trapping of a cell suffers the disadvantages of possible biological cross-infection and potential damage under mechanical suction. This could be overcome by the single focused-beam acoustical tweezers, as shown with a first trial in theory [Gong *et al*., J. Acous. Soc. Am. 154, A263 (2023)]. To selectively trap a cell in three dimensions, the beam's wavelength should approximate the typical size of cells, leading to a high frequency up to a few tens of megahertz. However, the acoustic absorption at such frequencies will be significant, and the acoustic streaming induced drag force [Li *et al*., Phys. Rev. Fluids (in press)] cannot be ignored. Hence, the 3-D trapping by a focused beam under the acoustic radiation force (ARF) as demonstrated previously may fail since the axial trapping is fragile. Here, we study the ability of 3-D trapping under the resultant force of the ARF and streaming-induced drag force and summarize the prior parameters

**4:15**

**1pPA8. Creating logic gates using actuated phononic crystals.** Yuanyan Zhao (Informatics, Univ. of Sussex, Sussex House, Falmer, Brighton BN1 9RH, United Kingdom, yz467@sussex.ac.uk), Sriram Subramanian (Univ. College London, London, United Kingdom), and Gianluca Memoli (Univ. of Sussex, Brighton, United Kingdom)

In recent times, the use of acoustic metamaterials has opened to acoustics many technical developments that were initially associated only with light. In this work, we discuss how this trend brings naturally to the developing field of acoustic computing. In particular, we show how multi-level logic gates can be created using mechanically-actuated phononic crystals. We show how to design unit cells that exploit topological symmetries (or the lack of them) to create basic ON/OFF switches. We demonstrate how these switches can be combined into the main logic gates (AND, OR, NOT, and XOR) and how these might be used in practice. Finally, we realise two of these devices, based on audible sounds, and use ad-hoc measurement methods to assess their performance compared to theory.

**4:25**

**1pPA9. Realising acoustic projectors using systems of metamaterial lenses.** Chinmay Rajguru (Eng. and Informatics, Univ. of Sussex, Falmer, United Kingdom) and Gianluca Memoli (Univ. of Sussex, Informatics, Brighton, United Kingdom, G.Memoli@sussex.ac.uk)

In this work, we demonstrate how the use of acoustic metamaterials allows using concepts from optics even when ray optics is not applicable. In particular, we use two compact metasurfaces, designed to be acoustic lenses, to realise a Keplerian acoustic telescope. We use the telescope in front of a loudspeaker to realise a sound projector, capable of delivering sound in specific positions within a diameter of 4m from the source. We first evaluate the accuracy and the limitations of our device by running classical microphone measurements, highlighting the shape of the delivered field and the contrast of the "image" with the surroundings. We then run two indoor localization experiments involving human participants, using either tonal signals or classical music. In the first experiment, we set the device to deliver sound in a room and asked participants to find it, while they were conducting a soundwalk across the space. In the second experiment, the participants were sitting in a fixed position and we moved the sound around them, once again asking them to locate where they thought it was coming from. Our results show the similarities of concepts like "aberrations" and "focusing" with optics, when metamaterials are used even when a single loudspeaker, opening possibilities for localised acoustics special effects.

**4:35**

**1pPA10. The notion of berry phase in classical quantum analogous system.** Kazi Tahsin Mahmood (Mech. Eng., Wayne State Univ., 5050 Anthony Wayne Dr., Ste. #2100, Detroit, MI 48201, hk2799@wayne.edu) and M. Arif Hasan (Mech. Eng., Wayne State Univ., Detroit, MI)

The geometric or Berry phase has become essential in bridging the classical and quantum fields. These phases arise from the cyclic change of the system in a specific path. A quantum analogous classical system is considered to analyze the Berry phase through the evolution of the superposition of states. A classical granular system driven harmonically can create an elastic bit equivalent to a quantum bit, which is manipulated through the driver's frequency, amplitude, and static precompression. The granular system is adaptable to harmonic excitation due to the Hertz-type contact, resulting in different responses, from nonlinear to linearized. We analyze the Berry phase in both cases, where the elastic bit can be manipulated by varying the external excitation, and in a nonlinear system, time changes the superposition of states in the Hilbert space. This results in the accumulation of the Berry phase in time. We show the accumulation of the Berry phase both theoretically and experimentally. This study is important, as the Berry phase accumulation in a classical system shows its uses in topological computing. The Berry phase proves that the linearized and nonlinear systems are decoherence-free, resulting in a powerful tool for information processing.

# Session 1pPP

## Psychological and Physiological Acoustics: Psychological and Physiological Acoustics I

Chad Bullard, Chair
*Indiana Univ., 2631 East Discovery Parkway, Bloomington, IN 47408*

**Chair's Introduction—3:00**

### *Contributed Papers*

**3:05**

**1pPP1. Sound travels: Exploring the effects and meanings of sound across variable free-choice learning environments.** Nickolay Hristov (TERC, 2067 Massachusetts Ave., Cambridge, MA 02140, nickolay_hristov@terc.edu), Donnelley Hayde, Justin Reeves Meyer (Ctr. for Research and Evaluation, COSI, Columbus, OH), Kim Kawczinski (Music Theory and Cognition, Northwestern Univ., Evanston, IL), Laura Weiss (Ctr. for Res. and Eval., COSI, Columbus, OH), Elise Levin-Güracar (TERC, Cambridge, MA), Joe Heimlich (Ctr. for Res. and Eval., COSI, Columbus, OH), Daniel Shanahan (Music Theory and Cognition, Northwestern Univ., Evanston, IL), and Martha Merson (TERC, Cambridge, MA)

Sound Travels is a US-based, federally-funded collaboration between sound researchers, learning researchers, and educational practitioners working to understand the role of soundscapes on free-choice, out-of-school learning experiences. In this paper, members of our research team describe how we have combined approaches from acoustic ecology and visitor studies to navigate the affordances and challenges of studying sound across several complex leisure settings (a science museum, a botanic garden, a park, and a zoo). As an exploratory and transdisciplinary project, our initial work has involved significant deliberation about how to meaningfully and effectively gather data in highly variable acoustic environments, as well as what types and characteristics of sound data are most salient to understanding visitors' experiences of sound. In addition to grappling with these technical questions, we have also worked to ensure that our research does not detract from positive visitor experiences in these spaces and that it directly engages perspectives from practitioners and visitors about cognition, affect, and culture. We will describe the logic of the methods we have used to date (stationary ambient recordings, a post-experience visitor questionnaire, and a "sound search" in which visitors record video clips), as well as our plans for further study.

**3:15**

**1pPP2. Influence of recording technique and ensemble size on apparent source width.** Renzhi Guo (School of Architecture, Rensselaer Polytechnic Inst., 31 25th St., Apt. 31, Troy, NY 12180, kerou273.15@gmail.com) and Jonas Braasch (School of Architecture, Rensselaer Polytechnic Inst., Troy, NY)

The impression of listeners to aurally "see" the size of a performing entity is crucial to the success of both a concert hall and a reproduced sound field. Previous studies have looked at how different concert halls with different lateral reflections affect apparent source width. Yet, the perceptual effects of different source distributions with different capturing techniques on apparent source width are not well understood. This study explores how listeners perceive the width of a symphony orchestra by using four stereo and one binaural recording techniques and three wave field synthesis ensemble settings. Beethoven's Symphony No. 8, performed by wave field synthesis in three ensemble settings, was recorded using these five recording techniques and at two distances in EMPAC concert hall. Subjective experiments were conducted using stereo loudspeakers and headphone to play back the recording clips asking the listeners to rate the perceived wideness of the sound source. Results show that recording techniques greatly influence how wide an orchestra is perceived. The primary mechanism used in judging auditory spatial impression differs between stereo loudspeaker and headphone listening. When well-written symphonic music is recorded and reproduced by two-channel stereophony, the changes in instrument positions in terms of increasing or reducing the physical source width do not lead to an obvious increase or reduction on the spatial impression of the performing entity.

**3:25**

**1pPP3. The effect of reverberation time on the spatial relationships between interaural differences in a room.** William Hartmann (Michigan State Univ., 749 Beech St., East Lansing, MI 48823, wmh@msu.edu)

The interaural level difference (ILD) and the interaural phase difference (IPD) are the two important physical features whereby a listener can localize a sound source in space. In an anechoic environment, the ILD and IPD measured at the ears are similar functions of angle as the head rotates. But in a room environment these interaural differences behave very differently as the head moves in the room. Experiments reported in 2022 (J. Acoust. Soc. Am. abst. **152** A91) indicated that the ILD and IPD, as measured along a straight line path, appeared to be Hilbert transforms of one another, at least at a sufficiently low frequency such as 250 Hz. For instance, the Hilbert transform of the ITD agreed with the ILD without any adjustable scaling factor. Recent measurements made in various room environments show that the smaller the direct to reverberant ratio, the better the Hilbert transform relationship describes the interaural differences. Shifts of the Hilbert transforms compared to interaural differences along the straight line path, vanish as the reverberation time approaches four seconds. Because functions and their Hilbert transforms are orthogonal, the interaural differences are maximally confusing as a listener moves in a room.

**3:35**

**1pPP4. Pilot study of annoyance of wind farm noise classified by cluster analysis.** Anne C. Balant (Dept. of Commun. Disorders, SUNY New Paltz, 1 Hawk Dr. HUM13, New Paltz, NY 12561, balanta@newpaltz.edu), Heather L. Lai (Engineering Programs, SUNY New Paltz, New Paltz, NY), and Chih-Yang Tsai (School of Business, SUNY New Paltz, New Paltz, NY)

From a publicly available database of 3000 10-s samples of wind farm noise (WFN) (https://open.flinders.edu.au/articles/dataset/Benchmark_wind_farm_noise_data_set/19618023/1), we extracted 840 samples that had no appreciable environmental sounds using YAMNet analysis, spectrogram inspection, and listening. The 31 parameters that have been used to identify amplitude modulation (AM) via machine learning (https://github.com/ducphucnguyen/WFN_AM_Detection) were calculated for the 840 identified samples. The resulting values were used to group the samples via cluster analysis. This separated the files into clusters with qualitative differences, e.g., broad spectrum with little or no apparent amplitude modulation (AM), broad spectrum with apparent AM, predominantly low-frequency spectrum with pulsating low-frequency tones, etc. We present the results of listening tests on noise samples that are typical of the different clusters. Samples were rated by

young adults with no prior listening training who passed a hearing screening. Testing was conducted in a single-walled audiometric test suite. Participants responded to questions regarding their perception of the noise characteristics, rated the annoyance of the samples using a graphical user interface, and ranked their annoyance using a paired comparison method. The results obtained via the two methods will be described and compared.

**3:45**

**1pPP5. Fiftieth anniversary of the Environmental Protection Agency's "noise levels" monograph.** Daniel Fink (The Quiet Coalition, 60 Thoreau St., Ste. 261, Concord, MA 01742, DJFink@thequietcoalition.org)

In 1974, as mandated by Congress in the Noise Control Act of 1972, the Environmental Protection Agency (EPA) published *Levels of Environmental Noise Requisite to Protect Public Health and Welfare with an Adequate Margin of Safety*. The Foreword stated, "not all of the scientific work that is required for basing such levels of environmental noise on precise objective factors has been completed…Nonetheless, there is information available from which extrapolations are possible and about which reasoned judgments can be made." Based on then-available science, the EPA calculated that the noise level to prevent hearing loss was $L_{eq(24)} =< 70$ dB; to prevent outdoor activity interference and annoyance $L_{dn} =< 55$ dB for areas in which quiet is a basis for use and $L_{eq(24)}=< 55$ dB for areas in which people spend limited amounts of time; for indoor residential areas $L_{dn}=<45$ dB and $L_{eq(24)}=<45$ dB for other indoor areas. The monograph specified that these numbers were not federal standards. Remarkably, subsequent research published in thousands of peer-reviewed articles, and analyses leading to noise exposure recommendations from the World Health Organization, have largely confirmed the accuracy of the EPA's fifty-year old calculations. They got it right! This milestone anniversary should be celebrated.

**3:55**

**1pPP6. Relating self-reported auditory difficulties, suprathreshold auditory processing, and cognition after brain injury.** Tess K. Koerner (VA RR&D NCRAR, 3710 SW US Veterans Hospital Rd, Portland, OR 97239, koern030@gmail.com) and Frederick J. Gallun (Oregon Hearing Research Ctr., Oregon Health and Science Univ., Portland, OR)

The Portable Automated Rapid Testing (PART) app increases access to psychophysical tasks that are not typically available outside of specialized laboratory settings. PART contains a wide array of suprathreshold auditory and cognitive processing measures that can be completed in a short amount of time, making it attractive to audiologists who are interested in clinically assessing different aspects of auditory function. This may be particularly useful for testing normal-hearing patients with auditory complaints that are not well predicted by standard audiometric tests, such as those with a history of mild traumatic brain injury (mTBI). To begin exploring this possibility, this study aimed to identify whether performance on a battery of PART tests is predictive of auditory complaints in participants with $(n = 27)$ and without $(n = 28)$ a history of mTBI. Regression modeling was used to determine relationships between self-reported auditory difficulties and performance on measures of spectrotemporal modulation detection, frequency modulation detection, spatial release from masking, as well as memory and attention. Findings contribute to our understanding of the effects of mTBI on auditory processing. The importance of integrating remote testing platforms into research on understudied patient populations and of interdisciplinary collaboration with clinicians will also be discussed.

**4:05**

**1pPP7. Indicators of noise damage: Association between lifetime noise-exposure questionnaire, high-frequency audiogram, and categorical loudness scaling.** Leny Vinceslas (Computer Sci., UCL, 169 Euston Rd., London NW1 2AE, United Kingdom, l.vinceslas@ucl.ac.uk), Vit Drga (Computer Sci., UCL, London, United Kingdom), Jesko Verhey (Dept. of Exp. Audiol., Otto von Guericke Univ. Magdeburg, Magdeburg, Sachsen-Anhalt, Germany), and Ifat Yasin (Computer Sci., UCL, London, United Kingdom)

Some effects of noise-induced hearing loss may be evident at the neural level, such as cochlear synaptopathy but not necessarily manifest at the audiogram level (0.25–8 kHz); although indicators of early noise-induced hearing loss may be observable at higher audiometric frequencies. The current study compares two populations; both populations have normal hearing up to 8 kHz; however, one population has minimal lifetime exposure history for high-level noise, and the second population has previous and ongoing significant noise-exposure history. The individual's experience of noise exposure is assessed with a structured interview approach eliciting the level and duration across a range of noise exposure activities. Categorical loudness scaling (CLS) is used to quantify individual loudness perception. The slope of the low-level portion of the CLS function may be associated with increasing audiometric threshold. Different fitting procedures for the CLS were evaluated. The CLS slopes were compared with audiograms (0.25-16 kHz) as well as two evaluations of uncomfortable loudness levels (ULL); one using the traditional audiometric tonal stimuli and the second using the CLS broadband noise stimuli. This study investigates the extent to which noise damage can be deduced from a lifetime noise-exposure questionnaire, high-frequency audiogram, and categorical loudness scaling data.

**4:15**

**1pPP8. Towards the detection of Alzheimer's disease through eye movement changes using a hearable.** Miriam Boutros (Elec. Eng., École de technologie supérieure, 1100 Notre-Dame St. W, Montreal, Quebec H3C 1K3, Canada, miriam.boutros.1@ens.etsmtl.ca), Arian Shamei (Elec. Eng., École de technologie supérieure, Montreal, Quebec, Canada), Christopher Niemczak (Geisel School of Med., Dartmouth College, Hanover, NH), and Rachel Bouserhal (Elec. Eng., École de technologie supérieure, Montréal, Quebec, Canada)

Hearables are wearable devices with in-ear microphones that utilize the occlusion effect to detect amplified low-frequency signals propagated by tissue and bone conduction, such as eardrum oscillations caused by eye movements [Greuters *et al.*, 2018, PNAS, 115(6)]. Saccades, which are rapid and simultaneous movements of both eyes, provide valuable insight into a person's motor abilities and can be used to assess cognitive dysfunction. Research indicates that individuals with Alzheimer's disease exhibit delayed, slow, and hypometric (i.e., undershooting) saccades, along with less fixation stability [Fletcher & Sharpe, 1986. An. Neuro. 20(4)]. In this project, 35 patients with Alzheimer's disease or mild cognitive impairment, along with 35 matched control participants, will undergo various experiments, such as a picture description task, while wearing an eye tracking device and a hearable. The objective is to correlate recorded eardrum oscillations from the hearable and the amplitude and trajectory of horizontal and vertical saccades from the glasses. In addition, we aim to predict AD group inclusion based on data collected from the hearable. The long-term goal of this project is to develop a wearable, non-intrusive, and easy-to-use device capable of identifying Alzheimer's disease and potentially predicting it earlier than the standard diagnosis timeframe.

**4:25**

**1pPP9. Can resting electroencephalography index the development of listening-related fatigue?** Cynthia Hunter (Speech-Language-Hearing, Univ. of Kansas, 1000 Sunnyside Ave, DHDC 3000, Lawrence, KS 66045, c.hunter@ku.edu), Lydia McMullen, and Kiana Magee (Speech-Language-Hearing, Univ. of Kansas, Lawrence, KS)

Sustained exertion of listening effort by people with hearing loss in order to recognize speech is thought to lead to mental fatigue. Listening-related fatigue has been measured primarily with self-reports. The current study tests the hypothesis that an objective neural measure of increased post-task power of alpha (8-12 Hz) and theta (4-8 Hz) neural rhythms from the resting electroencephalogram (EEG) can index the development of listening-related fatigue. Before and after a challenging listening task, 2 min of eyes-closed and eyes-open resting EEG and self-reports of fatigue on a visual-analogue scale were collected from older adults $(n = 28)$ with age-appropriate hearing (pure-tone average hearing loss from 1, 2, and 4 kHz $M = 31.94$, $SD = 16.74$). A numerical trend for the expected post-task increase in power was observed for all EEG measures. However, the difference was significant only for the alpha-band measures. Self-reported fatigue also increased significantly post-task, confirming that the listening task induced the subjective experience of fatigue. These findings indicate that oscillatory power collected from resting EEG before and after a listening task has potential to serve as an objective, brain-based indicator of listening-related fatigue in older adults, with alpha as the most promising frequency band.

## Session 1pSA

## Structural Acoustics and Vibration: Lab Tours in Structural Acoustics and Vibrations

Michael L. Dickerson, Cochair
*None, 170 S William Dillard Dr., Ste 103, Gilbert, AZ 85233*

Chengzhi Shi, Cochair
*Univ. of Michigan, 2350 Hayward St., Ann Arbor, MI 48109*

**Chair's Introduction—3:00**

### *Invited Papers*

**3:05**

**1pSA1. An overview of two vibration laboratories at Brigham Young University.** Micah Shepherd (Dept. of Phys. and Astronomy, Brigham Young Univ., N249 ESC, Provo, UT 84602, mrs74@byu.edu), Peter K. Jensen, Joshua T. Mills (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), Matthew S. Allen, Marcus Behling, Suzanna Gilbert, Jonathan Black, and Karl Wolfley (Mech. Eng., Brigham Young Univ., Provo, UT)

A virtual tour will be given of two research laboratories at Brigham Young University. In the first laboratory, an experimental apparatus for measuring the internal damping of thin beams will be shown. Here, a beam specimen is placed in a vacuum chamber. After pulling a vacuum, the beam is excited with an automatic force hammer and the response is measured with a single-point laser vibrometer. The velocity data is recorded using a data acquisition system controlled with a custom python script. In the second laboratory, three types of tests will be highlighted. In the first, several small shakers are used to study methods for reproducing the vibration environments that components experience in launch vehicles, and this is compared with traditional, single-axis testing. The second type of experiment is used to characterize structures with weak nonlinearities. The structures are excited impulsively and accelerometers are used to measure the effective natural frequency and damping. This is used to characterize structures with bolted or riveted joints as well as additively manufactured beams with pockets of unfused powder. Finally, an apparatus for inferring the stress in the Achilles tendon is demonstrated. Waves are excited in the tendon and the wave speed is measured by calculating the time delay between two accelerometers that are pressed on the surface of the skin.

**3:25**

**1pSA2. Virtual tour of MDAcostics LLC Lab.** Michael L. Dickerson (None, 170 S William Dillard Dr., Ste 103, Gilbert, AZ 85233, mld@mdacoustics.com)

MD Acoustics will provide a virtual tour of our commercial anechoic and reverberate chambers where we design, test, and experiment on products for industry and the government. We may show things like Pickleball paddle and ball development, car wash blowers development, IC vibration isolation development, and how to quiet buildings that have perforated aluminum panels. This is different than many labs taht just measure the acoustic performance of a product.

**3:45–5:00 Discussion**

## Session 1pSCa

# Speech Communication: Student Choose-Your-Own-Adventure—Acquisition and Bilingualism

Natasha Warner, Chair
*University of Arizona, Tucson, AZ*

### *Contributed Papers*

**3:00**

**1pSCa1. Cross-linguistic voice variations in Korean-English bilinguals.** Haneul Lee (English Lang. and Literature, Seoul National Univ., 1 Gwanak-ro, Gwanak-gu, Seoul 08826, Korea (the Republic of), hanale@snu.ac.kr) and Harim Kwon (English Lang. and Literature, Seoul National Univ., Seoul, Korea (the Republic of))

Languages often exhibit distinct pitch ranges and voice qualities. To investigate how bilingual speakers modulate their voice quality and pitch in their two languages, this study examines voice variation in Korean-English bilingual speakers. We recorded 30 Korean-English bilinguals (F = 15, all more dominant in Korean) reading "The North Wind and the Sun" in their two languages, and measured F0, SHR, and two spectral tilt measures (H1*-H2*, H1*-A1*) at 5 ms intervals in all sonorant sounds. The results showed that the bilingual speakers' two languages were indeed different as reflected in the four measures, with the differences mediated by their English proficiency and gender. Overall, mean F0 and F0 variation were greater in Korean than in English. Interestingly, higher English proficiency correlated with greater F0 variation in English, making it more similar to Korean. Bilingual speakers produced English with stronger subharmonics, suggesting a creakier phonation, than Korean, indicated by higher SHR values. Finally, spectral slope measures showed gender-specific patterns: males exhibited greater values (suggesting breathiness) in Korean while females showed the opposite. These findings suggest that bilingual speakers have distinct voice in their two languages, but their vocal adjustments are influenced by language proficiency and gender of the speakers.

**3:05**

**1pSCa2. Phonetic L2 development and phonetic attrition in L1 Arabic L2 English immigrants' voiced stop production.** Dema Smadi (The Dept. of Linguistics & TESOL, The Univ. of Texas at Arlington, 4201 City Pt W Apt. 4306, North Richland Hills, TX 76180, dxs9705@mavs.uta.edu)

This study investigates the impact of L2 development in L1 Arabic L2 English speakers on their L1 voiced stop production during a word reading task potentially indicating a use-driven aspect to phonetic attrition in immigrants' speech. The study involved 10 participants who have been residing in the US for durations ranging from 1 to 24 years. Previous research on the influence of L2 exposure on L1 speech found conflicting results: some studies suggest stronger phonetic attrition following prolonged immersion, while others emphasize changes in the early stages of exposure (Flege, 1987; Chang, 2012; 2013). They used The Speech Learning Model (SLM) to explain their findings by focusing on SLM predictions on L1 production. However, addressing changes in L1 speech without accounting for L2 speech and its developmental effects, a gap noted in previous studies, seems inaccurate. The L1-L2 link is essential to SLM. Therefore, this study aims to address this gap. The results indicate a correlation between L2 development and L1 production of voiced stops, with voice onset times being the measured phonetic dimension. Participants demonstrating higher proficiency in L2 tend to exhibit L1 speech that more closely mirrors L2 characteristics, suggesting a link between L1 production and L2 development.

**3:10**

**1pSCa3. Stroop task response time: A comparison between sequential bilinguals.** Sophia Launay-Fallasse (Speech-Language-Hearing Sciences, Loyola Univ. Maryland, 4501 N Charles St., Baltimore, MD 21210, smlaunay-fallasse@loyola.edu), Kathleen Siren, and Tepanta Fossett (Speech-Language-Hearing Sciences, Loyola Univ. Maryland, Baltimore, MD)

Research on bilingual individuals provides evidence of a cognitive advantage compared with monolingual speakers. The dominant hypothesis is that language switching for bilingual individuals stimulates the executive function system helping to minimize cognitive interference and allow for multitasking situations. That is, bilingual individuals likely need to actively suppress interference from one language when they are in another language environment. Utilizing the Stroop task, a neuropsychological experiment measuring the brain's speed and capacity to process competing information, researchers have demonstrated that bilingual speakers perform better than monolingual speakers. However, there is a paucity of research investigating the effect of order of language acquisition. This investigation examines accuracy and speed of verbal responses during a written word Stroop task for bilingual individuals who learned English first compared to bilingual individuals who learned English second. Twenty participants, ten whose language acquisition order was English then Spanish and ten whose language acquisition order was Spanish then English, completed the Stroop task in English. Preliminary results demonstrate no differences between the groups, suggesting language acquisition order does not impact the bilingual advantage in executive functioning.

**3:15**

**1pSCa4. The connection between articulatory skill and higher-level executive function in monolinguals and bilinguals.** Imani Lyte (Communications & Performing Arts, City Univ. of New York - Kingsborough Community College, 20001 Oriental Blvd., Brooklyn, NY 11235, imani.lyte47@students.kbcc.cuny.edu), Tanzeela Jahangir, and Laura Spinu (Commun. & Performing Arts, City Univ. of New York - Kingsborough Community College, Brooklyn, NY)

While the existence of a bilingual advantage in higher-level executive function remains controversial, recent studies showed bilinguals consistently outperform monolinguals in learning new patterns of pronunciation (in both natural or artificially constructed novel dialects). Since audition and articulatory control are involved in this type of learning, the question arises to what extent these lower-level functions are enhanced in people with superior performance on phonetic learning tasks. Dugaillard & Spinu (2019) found that sequential bilinguals outperformed monolinguals and simultaneous bilinguals on a tongue-twister task (Goldrick & Blumstein 2006, McMillan&Corley 2010), suggesting an advantage in articulatory skill. In the current study, we aim to replicate the original findings with a new group of bilinguals and further explore the connection of articulatory skill with one aspect of executive function (i.e. interference/conflict resolution). The experimental tasks include (1) reading artificially constructed tongue-twisters such as "kifkivkivkif" three times in succession, matching a 150 bps rhythm on a metronome, and (2) a classic Simon task asking participants to react or inhibit their response to congruent and incongruent stimuli. We expect the findings to shed more light on

the bilingual advantage phenomenon and the connection between lower- and higher-level cognitive functions in both populations.

**3:20**

**1pSCa5. Is there an additive effect of speaking more than two languages on the performance of mental arithmetic tasks?** Heather Ye (KBCC, Brooklyn, NY), Samrat Chowdhury (KBCC, 2001 Oriental Blvd., Brooklyn, NY 11235, Samrat.chowdhury60@students.kbcc.cuny.edu), and Anastasiia Myslyk (KBCC, Brooklyn, NY)

Numerous studies suggest the bilingual brain exhibits cognitive differences compared to the monolingual one (Bialystok 2018, Spinu *et al.* 2018, Macnamara & Conway 2014, and Krizman *et al*. 2012), though these findings are still controversial (Dick *et al*. 2019, Marzecova 2015, Paap & Greenberg 2013). Among others, bilinguals have exhibited a greater digit span recall (Spinu 2023), and a greater ability to multitask (Bialystok *et al*. 2012). McKenzie & Spinu (2022) uncovered differences in performance between monolinguals and bilinguals when completing arithmetic tasks. The current study addresses the question whether speaking more than two languages can further improve mental arithmetic skills. We asked participants speaking one, two, or more languages to complete an arithmetic task in which they either memorized 4-digit sequences or performed additions on the sequences presented. The task was administered online using Pavlovia software, with each participant being supervised remotely via Zoom teleconferencing. While the experiment is still underway, we predict the existence of an additive effect of speaking multiple languages, with individuals speaking more languages outperforming those speaking less languages on mental arithmetic tasks. Our findings will add to the existing body of work concerning the cognitive differences of tri- and multilingual populations, which to date remain understudied.

**3:25**

**1pSCa6. A new perspective on the bilingual advantage debate: do monolinguals and bilinguals perform differently on arithmetic tasks?** Laura Spinu (Communications & Performing Arts, City Univ. of New York - Kingsborough Community College, Brooklyn, NY), Celeste C. Mckenzie (Linguistics, Kingsborough Community College, 2001 Oriental Blvd., Brooklyn, NY 11210, celestemckenzie13@gmail.com), and Tung-Hsin Ye (Kingsborough, Brooklyn, NY)

Research suggests bilinguals exhibit cognitive differences, sometimes in the form of advantages, compared to monolinguals (Spinu *et al*. 2023, Macnamara & Conway 2014, and Krizman *et al*. 2012), though these findings are still controversial (Dick *et al*. 2019, Marzecova 2015, and Paap & Greenberg 2013). Among others, bilinguals show greater serial memory span (Spinu 2022), inhibitory mechanisms, and multitasking ability (Bialystok *et al*. 2012). The goal of this experiment is to expand on previous findings by determining if monolinguals and bilinguals also perform differently on arithmetic tasks. Participants were asked to either repeat a 4-digit sequence (easy task) or produce the numbers resulting from adding 3 to each digit presented e.g. 5894 for 2561 (difficult task). The experiment was administered online using Pavlovia software with remote supervision via Zoom. For each speaker, we obtained accuracy scores. The two groups did not differ on their performance on the easy task, on which they scored highly in general, but bilinguals outperformed monolinguals on the difficult task. By taking the first step towards exploring an understudied direction in the bilingual cognition literature, we have added new data to the body of work supporting a cognitive advantage associated with bilingual experience.

**3:30–4:00 Discussion**

TUESDAY AFTERNOON, 19 NOVEMBER 2024 — 3:00 P.M. TO 5:00 P.M.

## Session 1pSCb

## Speech Communication: Student Choose-Your-Own-Adventure—Speech Articulation

Steven Lulich, Chair
*Indiana University, Bloomington, IN 47408*

### *Contributed Papers*

**3:00**

**1pSCb1. Implementing artificial intelligence models for automatic vocal tract image segmentation.** Kyle Ng (Linguistics, Univ. of Southern California, 3601 Watt Way, Grace Ford Salvatori Hall 301, Los Angeles, CA 90089-1693, kgng@usc.edu), Ameen Qureshi, Haley Hsu, Dani Byrd, and Khalil Iskarous (Linguistics, Univ. of Southern California, Los Angeles, CA)

Modeling articulatory representations is critical to the scientific study of speech production and has been addressed by quantifying dynamical movements of the vocal tract. However, modeling dynamic articulatory data has proven time-consuming and computationally taxing. For example, real-time imaging data of the vocal tract has been assessed using contour-tracking methods for image segmentation that require manually creating vocal tract templates and human supervision to correct image segmentation (e.g., Bresch & Narayanan 2009). Recently, new work has introduced multimodal approaches to model articulatory data (Jain *et al*., 2024), but these approaches still require manual initiation/correction. We combine previous segmentation methods such as the Chan-Vese Segmentation algorithm (Getreuer 2012) with AI models like the Segment Anything Model (SAM) (Kirillov *et al*. 2023) to test means for optimizing image segmentation of vocal tract articulators in real-time magnetic resonance imaging (rtMRI) data. We deploy non-linear filtering and parameter optimization so as to prioritize desirable image processing features, such as smoothing the image while sharpening the edges. Processing these optimized images into SAM allows for segmentation of critical vocal tract articulators and the airway. In sum, the goal of this work is to automatically segment without manual intervention moving articulators in rtMRI speech production data. [Work supported by NSF.]

3:05

**1pSCb2. Location and size of constriction in labiovelar and velar sounds in English.** Victor Wong (Linguistics, Univ. of British Columbia, 2613 West Mall, Vancouver, British Columbia V6T 1Z2, Canada, victorw1@student.ubc.ca), Dayeon Choi, Ragul Loganathan, Jahurul Islam, and Bryan Gick (Linguistics, Univ. of British Columbia, Vancouver, British Columbia, Canada)

Not all velar sounds are produced using a velar constriction. Previous research [Islam *et al.*, 2024, *JASA* 155] reveals both velar and palato-velar constriction locations in Parisian French velars. The current study investigates whether this variation is language-specific by comparing the size and location of constriction of velar sounds in English using an rtMRI speech corpus [Lim *et al.*, 2021, Scientific Data 8]. Using Whispgrid [Martin, 2024, *Github*] and Montreal Forced Aligner [McAuliffe *et al.*, 2017, Interspeech], Praat [Boersma & Weenink, 2024] textgrids were auto-generated and manually corrected in sentence and phone tiers to extract MRi freeze-frames from 10 participants split evenly between (5) L1 and (5) L2 English speakers. A Python script measuring Euclidean Distance located the size and constriction of English velar and labiovelar sounds between the palate and tongue. Results from MRI freeze-frames suggest that in English L1 and L2 speech, velar stops maintain a velar constriction and are not fronted across vocalic contexts found in French velar stops. Variation for velars indicates language-specific trends in constriction location.

3:10

**1pSCb3. Investigating speech articulatory control through dynamical model fitting.** Michael C. Stern (Linguistics, Yale Univ., 135 Howe St, Unit B, New Haven, CT 06511, michael.stern@yale.edu) and Jason A. Shaw (Linguistics, Yale Univ., New Haven, CT)

Dynamical models of speech articulation link the continuous trajectories of articulators with stable parameters of phonological knowledge (e.g., Browman & Goldstein, 1989). We examined the fit of a recently proposed dynamical model (Stern & Shaw, 2024) to electromagnetic articulography (EMA) recordings of twelve English speakers and twelve Mandarin speakers. While fits to (labial) consonant constrictions were excellent (mean $R^2 = .97$), fits to consonant release movements and vowel constrictions were slightly worse (mean $R^2 = .91$ and.89, respectively). We explore two implications of this result. (1) Consonant release and vowel constriction target achievement may be coordinated in time, unlike consonant constriction target achievement (Kramer *et al.*, 2023). The addition of a coupling term may improve model fit, constituting evidence for target-based gestural coordination in speech (Turk & Shattuck-Hufnagel, 2020). (2) Vowel constriction movements may be governed by two separate tasks, one controlling constriction location and one controlling constriction degree (Saltzman & Munhall, 1989). Model fit may be improved by separating vowel constriction dynamics into two systems, shedding light on the dimensionality of articulatory control.

3:15

**1pSCb4. Tongue shape variations in laughter and speech: Exploring movement patterns.** Da Yeon Choi (Linguistics, Univ. of British Columbia, 2613 West Mall, Vancouver, British Columbia V6T 1Z2, Canada, dayeoncchoi@gmail.com), Ragul Loganathan, Victor Wong, Jahurul Islam, and Bryan Gick (Linguistics, Univ. of British Columbia, Vancouver, British Columbia, Canada)

Laughter is a common non-verbal communicative function of the vocal tract [Krepsz *et al.*, 2024, CognitiveProcessing, 25(1)]. Previous studies using real-time magnetic resonance imaging (rtMRI) of the vocal tract found spontaneous (natural) laughter to be less speech-like compared to volitional (induced) laughter [Belyk & McGettigan, 2022, *Phil. Trans. Royal Soc. B*, 377(1863)]. Expanding on prior research on laughter production [Belyk & McGettigan, 2022] and ultrasound imaging comparing the production of spoken vowels to trombone notes [Heyne & Derrick, 2019, *Frontiers in Psych.*, 10(2597)], the present study employs ultrasound imaging to analyze tongue shape variations between spontaneous and induced laughter, to ascertain whether laughter draws on the speech movement inventory. Video stimuli were presented to elicit spontaneous laughter, and vowels [i, ɪ, e, ɛ, æ, ʌ, ə, ʊ, u, o, ɔ, ɑ] were produced to compare articulatory postures. Acoustic analyses using Praat [Boersma & Weenink, 2024] and ultrasound data will be presented comparing spontaneous vs. natural laughter and their similarity to vowels in the speech inventory. Induced laughter is predicted to draw on speech-like behaviour, exhibiting similar tongue postures to speech compared to natural laughter. Implications will be discussed regarding similarities and differences in tongue positioning between spontaneous, volitional laughter, and speech.

3:20

**1pSCb5. Multi-atlas-based segmentation of vocal tract anatomy for children using label fusion techniques from dynamic MRI.** Hahn Kang (Boston College, 125 Nashua St., Ste. 660, Boston, MA 02114, hkang14@mgb.org), Fangxu Xing (Radiology, Harvard Medical School, Boston, MA), Imani Gilbert (ECU, Greenville, NC), Jiyoon Kim (UIUC, Champaign, IL), Abigail R. Kostolansky (UIUC, Urbana, IL), Jamie Perry (ECU, Greenville, NC), Bradley P. Sutton (UIUC, Urbana, IL), and Jonghye Woo (Radiology, Harvard Medical School, Boston, MA)

Segmentation of oral and velopharyngeal anatomy is essential for quantifying anatomical structures and analyzing the dynamic and discrete movements of these structures during speech production. This aids in the diagnosis and treatment of speech-related disorders as the different speech articulators are able to be analyzed independently and in relation to their coarticulatory movements. Direct visualization of structures and musculature important for speech production is best attained using MRI. Manual segmentation, however, is labor-intensive and often suffers from poor reproducibility, requiring automatic segmentation techniques to enhance efficiency. To address this, we aim to compare multi-atlas-based segmentation techniques to delineate the tongue, velum, and adenoid. We first constructed five spatiotemporal atlases of speech tasks, each with over 80 time frames. We treated each time frame as a single atlas, with diffeomorphic registration of 10 to 40 time frames in steps of 10 from the anchor spatiotemporal atlas with the other four atlases. Manual labels were propagated across the four atlases. Two label fusion techniques were compared: majority voting and joint label fusion, evaluated using the Dice Similarity Coefficient (DSC). Our results demonstrate that joint label fusion outperformed majority voting, achieving a DSC of over 0.84 on all four atlases, with the highest reaching a DSC of 0.87. These findings indicate that joint label fusion offers a promising tool for velopharyngeal and vocal tract motion analysis.

3:25

**1pSCb6. Towards large-scale cross-speaker articulatory modeling of vowels.** Sean Foley (Univ. of Southern California, 3601 Watt Way, Grace Ford Salvatori Hall 301, Los Angeles, CA 90089-1693, seanfole@usc.edu) and Shrikanth Narayanan (Univ. of Southern California, Los Angeles, CA)

While previous studies have attempted to decompose vowel articulatory data into a set of basis factors, these studies have often been limited in both scale and the data being sparsely sampled, limiting interpretability and generalizability of the results (Nix *et al*. 1996 and Serrurier *et al*. 2019). In this study, the data were analyzed from 36 (23F, 13M) American English speakers producing 13 vowels in bVt sequences obtained using real-time MRI. Midsagittal tongue contours were obtained during vowel productions for all speakers using a semi-automated segmentation algorithm (Jain *et al*. 2024). Frames corresponding to the vowel articulation were segmented using MFA and simultaneously recorded audio. A combination of Procrustes analysis for cross-speaker normalization and guided PCA were employed to decompose the pooled articulatory space into a set of vowel "primitives." 71% of the variation within the articulatory space is captured in the first three principle components, with the first component representing variation in tongue body advancement/retraction, the second component fronting/backing, and the third lesser degrees of fronting/backing. This pattern is consistent when the data is split into male/female groups. Vowel-wise PCA loadings for the first two components mostly show expected degrees of separation between the dimensions of advancement/retraction and backness, though some distinctions appear to be diluted, likely attributable to the normalization procedure. [Work supported by the NIH.]

### 3:30

**1pSCb7. Effects of visual perception of gravity on tongue posture.** Masha Chernets (Linguistics, Univ. of British Columbia, 2613 West Mall, Vancouver, BC V6T 1Z2, Canada, achernet@student.ubc.ca), Amy Hong, Victor Wong, Anna S. Ferguson, Jahurul Islam, Yadong Liu, and Bryan Gick (Linguistics, Univ. of British Columbia, Vancouver, BC, Canada)

Visual perception of gravity change may impact tongue posture during speech production. Previous studies [Chander *et al.*, 2019, Behav. Sci., 9(11)] report anticipatory visual disturbances induce compensatory postural behavior in the human body and [Philips *et al.*, 2022, Theor. Iss. Ergon. Sci., 23(1)] showed the magnitude of postural response is dependent on velocity and direction of movement. Prior research [Assländer *et al.*, 2023, Sci. Rep., 13(1), 2594] investigated the visual component impact on body posture balance in virtual-reality. Building on [Shamei *et al.*, 2023, Sci. Rep. 13(1), 8231] investigation of postural adaptation in microgravity, the current study investigates postural behavior during spoken production of elongated vowels through a gravitational virtual-reality environment. Ultrasound imaging measured the impact of tongue posture for vowel production in an immersive virtual-reality rollercoaster through gravitationally stable, rising, and falling conditions. Acoustic analyses compared the production of elongated vowels [i, e, o, u, ɑ] across conditions. The falling condition is predicted to have the highest F1 followed by the leveled, and raising conditions. F1 will be reported by comparing the tongue height across conditions. Implications for postural adaptation in virtual-reality conditions will be discussed.

**3:35–4:35 Dsicussion**

TUESDAY AFTERNOON, 19 NOVEMBER 2024 — 3:00 P.M. TO 5:00 P.M.

## Session 1pSCc

## Speech Communication: Student Choose-Your-Own-Adventure - Speech Perception

Benjamin V. Tucker, Chair

*Commun. Sci. and Disorders, Northern Arizona Univ., 208 E. Pine Knoll Dr., P.O. Box: 15045, Flagstaff, AZ 86011*

### *Contributed Papers*

### 3:00

**1pSCc1. How well do acoustic parameters correlate our perception of voice quality?** Yat Chun Au (Academic Unit of Human Communication, Learning, and Development, Faculty of Education, The Univ. of Hong Kong, Room 759, Meng Wah Complex, Pokfulam, Hong Kong, auandrew@connect.hku.hk) and Manwa L. Ng (Academic Unit of Human Communication, Learning, and Development, Faculty of Education, The Univ. of Hong Kong, Pokfulam, Hong Kong)

Auditory perceptual judgments are essential for assessing voice disorders and monitoring treatment progress. This study aimed to examine the acoustic foundations of perceived abnormalities in individuals with and without voice disorders, focusing on the correlation between the Grade, Roughness, Breathiness, Asthenia, Strain (GRBAS) scale and various acoustic parameters. A total of 1217 voice recordings were obtained from the Perceptual Voice Qualities Database (PVQD) and the Saarbruecken Voice Database (SVD). Each recording was perceptually evaluated by five experienced speech therapists based on the GRBAS scale. Only ratings of more than 80% inter-rater agreement was used for further analysis. Acoustic analyses were obtained using Parselmouth, a Python library for the Praat software. Statistical analysis using Spearman's rank correlation revealed strong correlations between spectral and cepstral results and GRBAS scale. Specifically, the Grade and Breathiness components showed significant negative correlations with the CPPS parameters, with and without voice detection. There was a strong inverse relationship between Harmonics-to-Noise Ratio (HNR) and the Grade and Roughness. Jitter and shimmer demonstrated robust positive correlations with Grade, Breathiness, and Roughness. These findings provide valuable insights into the relationships between acoustic parameters and perceptual GRBAS scale, informing future research and clinical practice in voice and speech pathology.

### 3:05

**1pSCc2. An Ideal observer's classification accuracy of speakers' diverse gender identities improves with prior experience.** June M. Contreras (Speech Language and Hearing Sciences, Univ. of Texas at El Paso, 3532 Keltner, El Paso, TX 79904, jmcontreras2@miners.utep.edu) and Brandon "Brooke" Merritt (Rehabilitation Sciences, The Univ. of Texas at El Paso, El Paso, TX)

Kleinschmidt *et al.* (2018) show it is possible to probabilistically infer sex from acoustic cues using Kleinschmidt and Jaeger's (2015) ideal adapter framework. Kleinschmidt *et al.*'s implementation of this framework predicts that listeners condition acoustic cue distributions on socio-indexical variables they consider meaningful based on experience. Because Kleinschmidt *et al.* only included (presumably) cisgender speakers in their study, it is unclear how their implementation performs when inferring gender from speakers of diverse gender identities. Here, we explore the model's accuracy when inferring diverse gender identities using Kleinschmidt *et al.*'s methods. Two experiments are conducted where models infer transgender and cisgender speakers' gender identities based on different acoustic cues [fundamental frequency ($f_o$), the first two formants, and $f_o$ + formants]. In experiment one, the models are exposed to all cisgender speakers and a percentage of the transgender speakers that increases over training phases. Models then infer the gender identity of speakers to whom they are not exposed. Experiment 2 expands on Experiment 1 by first exposing the models to all transgender speakers and then examining their predictive accuracy. We expect that the models' classifications will be most accurate when provided with salient, gender-coded acoustic features of $f_o$ in conjunction with formants.

**3:10**

**1pSCc3. Accented sentence and word recognition: Humans versus whisper automatic speech recognition.** Junrong Chen (Cognitive Science, Univ. of California, San Diego, 9450 Gilman Dr., La Jolla, CA 92092, juc036@ucsd.edu), Jan Kwong, and Sarah C. Creel (Cognitive Science, Univ. of California, San Diego, La Jolla, CA)

Despite advancements in speech recognition technology, questions remain about model generalizability and how much models mirror human perception. These questions are addressed by comparing OpenAI's Whisper model and 75 human transcribers on 300 English sentences (20 speakers, half F, half M, half US-accented, half [Mexican-]Spanish-accented). Sentences ended in 100 target words, with 1/3 high-predictability sentences *(The farmer milked the cows*) and 2/3 varying degrees of low-predictability *(The farmer/barmer milked the nose)*. Target-word error rate (WER) was examined for final words in sentences and for isolated final words (recordings excised from same sentences). WER decreased with increasing model size, but was higher for Spanish-accented than US-accented speech, suggesting imperfect generalizability. Both models and humans benefited from more-predictable vs. less-predictable sentences. However, using isolated-word WERs as a baseline revealed that sentence context affected models and humans differently: humans benefited only from high-predictability sentences, while models benefited somewhat from *any* sentence context. Humans outperformed models on isolated words, suggesting that Whisper may have a restricted distribution of single-word utterances or may need lengthier acoustic context than humans. Findings suggest that more inclusive, varied training data may yield more generalizable ASR. Potential for using ASR to model human speech adaptability is discussed.

**3:15**

**1pSCc4. On the temporal aspect of post-consonantal fundamental frequency perturbation in English voicing perception.** Chiung-Yu Chang (Psychological and Brain Sciences, Univ. of Massachusetts Amherst, 135 Hicks Way, Amherst, MA 01003, chiungyuchan@umass.edu) and Lisa D. Sanders (Psychological and Brain Sciences, Univ. of Massachusetts Amherst, Amherst, MA)

The fundamental frequency (F0) after English voiceless obstruents is generally higher than their voiced counterparts. This difference is usually largest at the vowel onset and diminishes before the vowel ends. Past research has established that the F0 perturbation influences English listeners' voicing perception. While many studies have highlighted the effect of F0 onset height, the temporal extent of the F0 perturbation is less investigated. In the present study, listeners categorized bilabial stops as voiced or voiceless. We manipulated the voice onset time (VOT), the F0 contour, and the presence of background noise. Except for one level contour at the baseline F0 value, all other F0 contours started at 20 Hz higher than the baseline. Critically, these contours fell to reach baseline at different times, rendering them different slopes. The experimental results revealed an interaction between VOT, F0, and noise. Specifically, when the stimuli were presented in noise and had VOTs near the category boundary, listeners were more likely to categorize F0 falling at slower rates as voiceless than those at faster rates. These results suggested that the duration of F0 perturbation also affects voicing perception in English and might play a role in developing tonal languages.

**3:20**

**1pSCc5. Within speaker rate variability limits habitual speech rate tracking.** Sarah Kapp (Univ. of Freiburg, Sickingenstreet 52, Freiburg, Baden-Württemberg 79117, Germany, sarah.kapp@ru.nl), Giulio Severijnen (Int. School for Adv. Studies, Trieste, Italy), Adriana Hanulikova (Univ. of Freiburg, Freiburg, Germany), and Hans Rutger Bosker (Donders Inst. for Brain, Cognition and Behaviour, Nijmegen, Netherlands)

Listeners track the habitual speech rate of talkers and use this information to resolve temporal ambiguities in vowels by normalizing vowel length for the surrounding speech rate. Previous studies supporting this effect have also shown that the habitual rate of one talker is perceived relative to the habitual rate of another talker. These studies mainly involved speakers with constant habitual rates. However, given the large within-speaker variability in speech rate, it remains unclear whether this mechanism is applicable in natural conversation. This study, therefore, assessed the influence of within-speaker rate variability of one talker on the perception of the Dutch vowel contrast /ɑ/–/a:/ in a second talker. 38 Dutch speakers heard two talkers speaking at different rates. Half of the participants heard Talker A speak at a normal rate and Talker B at a fast rate intermixed with normal rate trials. The other half heard the same Talker A but Talker B at a slow rate and normal rate. Contrary to previous studies, we did not find that Talker B's rate affected perception of Talker A's ambiguous vowels. Therefore, habitual rate tracking seems to be hindered by within-speaker variability in speech rate.

**3:25**

**1pSCc6. Contextual effects on the perception of English interdental fricatives /θ/ by Mandarin Chinese learners.** Yuou Sheng (OISE, Univ. of Toronto, 3589 Kariya Dr., Mississauga, Ontario L5B 3J2, Canada, yuou.sheng@mail.utoronto.ca)

Mandarin Chinese lacks interdental fricatives, causing Mandarin Chinese learners to struggle in perceiving these sounds in English. This study explores how word position and vocalic context affect the perception of /θ/ by Mandarin Chinese learners. We test Mandarin Chinese learners and native English speakers using words containing the sound /θ/ in word-initial, medial, and final positions (e.g., think, nothing, bath), and in different vocalic contexts (e.g., think vs. thank). Participants complete a forced-choice identification task, listening to the target words in a carrier sentence and selecting the perceived word from two options. We record reaction time and accuracy for statistical analysis. We use generalized additive mixed modeling for reaction time and logistic mixed-effects modeling for response accuracy. We expect Mandarin Chinese learners to show longer reaction times and higher accuracy in perceiving /θ/ in the word-initial position than in media and final positions. Additionally, we anticipate that Mandarin Chinese learners perceive /θ/ more accurately when it is followed or preceded by a high vowel rather than a non-high vowel. Understanding these perception differences and contextual influences can inform more effective language teaching strategies, particularly in pronunciation training for Mandarin Chinese learners.

**3:30–4:00 Discussion**

## Session 1pSCd

# Speech Communication: Student Choose-Your-Own Adventure - Production

Keith Johnston, Chair
*University of California-Berkeley, Berkeley, CA*

### *Contributed Papers*

**3:00**

**1pSCd1. Acoustic variation in Buenos Aires /sC/ production: linguistic and social factors.** Amber Galvano (Linguistics, UC Berkeley, Dwinelle Hall #2650, Berkeley, CA 94704, amber_galvano@berkeley.edu)

/s/ debuccalization, or "pre-aspiration," is well-known in Buenos Aires (Porteño) Spanish. Recent research on Porteño /st/ suggests its most common variant for some young speakers contains both pre- and post-aspiration. Using sociolinguistic interview data from 20 speakers, I explore (i) the prevalence of lengthy VOT in /sC/; (ii) whether variants differ in HNR, intensity, or spectral moments; (iii) whether, accounting for coarticulation, frication in /st/ is different from /sp sk/; and (iv) the role of social factors. A clearer picture of Porteño /sC/ will inform our understanding of why developments in one variety (i.e., Andalusian) may or may not follow suit in another (i.e., Porteño). Intervocalic /sC/, either word-medial or cross-word, were analyzed ($n = 4084$). With 29ms as lengthy VOT, tokens were coded as: /s/-retention ($n = 355$); retention with lengthy VOT ($n = 59$); pre-aspiration ($n = 2941$); pre-aspiration with lengthy VOT ($n = 672$); or post-aspiration ($n = 33$). Note that pre-aspiration is dominant. Preliminary LMER modeling suggests fricative duration plus the interaction of fricative location (pre- versus post-closure) and place of articulation, as linguistic factors, and age, gender, sexuality, race/ethnicity, place of origin, and socioeconomic background, as social factors, are significant predictors of the acoustic variation.

**3:05**

**1pSCd2. Age-related acoustic changes in female speech production.** Siqi Li (Communication Sciences and Disorders, Northwestern Univ., 2240 Campus Dr., Evanston, IL 60208, siqili2028@u.northwestern.edu) and Pamela E. Souza (Communication Sciences and Disorders, Northwestern Univ., Evanston, IL)

Previous research has observed acoustic changes in aging speech caused by physiologic changes that occur during normal aging process, but with highly variable results across studies regarding what acoustic characteristics change and how (Tucker *et al*., 2021). Gender has been found to be one possible factor related to such variability (Torre & Barlow, 2009). Therefore, this study focuses on the age-related changes in female speech production specifically by comparing certain acoustic measures between younger and older female talkers. Participants ($n = 15 \times 2$ age groups) were instructed to read a set of sentences and recorded. Speech rates, fundamental frequencies (F0) and pitch variations (indexed as pitch velocity and pitch dynamicity; McCloy *et al*., 2015) were extracted for each talker. The results showed that older talkers tended to speak more slowly with lower mean F0, decreased F0 standard deviations, and decreased F0 ranges. The measures of pitch velocity and dynamicity are novel to the analysis of aging speech, and both turned out to be lower in older talkers, indicating more downdrift and smaller pitch excursions in their speech production. The implications of such acoustic changes for speech recognition will also be discussed.

**3:10**

**1pSCd3. Mapping the sounds of New Orleans: Acoustic measurements of vowel mergers.** Dana Serditova (English Dept., Univ. of Freiburg, Zaehringer St. 300, Freiburg 79108, Germany, dana.serditova@anglistik.uni-freiburg.de)

New Orleans is a city with a complex history, tumultuous social landscape, and distinctive linguistic features. Nevertheless, it shares some phonetic trends with other urban centers in the US South, such as the de-merger of vowel mergers (Tillery & Bailey, 2004; Labov *et al*., 2006; and Austen, 2020). This talk aims to discuss methods for acoustically measuring vowel mergers and to interpret the sociolinguistic implications of these measurements. I will examine the PIN∼PEN and the FEEL∼FILL mergers acoustically, drawing on a socially balanced and representative dataset of 115 sociolinguistic interviews. Using Pillai scores and Bhattacharyya's affinity to measure overlap, I will first demonstrate why the latter is preferred for statistical analysis. Additionally, for the FEEL∼FILL merger, generalized additive mixed models (GAMMs) are employed to model formant movement directly, and I will highlight the flexibility and nuanced nature of this methodology. Through acoustic analysis, linear mixed models, and GAMMs, I will show that there are overlapping trends influenced by factors such as age, gender, ethnoracial affiliation, vowel duration, and phonetic environment. Geographic variation in formant movement will be modeled directly and visualized on geospatial maps. This study advances our understanding of vowel mergers in the urban US South and provides a model for investigating merging in other socially diverse environments.

**3:15**

**1pSCd4. Phonetic imitation of a reduced voice onset time on word-initial voiceless stops by native English speakers.** Erica Dagar (Linguistics and TESOL, Univ. of Texas at Arlington, 21 Ridgewood Dr., Trophy Club, TX 76262-3404, erica.dagar@uta.edu)

This paper investigated whether native English speakers with exposure to a true voicing language (Spanish) would imitate a reduced voice onset time (VOT) on word-initial voiceless stops in English. Nielsen (2011) indicates native English speakers will not imitate a reduced VOT on word-initial voiceless stops since it is approaching the phonological boundary with voiced stops. However, there have been several studies that have found imitation toward a phonological boundary (Mitterer and Enertus 2008, Podlipsky and Simackova 2015). There were nine Native English speakers, some with Spanish language exposure. They completed an imitation study with word initial /p/ and /k/ words with a reduced and enhanced VOT in English. Results indicate that those with and without Spanish exposure do imitate a reduced VOT on word initial voiceless stops in English. The results from the English speakers without Spanish exposure conflict with previous studies that found that native English speakers will not imitate a reduced VOT on word initial voiceless stops to preserve the phonological boundary with voiced stops. The results indicate that native English speakers, whether they have exposure to a true voicing language or not, will approach the phonological boundary with voiced stops in a VOT imitation task.

**3:20**

**1pSCd5. Speech errors in fast speech of alveolar and alveo-palatal consonants: A preliminary study using ultrasound.** Misato Shimizu (Sophia Univ., 7-1 Kioi-cho, Chiyoda-ku, Tokyo, Japan, m.smz3310@gmail.com), Takayuki Arai (Sophia Univ., Tokyo, Japan), Ai Mizoguchi (Maebashi Inst. of Technol., Maebashi, Japan), Maho Morimoto (Chuo Univ., Tokyo, Japan), Weiyu Li (Sophia Univ., Tokyo, Japan), and Noriko Yamane (Hiroshima Univ., Hiroshima, Japan)

In this study, we observed speech errors that occurred when alternating between alveolar and alveo-palatal consonants during fast speech, through acoustic analysis and ultrasound tongue imaging. We selected phonologically parallel contrasting pairs in Japanese, /sa ɕa/ and /ta tɕa/. Three native Japanese speakers participated in the experiment. They completed a preliminary questionnaire and reported that despite having no history of speech or hearing disorders, they were aware of experiencing speaking difficulties. Participants were asked to pronounce bisyllabic phrases repeatedly in time with the metronome (BPM = 480). Their pronunciation was recorded using a microphone and ultrasound. The experimenter then auditorily judged each token as correct or errant pronunciation. We measured each consonant's spectral center of gravity (CoG) using Praat and analyzed ultrasound images of each consonant using AAA. We calculated the error rates for each token and found that Speaker 1 had a 44% error rate for both /sa ɕa/ and /ta tɕa/, Speaker 2 had a 25% error rate for /sa ɕa/ and 53% for /ta tɕa/, and Speaker 3 had a 27% error rate for /sa ɕa/ and 53% for /ta tɕa/. Additionally, the CoG and tongue contours during errors often differed from those during correct pronunciations.

**3:25**

**1pSCd6. Distinctions in negative voice onset time: Evidence from the labial velar stops of Gã (Niger Congo).** Danielle A. Quaye (Linguistics and TESOL, The Univ. of Texas at Arlington, 601 S. Nedderman Dr., 5th Floor University Hall #530, Box #19559, Arlington, TX 76010, danielle.quaye@mavs.uta.edu)

This study examines Voice Onset Time (VOT) in Gã, a Niger-Congo language, focusing on labial-velar stops (/kp/ and /gb/). Traditional approaches to VOT categorize stops into negative (voiced), short positive (voiceless unaspirated), and long positive (voiceless aspirated) areas. While the majority of literature states there are no distinctions in negative VOT, this research challenges that by finding distinct negative VOT subcategories in Gã. Using results from a field work study with 5 speakers, 159 tokens of /kp/ and /gb/ were manually measured. Both stops exhibit negative VOT, though in different ranges: /gb/ ranged from −5 to −348 ms (avg. −115 ms), while /kp/ ranged from −5 to −26 ms (avg. −14 ms). The implications are significant: this study disrupts traditional VOT norms by demonstrating distinct negative VOT categories, which is not discussed in phonetics. These findings suggest a need to reevaluate VOT categorizations, especially in relation to complex stop consonants like /kp/ and /gb/. Furthermore, this research underscores the importance of studying underrepresented languages like Gã to enrich our understanding of linguistic diversity and phonetic frameworks. Future research should explore how speakers perceive such sounds.

**3:30**

**1pSCd7. Acoustic comparison of rhotic acquisition in biofeedback versus motor-based treatment for residual speech sound disorder.** Marcela Lara (New York Univ., 246 Overlook Rd., Poughkeepsie, NY 12603, mpl6451@nyu.edu), Nina Benway, Megan Leece (Syracuse Univ., Syracuse, NY), Wendy Liang (New York Univ., New York, NY), Elaine Hitchcock (Montclair State Univ., Bloomfield, NJ), Jonathan L. Preston (Syracuse Univ., Syracuse, NY), and Tara McAllister (New York Univ., New York, NY)

Around 2%-5% of speakers aged 8 and up experience Residual Speech Sound Disorder (RSSD), which can persist despite years of intervention and may negatively impact a child's social and academic outcomes. Previous studies have investigated visual biofeedback as an alternative intervention for RSSD affecting the English rhotic /ɹ/. These studies have been inconclusive because of study design limitations and small sample sizes. Furthermore, most previous studies have focused on generalization gains at the end of a period of treatment, but theories of motor learning suggest that any differences between biofeedback and treatment will be most apparent early in treatment. Data for this study were sourced from an ongoing project, Correcting Residual Errors with Spectral, ULtrasound, Traditional Speech therapy Randomized Controlled Trial (C-RESULTS RCT), which aims to compare biofeedback and non-biofeedback interventions for RSSD affecting /ɹ/ using a well-powered randomized controlled trial design. In this study, eighty children with /ɹ/ misarticulation were randomized to receive traditional motor-based treatment or biofeedback treatment (with ultrasound and visual-acoustic biofeedback subtypes). We hypothesized that acoustic measures of /ɹ/ production accuracy would show faster gains in the first three intervention sessions (acquisition phase) for children who received biofeedback rather than motor-based treatment.

**3:35–4:00 Discussion**

## Session 1pSCe

# Speech Communication: Student Choose-Your-Own Adventure - Suprasegmentals and Prosody

Richard A. Wright, Chair
*University of Washington, Seattle, WA*

### *Contributed Papers*

**3:00**

**1pSCe1. Acoustic differences in prosody between primary and secondary emotions in English.** Sishi Fei (Dept. of Linguistics, The Univ. of Hong Kong, 9.30, Run Run Shaw Tower, Pokfulam Rd., Hong Kong, Pokfulam 999077, Hong Kong, sishi_fei@163.com)

Speech conveys emotions, with prosody decoding and representing emotional information through features like duration, intensity, and pitch. Emotions are classified as primary (e.g., happy) and secondary (e.g., encouraging). Primary emotions are innate, triggering instinctual responses, while secondary emotions involve complex cognitive processes and combinations of primary emotions. Previous studies have confirmed vocal differences across emotions, but differences between primary and secondary emotions in English are less explored. This study compares prosodic differences between primary emotions (sad, neutral, angry, excited, happy) and secondary emotions (apologetic, encouraging, assertive, concerned, anxious) in English. The research questions are (1) What are the differences in acoustic patterns between primary and secondary emotional prosody in English? (2) Which specific acoustic features correlate with these emotions? We analyze utterances by New Zealand English speakers from an emotional speech corpus, extracting prosodic features using the GeMAPS feature set in the Open-SMILE toolkit, which includes 23 acoustic parameters. Significant variations in prosodic features between primary and secondary emotions are expected, such as happy versus encouraging and excited versus anxious. We anticipate acoustic features like fundamental frequency, intensity, and duration to closely associate with these emotions, providing insight into acoustic markers of different emotions and enhancing applications in human-computer interaction.

**3:05**

**1pSCe2. Benefits of emotional prosody for sentence intelligibility and emotion recognition in speech-shaped noise: Acoustic and perceptual basis.** Jessica M. Alexander (Linguistics, Univ. of Texas at Austin, 305 E. 23rd St., STOP B5100, Austin, TX 78712, jessica.alexander@utexas.edu) and Fernando Llanos (Linguistics, Univ. of Texas at Austin, Austin, TX)

Speech naturally possesses emotional inflections, but the impact of vocal emotions on speech-in-noise processing is unclear. Here, we investigated the effects of emotional prosody on sentence intelligibility and emotion recognition across multiple noise levels. Participants ($N = 33$) transcribed semantically-neutral sentences produced in neutral, angry, and happy prosodies, embedded in speech-shaped noise at two SNRs (between-subjects: -12/-8, -10/-6 dB); participants then categorized the emotional valence of each sentence at the same two SNRs, as well as in quiet. Sentences produced with happy or angry prosodies were transcribed more accurately (linear mixed-effects: $p$ s<.05), revealing an intelligibility benefit for prosodically emotional speech. Transcription accuracy scores were further predictable ($r = .724$) from the speech-in-noise intelligibility index (ANSI S3.5-1997), suggesting that this benefit may be driven by a higher concentration of power in frequency bands that are more critical for intelligibility. Similarly, emotional valence categorization accuracy was higher for happy and angry prosodies (logistic mixed-effects: $p$ s<.001), with emotion-level differences modulated by SNR. Subsequent multidimensional-scaling analyses showed that perceptual distances between emotion categories warp at lower SNRs, leading to a negativity bias in categorization. In quiet, perceptual distances were robustly represented by relative differences in stimulus F0 variability and harmonic-to-noise ratio, compared to alternative cues tested.

**3:10**

**1pSCe3. The neural representation of emotional cues investigated using the speech frequency following response: A potential tool to evaluate speech prosody.** Maryam Karimi Boroujeni (School of Rehabilitation Sciences, Univ. of Ottawa, 200 Lees, Faculty of Health Sciences, Ottawa, Ontario K1S 5S9, Canada, mkari052@uottawa.ca), Sajad Sadeghkhani (School of Elec. Eng. and Comput. Sci., Univ. of Ottawa, Ottawa, Ontario, Canada), Saeid R. Seyednejad (Dept. of Elec. Eng., Shahid Bahonar Univ. of Kerman, Kerman, Iran (the Islamic Republic of)), Hilmi Dajani (School of Elec. Eng. and Comput. Sci., Univ. of Ottawa, Ottawa, Ontario, Canada), and Christian Giguère (School of Rehabilitation Sci., Univ. of Ottawa, Ottawa, Ontario, Canada)

Background: The Speech-evoked Frequency Following Response (sFFR) provides spctro-temporal data on speech processing in the auditory system. Its effectiveness in extracting prosodic features like variations in fundamental frequency (F0 contour) and intensity is uncertain. Objectives: This study examines how well sFFR tracks F0 contour in different emotions using a natural two-syllable word. It also explores talker's gender impact on F0 contours and gender disparity in encoding prosodic cues. Method: The word "balloon" spoken by male and female speakers with sad and happy emotions, elicited FFR from 16 adults (8 males, aged 18–31). A pitch estimation algorithm calculated root mean squared error and 5% accuracy to evaluate the response's fidelity to F0 contour under different conditions. Results: The sFFR tracked prosodic speech features, influenced by emotion type and talker voice characteristics. Participants identified emotions most accurately from sad male voices. Lower F0 trajectories corresponded to more reliable FFR responses, showing better tracking of male voices and sad emotions. No significant gender-related differences were observed in emotional data processing. Conclusion: These findings highlight sFFR's utility in capturing dynamic speech properties and its potential in clinical assessments. Future research should explore prosody processing in hearing-impaired individuals and consider integrating sFFR into diagnostic protocols.

**3:15**

**1pSCe4. Acoustic properties of utterance-final prosodic events in Embosi.** Yubin Zhang (Linguistics, Univ. of Southern California, Grace Ford Salvatori 301, Los Angeles, CA 90007, yubinzha@usc.edu), Yijing Lu (Linguistics, Univ. of Southern California, Los Angeles, CA), Annie Rialland (Lab. de Phonétique et Phonologie, Paris, France), Sarah Harper (Neurol. Surg. and Weill Neuroscience Inst., Univ. of California, San Diego, San Francisco, CA), and Louis Goldstein (Linguistics, University of Southern California, Los Angeles, CA)

In some Bantu languages, the utterance-final prosodic juncture has been found to exhibit multiple acoustic properties, like a low f0 and final devoicing (Rialland & Aborobongui, 2016; Zerbian, 2016). However, a detailed quantitative analysis of these acoustic properties is still lacking, and the

underlying causes are not well understood. To this end, we carry out an acoustic analysis of the utterance-final prosodic events in Embosi, a Bantu language spoken in the Republic of Congo. We analyze the mean f0, intensity and H1-H2* of the final three moras in an Embosi utterance. We hypothesize that the acoustic properties are correlates of the pulmonic pressure initiation dynamics, i.e., when subglottal pressure drops below the phonation pressure threshold towards the end of an utterance. Thus, f0 and intensity should be lowered and final-devoicing can also occur (Zhang, 2016). It is also likely that a glottal spreading or slackening gesture is present utterance-finally, reducing the vocal fold medial surface thickness. Then, the voice quality might become breathier, which can be characterized by a low f0 and an increase in H1-H2* (Zhang, 2016). The results show that utterance-finally both intensity and f0 are lowered but there is no evidence for an increase in H1-H2*. These findings are more consistent with the pulmonic pressure initiation hypothesis.

**3:20**

**1pSCe5. Phase-locking of oscillatory acoustic signals reflects syllable progression and variation.** Haley Hsu (Linguistics, Univ. of Southern California, 3601 Watt Way, Grace Ford Salvatori Hall 301, Los Angeles, CA 90089-1693, haleyhsu@usc.edu), Dani Byrd, Khalil Iskarous, and Louis Goldstein (Linguistics, Univ. of Southern California, Los Angeles, CA)

While a common linguistic view of the cognitive representation of speech is that it is composed of sequenced syllable units, how exactly syllables as abstract cognitive compositional structure relate to quantifiable patterns in the observable signals of articulation and acoustics remains opaque. Previous work has suggested that oscillatory acoustic properties can serve to link linguistic representations and physical events (Tilsen & Arvaniti 2013). We further probe this relationship by testing the temporal coordination between oscillatory signal measures—changes in spectral energy and in amplitude—and syllable boundary locations through the use of phase-locking analyses (Lancia *et al*. 2023). Results in both English and Tashlhiyt for vocalic and consonantal syllabic nuclei show significant phase-locking values (PLVs) and demonstrate that these signal measures track syllable progression across typologically different languages. Furthermore, the cross-language preferences in syllable nucleus types are reflected in their respective PLVs. Specifically, vocalic nuclei exhibit the highest PLVs, followed by wide aperture consonantal nuclei (sonorants), and lastly by consonantal nuclei with narrow-to-closed constrictions (obstruents). Overall, the findings demonstrate a tight coordination between abstract syllable units and quantifiable signal properties and additionally provide novel dynamical grounding for cross-linguistic nucleus preferences.

**3:25**

**1pSCe6. Neural oscillation to Mandarin lexical tone processing in bilingual children: A developmental study.** Rigel Baron (Communication Sciences and Disorders, St. John's Univ., 1025 Nugget St., Los Alamos, NM 87544, rigel.baron22@my.stjohns.edu), Kristal Reyes, Faith Chai, Angela Cheng, and Yan H. Yu (Communication Sciences & Disorders, St. John's Univ., Bayside, NY)

Tonal language experience is associated with structural and functional characteristics in the brain that underlie facilitated pitch perception. However, it is less understood what the developmental trajectory is for lexical tone processing at the cortical level, especially in bilingual children. In the present research, we investigated these developmental changes in terms of neuronal oscillations in response to Mandarin lexical tone contrasts. We measured lexical tone processing in bilingual English-Mandarin learning children and bilingual English-Mandarin young adults. Event-related brain potentials (ERPs) were recorded in an oddball paradigm with two tonal contrasts: one with a small acoustical difference, and one with a larger acoustical difference. We then analyzed the phase locking and amplitude modulation of ongoing oscillations in theta (4–8 Hz), alpha (8–14 Hz), and beta (14–30 Hz) bands to two types of lexical tone changes. Preliminary results suggest developmental changes were associated with strengthened phase locking of neural oscillations to both small and large acoustical contrasts, while stronger phase locking and higher power was observed in the larger acoustical contrast. Beyond this, the theta, alpha, and beta bands demonstrated different sensitivities to developmental changes. Such findings further our understanding of bilingual lexical tone development at the cortical level.

**3:30–4:00 Discussion**

## Session 1pSCf

# Speech Communication: Student Choose-Your-Own-Adventurei—Listening Environment, Instruments, and Tools

Matthew Kelley, Chair
*George Mason Univ. Fairfax, VA*

### *Contributed Papers*

**3:00**

**1pSCf1. Assessing children's classroom comprehension: Development of a closed-set listening comprehension test.** Ella Redden (Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, 901 S. Sixth St., Champaign, IL, eredden2@illinois.edu), Maria Angel, and Mary M. Flaherty (Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, Champaign, IL)

The effects of noise on children's word and sentence recognition have been extensively studied. However, the impact of noise on higher-level communication skills, such as listening comprehension, has received less attention. Currently, there are no widely available tests to assess children's listening comprehension in noise. Existing tests are often time-intensive and limited in trial number, restricting their utility for assessing comprehension across different conditions. This study aims to develop a test to measure children's sentence comprehension in noise. A closed-set format was chosen to ensure ease of testing and facilitate group assessment in real classrooms. The test comprises 50 sentences, each containing 10 words describing an object. For this study, target sentences were presented in both quiet and various noise conditions, using a four-alternative forced-choice picture-pointing procedure. Children (5-17 years) participated in test-retest rounds to establish reliability and feasibility. Working memory, receptive vocabulary, and language ability were also measured. Sensitivity analysis was employed to evaluate the test's ability to detect differences in comprehension across noise conditions. Item analysis was conducted to refine test items for clarity and difficulty. Preliminary results are promising and suggest the test could be a valuable tool for assessing listening skills in noisy classrooms.

**3:05**

**1pSCf2. Intelligibility of medically related sentences in quiet, speech-shaped noise, and hospital noise.** Sarah E. White (The Ohio State Univ., 6024 Nicholas Glen, Columbus, OH 43213, sarsar10green@gmail.com), Alex Holly (Univ. of Rochester, Chicago, IL), Tessa Bent (Speech, Language and Hearing Sci., Indiana Univ., Bloomington, IN), Erica E. Ryherd (Architectural Eng., Univ. of Nebraska - Lincoln, Omaha, NE), and Melissa Baese-Berk (Linguistics, Univ. of Chicago, Chicago, IL)

Noise levels in healthcare settings often exceed recommendations from health organizations. Previous work on the impact of noise on speech perception has been studied in classrooms and workplaces, showing decreased performance on working memory tasks and reduced intelligibility. However, the effect of noise on speech perception in medical settings has not yet been explored, even though successful communication between healthcare providers and patients is vital to quality healthcare delivery. Furthermore, previous studies typically focused on noise produced by buildings, equipment, or humansi—while all of these sources must be considered for noise within healthcare settings. This begs the question: how might hospital noise impact patients' ability to understand medical information? In this study, we explored perception of speech in conditions of quiet, hospital noise, and speech-shaped noise through an intelligibility task with participants aged 65 and older to understand the impact of hospital noise on speech perception. Additionally, we explored the role of word frequency and familiarity on speech intelligibility in noise by utilizing both standard sentences and medically related sentences. Finally, we collected data on participants' hearing ability, cognitive acuity, and experience in hospital settings, to explore additional factors that may interact with speech perception in noise. [Work supported by James S. McDonnell Foundation.]

**3:10**

**1pSCf3. Effect of non-human avatars on opinion convergence in remote spoken interactions.** Jiu Song, Charlize Ma, Raechel Kitamura (Dept. of Linguistics, Univ. of British Columbia, Vancouver, British Columbia, Canada), Alejandra Blanco (Dept. of Linguistics, Univ. of British Columbia, Totem Field Studios, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, alejandra.blanco.email@gmail.com), Liang Kai Fong, Jahurul Islam, and Bryan Gick (Dept. of Linguistics, Univ. of British Columbia, Vancouver, BC, Canada)

Opinion convergence in spoken interactions is influenced by various factors, such as interlocutor traits, social identity, and context [Pardo, 2022, *JPhon* 95]. Previous studies on communication via Zoom conferences have reported that differences in opinion can affect visual cues in vowel productions [Ma *et al.*, 2023, HISPCSL]. However, there is a general lack of understanding of how technology-mediated communication affects opinion convergence. This study investigated whether the availability of visual cues in online video communication has an effect on opinion convergence between people. We collected data from fifty college students (aged 18–30, with an average age of 19.96) who were randomly paired into 25 groups to participate in a 20-min Zoom discussion on a range of debatable propositions. The discussions were conducted using two conditions: no camera (black screen) or fox avatars set up by Zoom. Participants' opinions on the propositions were measured before and after the discussion. We will measure whether a difference in avatar vs. non-avatar conditions impacts the level of opinion convergence between pairs of speakers by comparing the changes in the levels of convergence between before and after conversations. Implications for the general effects of technology-mediated communication on opinion formation and convergence will be discussed.

**3:15**

**1pSCf4. The influence of firefighting experience on speech-in-noise recognition with two-way radios.** Abraham Chavez (Speech and Hearing Science, Univ. of Illinois at Urbana-Champaign, 901 S. Sixth St., Champaign, IL, ac108@illinois.edu), Tania Aguilar, and Mary M. Flaherty (Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, Champaign, IL)

Clear communication is critical for firefighters' safety, but high noise levels, hearing loss, and protective equipment impair speech understanding. This study investigates the performance of two-way radios in noise and examines their effects on speech-in-noise recognition related to noise levels, hearing thresholds, and firefighting experience. We first characterized the acoustic performance of Motorola APX 7000xe radios using MATLAB. Speech recognition was then tested using the Quick-SIN with two listener groups ($n = 12$ firefighters; $n = 20$ non-firefighters). Target sentences, recorded with KEMAR simulators, were presented in quiet and fire noise at fixed SNRs ($-9$ and $-15$ dB), yielding four test conditions. Realistic fire noise was simulated using

auditory steering to create a surround sound effect relative to the radio position. Listeners repeated sentences heard over headphones and were scored on correct keywords. Preliminary control group results show significant speech intelligibility decline with radio use, even in quiet. In noise, performance dropped further, with 59% correct at −9 dB SNR and 40% at −15 dB SNR. The radio system's bandpass filter prioritized speech frequencies (1700-3200 Hz) but still posed challenges. Data from 12 firefighters are currently being analyzed to assess how hearing thresholds and experience affect performance, guiding future communication strategies.

**3:20**

**1pSCf5. A comparison of voice quality measures across in-ear, outer-ear, and standard microphone recordings.** Xinyi Zhang (Elec. Eng., École de technologie supérieure, 1100 Notre-Dame St. W, Montreal, Quebec H3C 1K3, Canada, xinyi.zhang.1@ens.etsmtl.ca), Alessandro Braga, Arian Shamei, and Rachel Bouserhal (Elec. Eng., École de technologie supérieure, Montréal, Quebec, Canada)

Voice quality provides significant insights into one's health. Recent advancements in intra-aural devices have enabled the longitudinal monitoring of speech and its changes. These wearables can record speech from in-ear microphones (IEMs) and outer-ear microphones (OEMs). The laboratory gold standard of speech recording uses a high-sensitivity low-noise microphone placed in front of the mouth to capture signals accurately. Speech recorded inside an occluded ear canal differs considerably from this due to the bone-and-tissue conduction and the effect of ear occlusion. OEMs record speech transmitted solely through air like the standard microphones but are more influenced by indirect air conduction due to their positioning. This study compares voice quality measures across IEM, OEM, and the standard microphone (REF) using an open-access database. We employed linear mixed-effect modeling to analyze the effects of different microphone recordings on these metrics. Results indicate that while pitch control remains relatively intact, IEM and OEM exhibit varying deviations from REF. Overall, OEM resembles REF in jitter, whereas IEM is higher. IEM resembles REF in shimmer, while OEM is lower. For HNR, both IEM and OEM are higher than REF. Sex-based difference was also observed, and the correlation between microphone differences and fundamental frequency was explored.

**3:25**

**1pSCf6. The AnySpeech Project —Open-vocabulary keyword spotting and phonetic transcription in any language.** Farhan Samir (Univ. of British Columbia, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, fsamir@mail.ubc.ca) and Jian Zhu (Univ. of British Columbia, Ann Arbor, MI)

Performing multilingual phonetic analysis is challenging, not least due to the lack of performant tools for performing basic analyses on (non-English) speech data. Without the capacity to perform basic operations such as transcription or search through speech data for understudied languages, critical questions in cross-linguistic phonetic analysis remain elusive [Blasi *et al*. Trends in Cog. Sci. 26(12), 1153–1170 (2022)]. To this end, we will demonstrate our recent advances towards multilingual representation learning and automatic phonetic transcription, as part of our AnySpeech initiative. Specifically, we will demonstrate two tools. First, the CLAP-IPA model—a phoneme-to-speech model, capable of performing open-vocabulary keyword spotting in any language without any parameter updates, including languages that were not in the training dataset [Zhu *et al*., NAACL, 750-772 (2024)]. Second, the IPAPack transcription model, a lightweight transcription model, capable of annotating 473 different phones, including those that were lacking in prior models (e.g., click consonants in Bantu languages). We will demonstrate that our models can be easily downloaded and set up on consumer-grade laptops with little effort. We anticipate that our tools will enable researchers to analyze speech recording repositories at scale, unlocking answers to critical questions in cross-linguistic phonetic analysis.

**3:30**

**1pSCf7. Extracting low-dimensional signals from X-ray microbeam and magnetic resonance imaging articulatory data.** Jessica Campbell (Linguistics, Univ. of Southern California, 3601 Watt Way, Grace Ford Salvatori Hall 301, Los Angeles, CA 90089, jac95339@usc.edu), Haley Hsu, and Louis Goldstein (Linguistics, Univ. of Southern California, Los Angeles, CA)

Low-dimensional signals derived from speech allow for more efficient computational analysis than do raw speech signals and have been argued to be involved in neural synchronization to speech (e.g., Assaneo & Poeppel 2018, Giraud *et al*. 2007). While there are multiple such signals derived from the acoustic domain, little research focuses on low-dimensional signals derived from *articulation*, which provide a global characterization of movement of the vocal tract (c.f. Orlando & Palo 2023 and Poeppel & Assaneo 2020). We discuss one such signal, the articulatory modulation function (Goldstein 2019 and Campbell *et al*. 2023). In-progress research has found reduced stability of articulatory modulation in speakers with ALS; thus, the signal may serve as an index of inter-articulator coordination. In this tutorial, we describe uses for and methods of extracting low-dimensional signals from X-Ray Microbeam (XRMB) data (Westbury *et al*. 1994) and real-time MRI data. While a standard method using XRMB data has been established, no such standard exists for MRI data. We discuss the advantages and disadvantages of two MRI approaches: (1) pre-segmented contours and (2) principal component analysis. [Work supported by NIH and NSF.]

**3:35–4:05 Discussion**

## Session 1pUW

# Underwater Acoustics: Lightning Round Led by Students

Natalie Kukshtel, Chair
*Applied Ocean Physics & Eng., Woods Hole Oceanographic Insti., 98 Water St., Woods Hole, MA 02543*

### *Contributed Papers*

**3:00**

**1pUW1. Leveraging passive acoustic identification AID tag for autonomous underwater vehicle docking.** Nizar Somaan (Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr. NW, J. Erskine Love Building, Office 131 & Rm. 130, Atlanta, GA 30332, nsomaan3@gatech.edu), Ananya Bhardwaj, and Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

Accurate positioning is essential for the navigation of autonomous underwater vehicles (AUVs), especially during docking and homing. This presentation explores the feasibility of using passive Acoustic Identification (AID) tags for precise AUV localization at a docking station. As a proof of concept, scaled experiments used AID tags made of four concentric hemispherical acrylic layers, each creating a unique backscattered acoustic signature when ensonified by a broadband ultrasonic transducer in GaTech water tank. The tags in these tests have a 9-in. outer diameter and were tested at a frequency range of 700 kHz to 1.3 MHz over a distance of up to 6 m. To enhance detection in cluttered environments, a parameterized signal processing method was employed. Additionally, we will present analysis of long-range experiments to assess the tags' detectability at greater distances using larger AID tags with outer diameter of up to 24-in. outer diameter. These tests will operate at a frequency range of 100 to 600 kHz, aiming to reach distances of tens of meters. Implications for the performance of AID tags operating under real-life conditions will be discussed.

**3:05**

**1pUW2. Effect of spatial distribution of shallow water explosive sources over a muddy seabed on vector acoustic measures.** Robert W. Drinnan (Applied Physics Lab., Univ. of Washington, 1013 NE 40th St., Box 355640, Seattle, WA 98105-6785, drinnan@uw.edu), Peter H. Dahl, and David Dall'Osto (Applied Phys. Lab., Univ. of Washington, Seattle, WA)

Vector acoustic observations of underwater sound provide several properties of the sound field not accessible in pressure data. The sensor known as Intensity Vector Autonomous Recorder (IVAR) coherently measures pressure and 3-axis acceleration, enabling estimates of both kinetic and potential energy, and other vector acoustic measures involving the coherent combination of acoustic pressure and velocity. IVAR measurements of signals originating from broadband explosive sources (SUS charges) made during the Seabed Characterization Experiment (SBCEX) are analyzed. The SUS deployments made on and around the perimeter of the New England Mud patch (NEMP), a thick depression of mud sediments on the continental shelf roughly 95 km south of Cape Cod, reveal a spatial variation of the energetic and vector acoustic properties. Changes in bathymetry, mud thickness, and sediment layer composition along the source-receiver track are modeled and compared to measured data. The advantage of using these vector properties for geoacoustic inference is discussed.

**3:10**

**1pUW3. An acoustic dataset augmentation generator for ship noise classification.** Dugald Thomson (Dept. of Oceanography, Dalhousie Univ., Rm. 3635-1355 Oxford St., Halifax, Nova Scotia B3H 4R2, Canada, dugald@dal.ca), Jessica Topple (DRDC, Halifax, Nova Scotia, Canada), and David R. Barclay (Oceanography, Dalhousie Univ., Halifax, Nova Scotia, Canada)

Automated target recognition using ship noise has advanced rapidly with machine learning, yet remains constrained by a limited number of benchmark datasets due to a scarcity of labeled data. This study aims to enhance ship classifier performance by addressing dataset limitations through an innovative Acoustic Dataset Augmentation Generator. Leveraging acoustic data from Arctic hydrophones and incorporating simulated ship noise via a Passive Sonar Simulator alongside ambient noise variations, this generator simulates diverse environmental conditions to enrich training datasets. By filling gaps and introducing realistic noise sources, it aims to improve model robustness and performance. The research evaluates model effectiveness using the augmented dataset, focusing on identifying and mitigating misclassifications.

**3:15–3:30 Discussion**

**3:30**

**1pUW4. Send the important stuff back: An innovative approach to onboard stream processing for remotely deployed passive acoustic monitoring systems.** Joseph Ross (Oceanography, Naval Postgraduate School, 301 South Rd, Rm. 414, Chapel Hill, NC 27514, josephross004@gmail.com), Tetyana Margolina, and John E. Joseph (Oceanography, Naval Postgraduate School, Monterey, CA)

Real-time monitoring of underwater soundscapes is necessary for rapid assessment of oceanic ecosystem health, controlling anthropogenic acoustic pollution, and establishing multi-scale spatiotemporal variability of the ocean acoustic environment. Such an assessment requires progression from existing regional and specialized passive acoustic monitoring (PAM) observational systems to a large-scale systematic passive acoustic monitoring of the World Ocean, complementary to the ARGO float Integrated Marine Observing System. This research develops an innovative approach to onboard stream processing of acoustic data on underwater autonomous mobile platforms using PyPAM, a Python-based package for calculating hybrid millidecade spectra and statistics. The design optimizes limited computational resources while preserving sufficient information to characterize the sampled soundscape's noise levels and assemblage of sound sources. The association between frequency-temporal acoustic signatures of biophonic, geophonic, and anthropophonic sounds and corresponding features in percentiles of Spectral Probability Density have been established from PAM data collected at stationary receivers and test deployments of PAM-enabled underwater profilers. Financial support to Joseph Ross was provided by

ASA SURIEA. Hardware development was supported by NPS CRUSER program. The PyPAM software is developed by Clea Parcerisas (LifeWatch). The research is conducted at NPS in collaboration with John Ryan (MBARI) and Josh Laney (Seatrec Inc.).

**3:35**

**1pUW5. Analysis of underwater radiated noise from ships using distributed acoustic sensing technology.** Erfan B. Horeh (School of Oceanography, Univ. of Washington, 4746 11th Ave. NE #202, Seattle, WA 98105, erfanbh@uw.edu), Shima Abadi (School of Oceanography, Univ. of Washington, Seattle, WA), and William S. Wilcock (School of Oceanography, Univ. of Washington, Seattle, WA)

Distributed Acoustic Sensing (DAS) technology enables continuous monitoring of acoustic vibrations along fiber optic cables, providing high-resolution spatial and temporal data. This study explores the application of DAS technology for analyzing underwater radiated noise from ships. In November 2021, four days of DAS data were collected using two cables from the Ocean Observatories Initiative Regional Cabled Array, extending offshore central Oregon. Numerous ship passages occurred over these cables, with information available through the Automatic Identification System (AIS). DAS data were collected using two different interrogators on two fibers in each cable, providing an opportunity to investigate the differences in ship noise detected on different fibers within the same cable and across different DAS systems with varying configurations. Preliminary analysis of two large ships shows that their noise is clearly detected in the DAS measurements. By combining these measurements with AIS data, we can accurately locate the receiving channels on the cable. This talk will expand on these observations to perform a comparative analysis of the measurements across different fibers and interrogators. These findings demonstrate the capability of DAS technology in maritime acoustic monitoring and its potential to enhance our understanding of underwater noise pollution.

**3:40**

**1pUW6. Localizing North Atlantic Right Whales using a deformable sonobuoy grid.** Kamden P. Thebeau (Dept. of Oceanography, Dalhousie Univ., PO BOX 15000, Halifax, Nova Scotia B3H 4R2, Canada, kamden.thebeau@dal.ca), David R. Barclay (Oceanography, Dalhousie Univ., Halifax, Nova Scotia, Canada), and Carolyn Binder (Defence Research and Development Canada, Dartmouth, Nova Scotia, Canada)

In 2018, a large-scale data collection effort was conducted in the Gulf of St. Lawrence, a known feeding ground for North Atlantic Right Whales (NARW), over two days. On each day, visual surveys were conducted, 32 sonobuoys were deployed to gather directional acoustic time series, and a Slocum glider operated in the area to collect oceanographic data. Following the collection phase, the acoustic data were manually annotated with a focus on NARW vocalizations. This project uses the multi-modal dataset to test and compare the results of three localization algorithms for NARW calls. The first method of localization used the directionality of the calls, where probability density maps were created by overlapping the bearing statistics across each of the relevant sonobuoys, commonly referred to as cross-fixing. To improve the bearing distributions, the signals were isolated with thresholding after using a conditional whitener and power-law statistic on its short-time Fourier transform. In the next approach, localization was performed using a spherical interpolation method to initialize a maximum likelihood time-difference-of-arrival algorithm. Finally, matched-field processing was used to model replica fields at potential source locations and correlate the replicas with the received pressure on the hydrophones.

**3:45**

**1pUW7. Seafloor quantification and characterization using multibeam echosounder backscatter data.** Henry Manik (Dept. of Marine Science and Technol., FPIK IPB Univ., BogorWest Java, Indonesia), Dadang Handoko (Marine Technology Study Program FPIK IPB Univ., Bogor, Indonesia, handoko.dadang@apps.ipb.ac.id), M Fadhil Ilham, and Moh Rafif Rabbani (Marine Technology Study Program FPIK IPB Univ., Bogor, Indonesia)

Indonesia has many small islands to manage for sustainable development goals. The urgency of this research lies in seafloor classification technology using acoustic technology. Seafloor classification plays a crucial role in mapping and monitoring marine ecosystems, marine resource management, research, military operation, and engineering activities. We used multibeam echosounder (MBES) for seafloor quantification and characterization in Pari Island waters, North Jakarta, Indonesia. MBES instrument provided bathymetry information and backscatter data. The bathymetry data provide seafloor depth while the backscatter data are used to classify seafloor sediment. The sediment grab sampling were taken to validate the seafloor acoustic backscatter. We found in this study area, the measured acoustic backscatter strength of MBES is a function of the signal frequency, incident angle, and sediment type.

**3:50–4:05 Discussion**

## Session 2aAA

# Architectural Acoustics: Heating, Ventilation and Air Conditioning (HVAC) Noise Challenges and Solutions

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

Joseph Keefe, Cochair
*Ostergaard Acoust. Associates, 1460 US Highway 9 North, Ste. 209, Woodbridge, NJ 07095*

### *Invited Papers*

**10:00**

**2aAA1. Propagation of uncertainty in HVAC noise prediction.** Samuel H. Underwood (Durham School of Arch. Eng. and Construction, Univ. of Nebraska-Lincoln, 1110 S 67th St., Omaha, NE 68182-0816, samuelunderwood@huskers.unl.edu)

Predictions of noise generated by Heating, Ventilation, and Air Conditioning (HVAC) systems bear uncertainty that can be estimated from the uncertainty associated with each input variable in the source-path-receiver calculation. In this study, the propagation of uncertainties from the measurement of source sound power levels, empirically modeled attenuation, regenerated noise from duct path elements, and receiver room corrections are estimated for a simple duct system example. Additional attention is given to other "known unknowns" in HVAC noise prediction which still remain unaddressed decades after they were first identified. Practical implications of uncertainty are also discussed.

**10:05**

**2aAA2. Honing data and calculations with scrutiny, judgment, and experience.** K. Anthony Hoover (McKay Conant Hoover, 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362, thoover@mchinc.com)

Noise control evaluations should involve more than simply inputting received data into an algorithm. If the data looks suspect, it deserves further review. Each element in the system might deserve some judgment about how it fits into a calculation scheme. There are usually several possible approaches to evaluate a particular system. Various added concerns, such as pressure drops, and space limitations, need to be considered. Scrutiny, judgment, and experience tend to enhance each other, leading to evaluations that become much more efficient and reliable.

### *Contributed Paper*

**10:10**

**2aAA3. Analysis of methods for converting mechanical equipment manufacturers' sound pressure spectra to sound power spectra.** Spencer Zack (Resonance Acoust., 364 Bush St., Floor 2, San Francisco, CA 94104, szack@resonanceac.com), Matthew Christian, and Peter Holst (Resonance Acoust., San Francisco, CA)

To accurately predict mechanical noise levels, obtaining mechanical unit sound power spectra is critical. However, some mechanical equipment manufacturers only publish sound pressure level data. Furthermore, the measurement standards ISO 3745 and JIS 8616, used by many international HVAC equipment providers (e.g., LG, Daikin, Mitsubishi), combine radiated, inlet, and discharge noise into a single dataset. It becomes necessary to determine the sound power level for each path (i.e., discharge, inlet, and radiated) from this single set of sound data. As a result, sound pressure-to-power conversion methods can frequently yield inaccurate results. A more robust calculation procedure is required to increase the precision of the sound pressure to sound power conversion. This study focuses on the methods used to estimate sound power spectra from international HVAC manufacturer-provided sound pressure levels by applying acoustic principles and standards. This presentation discusses the development process of these tools, the theoretical basis of the conversion method, and the validation results.

## *Invited Papers*

**10:15**

**2aAA4. Preliminary testing for ASHRAE research project 1919: The effects of duct size and aspect ratio on flow noise in elbows.** Raine Stewart (Research & Development, Vibro-Acoustics, by Swegon, 3 Keensford Ct., Unit 1, Ajax, Ontario L17 0K4, Canada, raine.stewart@swegon.com) and Karl Peterman (Research & Development, Vibro-Acoustics, by Swegon, Ajax, Ontario, Canada)

The goal of RP-1919 is to investigate the regenerated noise and breakout noise of elbow fittings, eventually resulting in an algorithm for a variety of flow rates that considers a duct's size and aspect ratio. This algorithm is intended for more accurate noise prediction of elbow fittings, to better inform building design. The practical application of this project necessitates a test method that is applicable to real-world building systems. Sound intensity measurements are a logical choice to investigate the elbow fittings, with one notable challenge: differentiating airflow noise from elbow fitting noise, especially at high flow rates that an intensity probe would not typically accommodate. Preliminary testing has been done to investigate the use of a blimp style windscreen and faux-fur muff with intensity scanning; this windscreen configuration can be seen used in film and TV production to get high quality audio where extreme conditions may affect sound. This presentation will discuss initial findings from the windscreen investigation.

**10:20**

**2aAA5. Whistle while you ductwork: Predicting air leakage noise.** Brandon Cudequest (Threshold Acoust., 141 W Jackson Blvd., Ste. 2080, Chicago, IL 60604, bcudequest@thresholdacoustics.com)

Improperly sealed ductwork creates system inefficiencies and noise. When leaky ducts are exposed to occupied spaces, these radiated noises can be a significant nuisance as their characteristic is often hissy, tonal, and whistle-like. This presentation will offer an empirical method for predicting noise resulting from air leakage. The algorithm only requires knowledge of the duct velocity and size. Example scenarios and limitations of the method will be presented.

**10:40**

**2aAA6. When noise mitigation of rooftop units does not work as expected.** David Manley (DLR Group, 6457 Frances St., Omaha, NE 68106, dmanley@dlrgroup.com) and Blake Krapfl (DLR Group, Minneapolis, MN)

A duct silencer was installed in the return air ductwork of a rooftop unit as part of noise mitigation in an existing building. After installation, the resulting noise levels had only been reduced by approximately half of what was calculated. This presentation will discuss the strategies the team took to investigate why the results did not match the calculations and how the team ultimately discovered the outdoor air damper settings of the rooftop unit were the cause of the noise problem.

**10:45**

**2aAA7. Vertical unit ventilator headaches.** Joseph Keefe (Ostergaard Acoust. Associates, 1460 US Highway 9 North, Ste. 209, Woodbridge, NJ 07095, jkeefe@acousticalconsultant.com)

Electrification retrofit efforts are underway in New York City public schools. In order to mitigate greenhouse gas emissions, these projects involve the removal of natural-gas-fired HVAC equipment and the installation of electric heat pumps serving classrooms and other school spaces. Although ducted air distribution is often a preferred HVAC system design for new construction, particularly for acoustics, such configurations are generally not feasible for existing buildings. Accordingly, in-room HVAC devices are common for electrification projects, which introduces noise challenges. This presentation focuses on the discoveries and challenges associated with the design of a few projects using vertical unit ventilators in school classrooms.

**10:50**

**2aAA8. Radiant panels and chilled beams for thermal and acoustical design.** Jessica S. Clements (Acoust., SmithGroup, Atlanta, GA) and Kirsten Barringer-Cook (Acoust. Studio, Newcomb & Boyd, LLP, 303 Peachtree Ctr. Ave. NE, Ste. 525, Atlanta, GA 30303, kbarringer-cook@newcomb-boyd.com)

Radiant panels and chilled beams are becoming more prevalent in projects using open ceilings and low velocity airflow designs. Radiant panels are being considered with open ceiling designs for thermal efficiency and modern appearances. These panels also offer options to include acoustical absorption, however, which brings challenges that must be addressed with the mechanical engineer and the acoustical consultant. Chilled beams promise better thermal regulation and low created noise, but those promises rely heavily on the installation. This lightning presentation will endeavor to give a high-level review of these two products and how to consider them in your design.

**10:55**

**2aAA9. Identifying and mitigating resonance from a valve serving hibernaculum.** Joseph F. Hackman (Newcomb & Boyd, LLP, 303 Peachtree Ctr. Ave. NE, Ste. 525, Atlanta, GA 30303, jhackman@newcomb-boyd.com) and R. Bruce DeRoo (Newcomb & Boyd, LLP, Centennial, CO)

This case study examines the challenges related to identifying and mitigating unwanted noise from a mechanical system serving hibernaculum, used for lemur torpor cycles. To support the torpor cycle, the design team determined a steady state background noise goal. Once installed, intermittent noise was discovered that the lemur research team felt might interfere with the rest cycle negatively affecting the lemurs and the results of any research. This presentation will review the investigation team's methodology in identifying the main source of the noise, which was a coolant fluid value serving a fan powered box, review some of the measurement results, and discuss the solutions determined with the chamber design team. The implementation of the mitigation solution is underway and may be included in the presentation if completed by the time of presentation.

**11:00–11:15 Discussion**

## Session 2aAB

## Animal Bioacoustics: Animal Bioacoustics Virtual Lab Tours

Micheal Dent, Cochair
*Univ. at Buffalo, Suny, B76 Park Hall, Buffalo, NY 142260*

Laura Kloepper, Cochair
*Dept. of Biol. Sci., Univ. of New Hampshire, 230 Spaulding Hall, Durham, NH 03824*

**Chair's Introduction—10:00**

### *Invited Papers*

**10:05**

**2aAB1. A look inside the Comparative Bioacoustics Laboratory: Mousing around.** Sevda Abdavinejad (Univ. at Buffalo, SUNY, 206 Park Hall, Buffalo, NY 14260, sevdaabd@buffalo.edu), Riley McLaughlin, and Micheal Dent (Univ. at Buffalo, SUNY, Buffalo, NY)

Deep in the basement of the Psychology building at the University at Buffalo, some tenacious researchers are attempting to unravel the mysteries of animal communication. The laboratory mouse is the most recent creature of interest for the Dent Lab. Mice produce ultrasonic vocalizations with many similarities to human speech. That, and the fact that laboratories around the world are using genetically modified mice to understand more about the contributions of various genes on diseases, has made many want to understand more about the mouse's umwelt. In our laboratory, we focus on mouse bioacoustics. We record vocalizations produced by mice following different types of social encounters. We present these vocalizations and other stimuli to determine various choices by the mice in traditional preference experiments. Finally, most of our experiments involve operant conditioning to determine what mice perceive. In our tightly controlled psychoacoustic experiments, mice are trained to nose poke to tell us what they can detect, discriminate, and categorize. We have used these methods to compare the effects of age, sex, strain, and noise on hearing acuity. Through this combination of experiments, we are beginning to understand the bioacoustics of mice and other animals. [Work supported by NIH AG081747.]

**10:15**

**2aAB2. Welcome to the Bat Lab at Johns Hopkins University!.** Nikita Finger, Davi Drieskens (Psychol. and Brain Sci., Johns Hopkins Univ., Baltimore, MD), and Cynthia F. Moss (Psychol. and Brain Sci., Johns Hopkins Univ., 3400 N. Charles St., Ames 200B, Baltimore, MD 21218, cynthia.moss@jhu.edu)

Welcome to a virtual tour of the JHU BatLab! Meet the animals at the heart of our research, echolocating bats, and get an inside look into our research on auditory perception, attention, memory, and the processes enabling social interactions, target tracking, and navigation in complex 3D environments. Echolocating bats coordinate sonar signal production with flight maneuvers in response to dynamic echo information and exhibit a rich array of natural sensory-guided behaviors. In this tour, we demonstrate the use of equipment, such as microphone arrays and high-speed video cameras, to quantify and analyze adaptive echolocation behaviors and 3D flight kinematics in two bat species, the insectivorous big brown bat and the Egyptian fruit bat. These species differ in their natural behaviors and modes of echolocation. Egyptian fruit bats navigate long distances to their foraging sites and use tongue-click echolocation, along with vision, to find food and travel to roosts. Big brown bats typically forage close to their roosting sites and use echoes from sonar vocalizations to find small insect prey in the dark. Join us to explore bat's use of echolocation and gain a deeper appreciation for the research that unlocks the secrets of these remarkable creatures and their sensory-guided behaviors.

**10:25**

**2aAB3. Syracuse University Bioacoustics and Behavioral Ecology Laboratory.** Susan Parks (Dept. of Biol., Syracuse Univ., 107 College Place, Syracuse, NY 13244, sparks@syr.edu), Dana Adcock, Dana Cusano, Alex Eschmann, Valeria Perez-Marrufo, Maya Philipp, Melanie Smith, and Sara Tennant (Dept. of Biol., Syracuse Univ., Syracuse, NY)

The Bioacoustics and Behavioral Ecology Laboratory at Syracuse University studies the ecology and evolution of acoustic signaling. Diverse research topics in the lab span the fields of behavioral ecology, bioacoustics, biological oceanography, and conservation biology. The current projects in the lab involve studies of both marine and terrestrial animals ranging from characterization of the acoustic behavior of species to experimental studies investigating the behavioral functions of sounds and the impacts of noise on communication. Research interests include studying the use of sound for communication, the evolution of acoustic signals, soundscapes, hearing abilities, and the impacts of noise on development, behavior, sound production and reception. This tour will introduce current lab group members and give a tour of the facilities, equipment, and software we utilize to undertake our research endeavors.

**10:35**

**2aAB4. Diverse models, unified goals: Discovering auditory structure and function in the Lauer Lab.** Sergio Vicencio-Jimenez (Otolaryngology-HNS, Johns Hopkins Univ., School of Med., 521 Traylor Bldg., 720 Rutland Ave., Baltimore, MD 21205, vicenciojimenez@gmail.com), Srijita Paul, Suhani Aggarwal, and Amanda M. Lauer (Otolaryngology-HNS, Johns Hopkins Univ., School of Med., Balitmore, MD)

Lauer Lab members conduct a wide range of experiments designed to understand the complex relationships between the structure and function of the auditory system and its role in health and disease. You might see experimenters solving questions about the effects of hearing loss on brain function and behavior, in natural aging, and in diseases such as Alzheimer's. On an ordinary day, you would see these questions being answered from multiple complementary perspectives. You would find scientists working on laboratory and wild mice and tissue specimens from a range of species such as bats, marmosets, all the way up to humans. You could see that this variety of models is matched only by the diversity of ways they study them, providing plenty of opportunities for any researcher to find tools that they find fascinating. From anatomical approaches, like confocal microscopy to investigate the brain and cochlea, physiological techniques to measure hearing including auditory brainstem responses and otoacoustic emissions, optogenetics to study auditory efferents, and behavioral studies investigating the role of hearing in behavior, cognition, and affective states. Join us for a virtual tour and witness the exciting discoveries unfolding in the Lauer Lab! [Work supported by Rubenstein Fund for Hearing Research (R03AG081747 and R01DC016641). ]

**10:45**

**2aAB5. The Brown BatLab.** James A. Simmons, Reese N. Fry (Neuroscience, Brown Univ., Providence, RI), and Andrea M. Simmons (Cognitive & Psychol. Sci., Brown Univ., 190 Thayer St., Box 1821, Providence, RI 02912-9067, Andrea_Simmons@brown.edu)

In the Brown BatLab, we study biosonar in big brown bats using psychophysical and behavioral techniques. In this presentation, we will show how bats are trained to perform in two-alternative forced choice psychophysical experiments, and how their attention to and localization of sounds can be studied using this technique. We will show videos of bats flying through obstacle arrays in a flight room and how their biosonar sounds change with the density of obstacles. We will show the output of an acoustic camera that localizes the bat's broadcasts and echoes as it is performing these tasks in the laboratory and as it flies with conspecifics outdoors in the natural environment.

**10:55**

**2aAB6. Bioacoustic studies of amphibious mammals at the Cognition and Sensory Systems Laboratory.** Seri Aldana (Inst. of Marine Sci., Univ. of California Santa Cruz, 115 McAllister Way, Santa Cruz, CA 95060, staldana@ucsc.edu), Kiran Jagait, Hannah Jackson (Inst. of Marine Sci., Univ. of California Santa Cruz, Santa Cruz, CA), Noah Packard, Ryan A. Jones (Ocean Sci., Univ. of California, Santa Cruz, Santa Cruz, CA), Brandi Ruscher (Ocean Sci., Univ. of California, Santa Cruz, Santa Cruz, CA), Caroline Casey, Jillian M. Sills, and Colleen Reichmuth (Inst. of Marine Sci., Univ. of California Santa Cruz, Santa Cruz, CA)

The Cognition and Sensory Systems Laboratory at the University of California Santa Cruz is a unique facility dedicated to exploring the acoustic biology of amphibious marine mammals. For over 40 years, our laboratory has combined psychoacoustic and bioacoustic approaches to study marine mammal sound reception and production. Through collaboration with colleagues and facilities around the world, we have used behavioral methods to evaluate hearing and masking profiles for seals, sea lions, walruses, and sea otters. We also work to understand vocal behavior in the lab and field by classifying the acoustic repertoires of pinnipeds, describing temporal patterns of call production, and documenting the behavioral contexts in which calls are produced. Our work demonstrates how these acoustically reliant species are able to overcome the conflicting sensory demands that are inherent to their amphibious lifestyles. Furthermore, much of our research informs predictions of how noise influences hearing and communication in free-ranging animals and supports efforts to mitigate potentially harmful effects of anthropogenic noise in the marine environment. In tandem with its research objectives, this program plays a crucial role in fostering the careers of young scientists who are involved in every aspect of our work.

**11:05**

**2aAB7. Curtin University's Centre for Marine Science and Technology: Virtual lab tour.** Christine Erbe (Ctr. for Marine Sci. and Technol., Curtin Univ., B301, Kent St., Perth, Western Australia WA 6102, Australia, C.Erbe@curtin.edu.au), Iain M. Parnum, and Shyam Madhusudhana (Ctr. for Mar. Sci. and Technol., Curtin Univ., Perth, Western Australia, Australia)

Founded in 1985, the Centre for Marine Science and Technology (Perth, Western Australia) comprises an interdisciplinary group of physical, biological, data, and computer scientists, as well as engineers, using active and passive acoustic techniques to study the ocean. We work in close partnership with offshore industry and government on marine soundscapes, sound generation and propagation, underwater noise and its impacts on marine fauna (terrestrial and marine) animal sound production and reception, acoustic ecology, and marine habitat mapping and modeling. In 2023, we opened our Centre of Ocean and Earth Science and Technology at Curtin Mauritius (Telfair, Mauritius), focussing on marine and terrestrial (bio-)acoustics. Let us take you on a tour of our two labs, meet some of our staff and students, and let them excite you about their projects: what do diving birds hear; how do we predict the noise of yet-to-be-built equipment; how can we manage and mitigate the effects of noise; what habitat characteristics are suitable for endangered insects, birds, and whales; can we acoustically enhance habitats; do whales feed on migration; can acoustic behavior predict physical and functional behavior; are long-term trends in Antarctic whale presence related to climate change; etc.?

11:15

**2aAB8. Leveraging big data and deep learning to uncover the connection between biosonar sensing and flapping flight in bats.** Rolf Müller (Mech. Eng., Virginia Tech, ICTAS II, 1075 Life Science Cir (Mail Code 0917), Blacksburg, VA 24061, rolf.mueller@vt.edu), Grace Chen (Mech. Eng. and Mater. Sci., Washington Univ. St. Louis, St. Louis, MO), Brandon Dang (Mech. and Aerospace Eng., Oklahoma State Univ., Stillwater, OK), Joshua Foley (Biol. Sci., East Tennessee State Univ., Johnson City, TN), Maha Khalid (Psychol. Sci., Rice Univ., Houston, TX), Charles Koduru (Robotics and Mechatronics Eng., Kennesaw State Univ., Marietta, GA), Robert Lorence (Elec. and Comput. Eng., Virginia Tech, Blacksburg, VA), Chi Nnoka (Mech. and Aerospace Eng., Univ. at Buffalo, The State Univ. of New York, Amherst, NY), Christopher Payne (Comput. Sci., Virginia State Univ., Petersburg, VA), Rhys Shallbetter (Mech. Eng., Virginia Tech, Blacksburg, VA), and Tiffany Veliz (Pathology and Genomic Med., Thomas Jefferson Univ., Philadelphia, PA)

Understanding the coupling of biosonar sensing and flapping flight in bats poses a challenge, because control of the many degrees of freedom in the flight apparatus of bats is not readily reconciled with the low-dimensional nature of the biosonar inputs. This is particularly true for bat groups that are capable of dexterous maneuvers in dense vegetation and rely on biosonar as their primary source of sensory information about their environments. To uncover how bats with sophisticated biosonar sensing and agile flight are able to control their aerial maneuvers based on the biosonar echoes, a laboratory setup with synchronized arrays of high-speed video cameras and ultrasonic microphones has been established to collected large volumes of data. The hardware setup is complemented by deep-learning methods to process the raw data from the arrays and discover informative patterns. The effectiveness of these methods opens the possibility to leverage the behavioral variability across individual and species for the discovery of common patterns and adaptations. To ensure that insight into biosonar and flapping flight of bats can be exploited for advances in bioinspired technology, biomimetic sonar and flapping-flight robots are being recorded under the same conditions as the bats to allow for side-by-side quantitative comparisons.

## *Contributed Paper*

11:25

**2aAB9. Distributed acoustic sensing for underwater passive acoustic monitoring of biophonic, geophonic, and anthropogenic sources—University of Washington Lab tour.** Quentin Goestchel (School of Oceanogr., Univ. of Washington, 1503 NE Boat St., Seattle, WA 98115, qgoestch@uw.edu), Erfan B. Horeh (School of Oceanogr., Univ. of Washington, Seattle, WA), Ethan F. Williams (Dept. of Earth and Space Sci., Univ. of Washington, Seattle, WA), John Ragland (School of Oceanogr., Uni. of Washington, Seattle, WA), Léa Bouffaut (K. Lisa Yang Ctr. for Conservation Bioacoustics, Cornell Univ., Ithaca, NY), Shima Abadi (School of Oceanogr., Univ. of Washington, Seattle, WA), David Dall'Osto (Appl. Phys. Lab. at the Univ. of Washington, Seattle, WA), Marine Denolle (Dept. of Earth and Space Sci., Univ. of Washington, Seattle, WA), Alexander S. Douglass (School of Oceanogr., Univ. of Washington, Seattle, WA), Brad P. Lipovsky, Katelyn Schoedl (Dept. of Earth and Space Sci., Univ. of Washington, Seattle, WA), and William S. Wilcock (School of Oceanogr., Univ. of Washington, Seattle, WA)

Anthropogenic noise in the ocean is one of the many stressors affecting marine biodiversity. Knowing animals and ships locations is key to conservation efforts. Geophony is also of research interest, as T-phases analysis can help to characterize ocean seismicity and measure ocean temperature. However, offshore passive acoustic monitoring remains challenging and expensive, necessitating continuous advancements in technology. Distributed Acoustic Sensing (DAS) is one possible complement that leverages existing infrastructure to provide continuous acoustic data from the seafloor in near-real-time, at an onshore facility. DAS uses fiber-optic cables as sensors, offering capabilities comparable to an array of thousands of directional hydrophones, whose sampling frequencies depend on the longest probing distance. Various datasets are presented here, including those from the fiber connection between Seattle and Bothell, the Ocean Observatories Initiative's Regional Cable Array off the coast of Oregon, and the MARS cable in Monterey Bay. In this virtual lab tour, we provide an insight into the workflow, from data collection to processing and data analysis. The results of this research at the University of Washington have led to new observational efforts on land, ice, and underwater, spanning fields as diverse as oceanography, seismology, engineering, and marine ecology. [Work partially supported by ONR]

## *Invited Paper*

11:35

**2aAB10. Virtual tour of Kloepper ecological acoustics and behavior lab.** Laura Kloepper (Dept. of Biol. Sci., Univ. of New Hampshire, 230 Spaulding Hall, Durham, NH 03824, laura.kloepper@unh.edu)

Our lab conducts bioacoustic research with bats, frogs, penguins, terns, moose, and more! This tour, taken from a student's perspective, will show our lab's working spaces and highlight active research projects with our group.

**11:45–12:00 Discussion**

WEDNESDAY MORNING, 20 NOVEMBER 2024 10:00 A.M. TO 12:00 NOON

## Session 2aAO

## Acoustical Oceanography: Careers in Underwater Sound

S. B. Martin, Chair
*Halifax, JASCO Appl. Sci., 20 Mount Hope Ave., Dartmouth, B2Y 4S3, Canada*

**Chair's Introduction—10:00**

### *Invited Paper*

**10:05**

**2aAO1. Careers in underwater sound.** David R. Barclay (Dept. of Oceanogr., Dalhousie Univ., P.O. Box 15000, Halifax, Nova Scotia B3H 4R2, Canada, dbarclay@dal.ca)

During this session, a moderator led discussion between panel members will be held. The panel members will represent various career outcomes of advanced training in acoustics, oceanography, and ocean science, including careers in academia, industry, government, non-profits, and others. The panel will then be open to questions from the virtual audience.

WEDNESDAY MORNING, 20 NOVEMBER 2024 10:00 A.M. TO 12:00 NOON

## Session 2aBA

## Biomedical Acoustics: Debate: Nanobubbles—Can They Do Anything?

Eleanor P. Stride, Chair
*Univ. of Oxford, Inst. of Biomed. Eng., Oxford OX3 7DQ, United Kingdom*

**Chair's Introduction—10:00**

### *Invited Paper*

**10:05**

**2aBA1. Debate: Nanobubbles–Can they do anything?** Eleanor P. Stride (Univ. of Oxford, Inst. of Biomed. Eng., Oxford OX3 7DQ, United Kingdom, eleanor.stride@eng.ox.ac.uk)

The seemingly paradoxical echogenicity of sub-micrometre or "nano" bubbles has become a topic of considerable debate over the past 5 years. The aim of this session is to discuss the existing evidence for the existence of nanobubbles, the mechanisms by which they are stabilized, the techniques and methodologies used to characterise them, and their potential in different diagnostic and therapeutic applications. The session will take the form of an open debate with short presentations being sought from researchers working on nanobubbles to address each question and then a discussion in which the whole audience will be invited to participate.

# Session 2aCA

## Computational Acoustics: Interactive Computational Acoustics Demonstrations

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

Michelle E. Swearingen, Cochair
*Construction Eng. Res. Lab., 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*

**Chair's Introduction—10:00**

### *Invited Papers*

**10:05**

**2aCA1. Demonstration of a web-based tool for interactively visualizing Taylor series approximations.** Noah J. Parker (Graduate Program in Acoust. Pennsylvania State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, np.acoustics@gmail.com) and Daniel A. Russell (Graduate Program in Acoust., Pennsylvania State Univ., University Park, PA)

Interactive visualizations have proven effective in teaching math and physics concepts fundamental to acoustics, because the interactivity further enhances student engagement by allowing them to actively explore and interact with the material. This presentation will introduce ongoing work at Penn State to create a comprehensive website of interactive animations and tools aimed at supporting the learning of fundamental acoustics topics. The talk will include a demonstration of a specific page designed to visually explain Taylor series approximations and show the impact of including higher order terms. The goal of this page being to help students towards a clear and intuitive understanding of the concept as it is used in acoustics.

**10:10**

**2aCA2. Interactive demo of Openwind, an online tool for simulating the acoustic response of wind instruments.** Augustin Ernoult (Inria, Sorbonne Univ., CNRS, Inst. Jean Le Rond d'Alembert, 4 Place Jussieu, Paris 75005, France, augustin.ernoult@inria.fr) and Juliette Chabassier (Modartt, Toulouse, France)

Openwind is a Python library (free and open source) dedicated to the simulation of wind instruments. It allows the calculation of the acoustic response in the frequency domain (impedance or admittance) or in the time domain (impulse response), from the geometry of the instrument (main bore, side holes, and valves). The wave propagation model can take into account thermo-viscous effects, non-uniform temperature, multiple radiation conditions through boundary conditions, etc. It is also possible to simulate the sound of brass, reed, and flute instruments. Discretization is done in space with 1D spectral finite elements and in time with energy consistent finite differences. The sensitivity of the instrument to design parameters can be calculated and visualized. A graphical interface, freely available online (*https://demo-openwind.inria.fr*), gives the possibility to perform frequency domain simulations without coding. It is mainly used by students (in science or musicology) and instrument makers (hobby, student, or professional). With this interface, it is easy to observe the effect of a geometric modification on the acoustic response (resonance frequency, mode shape, etc.) or a change in temperature or radiation conditions. The numerical aspects (mesh, order, etc.) are set automatically and can be further refined using the Python library.

### *Contributed Papers*

**10:15**

**2aCA3. Open-source real-time auralizations with Virtual Acoustics.** Pascal Palenda (Inst. for Hearing Technology and Acoustics, RWTH Aachen Univ., Kopernikusstr. 5, Aachen 52074, Germany, pascal.palenda@akustik.rwth-aachen.de), Philipp Schäfer, Lukas Aspoeck, and Michael Vorlaender (Inst. for Hearing Technol. and Acoust., RWTH Aachen Univ., Aachen, Germany)

Auralizations are a valuable tool for demonstrating acoustic phenomena. They can form a basis on which both non-experts and experts can discuss and evaluate acoustic scenarios. However, creating auralizations often demands expert knowledge due to their complexity. The open-source, real-time auralization framework "Virtual Acoustics" (VA) aims to simplify this process. It supports various physically based sound propagation models, from simple free-field scenarios to complex room acoustic and outdoor scenarios. Dynamic changes in the scene, such as moving sources and receivers, are also supported. In addition to the propagation models, different spatialization methods are available, including binaural rendering and Ambisonics. Its user-friendly, modular design abstracts the complexity of underlying models while offering a simple, unified interface. This

contribution presents some of the possible applications of the framework based on the examples included. A particular focus will be set on the auralization of a car pass-by that includes diffraction effects.

**10:20**

**2aCA4. Framework for real-time computation and streaming of soundscapes.** Rouben Rehman (Inst. for Hearing Technol. and Acoust., RWTH Aachen Univ., Kopernikusstraße 5, Aachen, NRW 52074, Germany, rre@akustik.rwth-aachen.de), Christian Dreier, Jonas Heck, Josep Llorca-Bofí, and Michael Vorlaender (Inst. for Hearing Technol. and Acoust., RWTH Aachen Univ., Aachen, Germany)

The auralization of dynamic soundscapes brings significant computational challenges, as multiple often moving sources have to be considered simultaneously. Making such auralizations interactive rules out pre-rendering, requiring the underlying computations to be performed in real-time instead. The “Virtual Acoustics” (VA) framework allows for such real-time applications by leveraging geometrical acoustics and efficient multi-threading. Delivering these simulations to a larger audience, however, has proven difficult, as they still require a lengthy setup and possible simulation complexity is highly dependent on available hardware. When also including high-fidelity 3D visualization, hardware requirements rise further, hindering remote deployment. We present a web-based approach that executes simulations on a server and streams the result in real-time, offloading heavy calculations to a sufficiently powerful machine. 3D graphics are rendered by Unreal Engine and streamed using Pixel Streaming. VA auralization is transmitted via WebRTC audio streams. Both visualization and auralization are ultimately aggregated in a single website for platform-independent access. User input is gathered by the website and relaid back to the server, making the experience interactive. Thereby, this approach can deliver high-fidelity virtual environments with minimal setup and hardware requirements for the user. The approach will be demonstrated by streaming an interactive city park soundscape.

**10:25**

**2aCA5. Three-dimensional visualization of acoustic tension fields and energy flow in sound fields.** Carlos E. Graffigna (Dept. of Basic and Technol. Sci., National Univ. of Chilecito, 9 de Julio 22, Chilecito, La Rioja F5360CKB, Argentina, cgraffigna@undec.edu.ar) and Domenico Stanzial (Energy Acoust. Lab., DFST-Univ. of Ferrara, CNR-IMM, Bologna, Bologna, Italy)

Building upon the foundational work of the article “On the Sound Tension and Action-Reaction Law in Acoustics” (Stanzial, 2023), which explores the theory and physical interpretation of Lagrangian force density in sound fields, this article shifts focus to the three-dimensional visualization of acoustic tension fields and both instantaneous and stationary visualizations of force densities fields. Tools, such as Maple, Blender, and Python, are utilized to generate three-dimensional videos with stereoscopic images suitable for platforms like YouTube, as well as for use with virtual reality devices. A step-by-step guide to the process is provided in the paper, highlighting potential improvements and directions for future work. Through these visualizations, a deeper understanding of sound field dynamics is enabled, offering new perspectives for research and education in acoustics.

**10:30**

**2aCA6. Implementation and application of complete radiation boundary conditions: A demonstration.** Thomas Hagstrom (Mathematics, Southern Methodist Univ., PO Box 750156, Dallas, TX 75275-0156, thagstrom@smu.edu)

The radiation of energy to the far field is a fundamental feature of many problems in acoustics. Efficient numerical algorithms for simulating sound propagation in the time domain must therefore include convergent near-field domain truncation procedures. For problems modeled by uniform or stratified far fields, complete radiation boundary conditions provide a provably optimal domain truncation algorithm. In particular, they are spectrally convergent in boundary condition order and can be placed quite close to scatterers or other inhomogeneities. However, the mathematical description of the radiation conditions looks complicated in comparison with less-efficient methods, such as perfectly matched layers or simple damping. In this demonstration, we show how complete radiation conditions can be implemented in a standard discontinuous Galerkin code as easily as other schemes. In fact, the implementation is shown to simply involve a single extra element with a nonstandard differentiation matrix.

**10:35**

**2aCA7. Investigating architectural acoustics modeling with the summer undergraduate research or internship experience in acoustics program.** Dante Christian (Pomona College, 150 E. 8th St., Claremont, CA 91711, dnchristian4@gmail.com), Robert Connick (Acentech, Cambridge, MA), and Alex Roehl (Acentech, Cambridge, MA)

Dante Christian was chosen to participate in an inspiring 10-week experience with the Acoustical Society of America’s Summer Undergraduate Research Experience in Acoustics (SURIEA) Program. Given his physics major and architectural background, SURIEA coordinators felt that he would thrive best as an intern at Acentech (an architectural acoustics firm). The research project that Dante landed on was an auralization of the Warner Theater’s rehearsal room—a structure in Erie, PA. Two of his mentors from Acentech visited to make reverberation time measurements and to ensure that the rehearsal room was behaving as it was intended to. Thus, these measurements came to serve as the backbone of his project’s structure. For Dante’s summer research, he went through the processes of modeling in SketchUp and in Trebel (a geometric and wave-based solver) to gather and simulate impulse responses for the Warner Theater’s Rehearsal room. He worked iteratively to assign absorption and scattering coefficients to the model that best fit the measured experimental results. This is because, especially, absorption coefficients of every material in the room were not specified. Overall, this research project taught Dante how to critically think about a space, its materials, and how its materials interact with the soundscape.

**10:40–12:00 Discussion and Demonstrations**

# Session 2aEA

## Engineering Acoustics: General Topics in Engineering Acoustics

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

### *Contributed Papers*

10:00

**2aEA1. A method for comparing candidate materials in subjective tests of flat-panel loudspeakers.** David A. Anderson (Elec. Eng., Univ. of Minnesota Duluth, 5230 Otsego St., Duluth, MN 55804, and10445@d.umn.edu) and Lukas Aanonsen (Elec. Eng., Univ. of Minnesota Duluth, Duluth, MN)

Flat-panel loudspeakers are often built using a moving-coil or piezoelectric exciter attached to a bending panel that radiates sound, and such devices are finding increased use in TV screens, computer screens, and in sensitive electronics where protection from dust or moisture prevents the use of traditional speaker devices. The overall acoustic response of these speakers is dominated by resonant frequencies which can be calculated from plate parameters such as material, dimensions, and thickness, but each material still has its own unique i'sound" that arises from internal damping rates, non-homogeneity, and other effects that must be measured empirically. As such, many hobbyists and designers are interested in finding the "best-sounding" material for use in flat-panel loudspeakers. In this presentation, a method is outlined for designing plates of various materials with off-the-shelf thicknesses that have near-identical resonances and excursion such that the unique "sound" of each material can be compared in subjective tests. Simulations of and measurements on plates made from acrylic, aluminum, and foamboard demonstrate the effectiveness of the proposed method.

10:10

**2aEA2. Toroidal shell electroacoustic transducer.** Divyamaan Sahoo (Elec. & Comput. Eng., Univ. of Massachusetts Dartmouth, North Dartmouth, MA) and David A. Brown (ECE/CIE, Univ. of Massachusetts Dartmouth, 151 Martine St., UMass Dartmouth, Fall River, MA 02723, dbAcousticsdb@gmail.com)

The design and modeling of an electroacoustic toroidal-hollow-shell transducer for underwater acoustics are considered. Radiation from a solid toroid was considered by Sherman and Parke (JASA 38, 1965) and is useful as an approximation for the radiation of a short cylinder. We explore the vibration of a hollow toroid vibrating in circumferential hoop-mode and in breathing tube-mode and present an equivalent electromechanical circuit model for acoustic radiation. Results are compared with a first prototype made from piezoelectric ceramics comprised of piecewise cylindrical tube sections. [Work supported in part by ONR and the MUST research program at the Umass Dartmouth and a subject of the author (DS) Master's Thesis.]

10:20

**2aEA3. Scaling and aspect ratio effects on the frequency-modulated ultrasonic signal generated via fluidic oscillator.** Viswa R. Sunkavalli (Zerstörungsfreie Prüfmethoden für das Bauwesen, Bundesanstalt für Materialforschung und -prüfung, Unter den Eichen 87, Berlin, Berlin 12205, Germany, viswa.sunkavalli@bam.de), Christoph Strangfeld, and Stefan Maack (Zerstörungsfreie Prüfmethoden für das Bauwesen, Bundesanstalt für Materialforschung und -prüfung, Berlin, Germany)

Traditional ultrasonic non-destructive testing methods in civil engineering require the use of coupling agents, leading to prolonged and labor-intensive measurement procedures along with the risk of surface damage. To alleviate these concerns, air-coupled ultrasonic actuation has been introduced as an effective alternative. However, the power loss due to impedance mismatchs at the interfaces remains an important limitation. To mitigate the power losses at the interfaces while characterizing the specimen thickness, we employ the fluidic oscillator as an ultrasonic source, wherein air acts as both the operating and coupling medium. The fluidic oscillator generates signals through self-sustained oscillations of the exiting air jet under continuous pressurized air supply, and therefore, eliminates the need for intricate design and manufacturing processes. The frequency spectrum characteristics and time-averaged acoustic fields of the fluidic oscillators are shown for the several configurations that were designed by varying the aspect ratio and scaling. Our preliminary investigations highlighted the dependence of the dominant spectral characteristics of a given fluidic oscillator on the mass flow rate of input air. Leveraging this observation, frequency-modulated chirp signals are produced by rapidly varying the flow rate, enhancing the signal-to-noise ratio for a reliable assessment of material characteristics.

10:30

**2aEA4. Self-excited thermoacoustic instability in a swirl tubular meso-scale combustor with oxygen-methane combustion.** Yiheng Guan (Dept. of Mech. Eng., Univ. of Canterbury, Christchurch, New Zealand), Dan Zhao (Dept. of Mech. Eng., Univ. of Canterbury, Private Bag 4800, Christchurch 8140, New Zealand, dan.zhao@canterbury.ac.nz), Ning Zhang (Dept. of Mech. Eng., Univ. of Canterbury, Christchurch, New Zealand), and Baolu Shi (School of Aerospace Eng., Beijing Inst. of Technol., Beijing, China)

Self-excited combustion instabilities can lead to structural vibration, overheating, and even engine failure in practical large-scale engines. It is highly undesirable. In this work, we experimentally and numerically investigated on self-excited combustion instability in a swirl tubular meso-scale combustor with oxy-fuel combustion involved. The oxygen mole fraction is experimentally varied from 25% to 100%. And the oxygen and methane are injected from 4 individual slits into a 100 mm long combustor. Its inner diameter is 16 mm. The slits width could be varied to promote mixing. Such 100% oxygen-methane combustion is able to completely remove NOX emissions from combustion systems. Under certain conditions, self-excited combustion-driven limit cycle oscillations at approximately 2000 Hz are experimentally observed. To simulate experimental tests, we conduct 3D RANS (Reynolds Average Navier Stokes) simulations in time domain, as the equivalence ratio is varied from 0.8 to 1.0 and swirling number is varied from 5.9 to 24.7. Limit cycle oscillations are numerically generated. Comparing with the experimentally measured pressure spectrum, a good agreement is obtained. In general, the present work shed lights on the generation mechanism of self-excited combustion instability in meso-scale combustor under oxy-fuel combustion conditions. It also provides a numerical tool to design a meso-scale combustor.

10:40

**2aEA5. Numerical investigations on a portable outdoor waste heat-driven standing-wave thermoacoustic engine.** Lixian Guo (Dept. of Mech. Eng., Univ. of Canterbury, Faculty of Eng., Christchurch 8140, New Zealand, lixian.guo@pg.canterbury.ac.nz) and Dan Zhao (Dept. of Mech. Eng., Univ. of Canterbury, Christchurch, New Zealand)

The present study focuses on developing a portable outdoor waste heat-driven thermoacoustic engine (TAE) with the capability to harness waste heat from commonly used camping gas stoves, enabling energy storage and acoustic power generation. To conserve space while enhancing heat-driven acoustic power output, the developed thermoacoustic system is composed of two quarter-wavelength standing-wave TAEs sharing a common closed end, forming a U-shaped configuration with two stacks symmetrically distributed on both sides. The numerical study simulates the combustion of an outdoor gas stove (primarily fueled by liquefied butane) to derive its temperature function. This function is then applied to the two-dimensional full-scale U-shaped TAE to yield self-excited thermoacoustic oscillations. To improve energy efficiency and acoustic power output, the study investigates the impact of the (1) temperature function generated by combustion, (2) working gas/medium, and (3) operating pressure. Additionally, nonlinear phenomena within the system are explored. The generated acoustic power can be further converted into electricity using piezoelectric elements or electric generators, serving as an emergency power source for mobile phone charging, car engine ignition, and emergency lighting. This research unveils significant potential for utilizing low-grade thermal energy to pioneer innovative outdoor thermoacoustic power sources.

10:50

**2aEA6. Exploring multi-qubit analogue operations through acoustic wave dynamics.** M. Arif Hasan (Mech. Eng., Wayne State Univ., 5050 Anthony Wayne Dr., Detroit, MI 48202, Hasan.Arif@Wayne.Edu), Pierre A. Deymier, Keith Runge (Mater. Sci. and Eng., Univ. of Arizona, Tucson, AZ), and Joshua Levine (Comput. Sci., The Univ. of Arizona, Tucson, AZ)

Quantum computing harnesses quantum phenomena like superposition and entanglement to surpass classical computers, showing promise across various fields. Current techniques utilize qubits in quantum circuits for parallel information processing, though managing and measuring these systems poses significant challenges. We propose a novel approach to simplify executing complex quantum algorithms without relying on multiple quantum gate operations. Our method introduces logical phi-bits—classical counterparts to qubits, using nonlinear acoustic waves in an externally driven acoustic metastructure. We demonstrate that complex multi-phi-bit unitary operations, akin to those in quantum circuits, can be conducted through a single action on this metastructure. This method starkly contrasts traditional quantum computing, which requires decomposed sequences of qubit gates for equivalent operations. The phi-bit system simplifies processes that are typically complex in quantum mechanics, potentially enhancing robustness and ease of implementation. Our results indicate that phi-bits could expand computational models by merging classical wave dynamics with quantum computational principles, thereby widening the potential of computational technologies. This research advances quantum-analogue computation and introduces new prospects for utilizing wave physics in information processing, thereby challenging and expanding existing paradigms in both classical and quantum computing. [Funding: NSF grant 2204382, 2204400, and 2242925.]

11:00

**2aEA7. Two levels of Bayesian Dissipation Analysis in impedance tube measurements.** ZIQI CHEN (School of Architecture, Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, chenz33@rpi.edu) and Ning Xiang (School of Architecture, Rensselaer Polytechnic Inst., Troy, NY)

Accurate impedance tube measurements are often compromised by additional absorption due to air dissipation, particularly at high frequencies. Various effects contribute to air dissipation in a tube, and numerous models have been developed to describe the dissipation. This work explores different dissipation models and estimates the dissipation accurately using Bayesian inference. The exploration involves two levels of Bayesian inference: the higher level of inference addresses the model selection and the lower level of inference estimates critical parameters in models, such as sound speed and dissipation factor. The experimental data are derived from impedance tube measurements of the empty tube, and the comparison is extended to high frequency. Based on the analytical results and experimental validation, Bayesian inference is able to estimate and select dissipation models for the impedance tube measurements at high frequencies.

11:10

**2aEA8. Numerical investigation on acoustic damping characteristics of dual Helmholtz resonators in presence of a grazing flow.** He Zhao (Dept. of Mech. Eng., Univ. of Canterbury, 388 Wigram Rd., Halswell Christchurch, Canterbury 8025, New Zealand, hzh164@uclive.ac.nz), Dan Zhao, and Jinshen Tong (Dept. of Mech. Eng., Univ. of Canterbury, Christchurch, New Zealand)

In this study, the acoustic damping performances of the dual Helmholtz resonators were numerically evaluated using a 3D model. The grazing flow passes tangentially through the resonator neck, with a Mach number range of $0 \leq \mathrm{Ma} \leq 0.1$. The numerical model operates by solving the linearized Navier–Stokes equations. The current model is validated through a comparison with experimental data. The model is then utilized to explore the effects of the dual Helmholtz resonators on acoustic transmission loss performance in the presence of a grazing flow. Three key parameters are examined: (1) different implementation configurations of the dual Helmholtz resonators [including Models (b), (c), and (d)], (2) the mean temperature of the grazing flow, and (3) the axial distance between the dual Helmholtz resonators. For comparison, the acoustic damping performance of these dual Helmholtz resonators is compared to the single Helmholtz resonator case [Model (a)]. It is observed that the dual Helmholtz resonators dramatically increase the transmission loss. Therefore, increasing the mean temperature is shown to be beneficial to enhance transmission losses in the presence of the grazing flow.

11:20

**2aEA9. A study of the application of global optimization for the arrangement of absorbing materials in multi-layered absorptive fluid silencer.** HONGMIN PARK (Naval Architecture and Ocean Eng., Seoul National Univ., 900 Bldg. 205 Room, 1, Gwanak-ro, Gwanak-gu, Seoul 08826, Korea (the Republic of), hongmini0202@snu.ac.kr) and Woojae Seong (Naval Architecture and Ocean Eng., Seoul National Univ., Seoul, Korea (the Republic of))

In naval ships and submarines, silencers are installed to reduce pipe flow noise. Due to limited space, the size of the silencers is restricted, requiring them to have excellent acoustic performance. In this study, to improve the acoustic performance of a multi-layer absorptive fluid silencer, the physical properties of the absorbing material, mesh, and design parameters were reviewed in detail compared to previous studies. A global optimization algorithm was introduced to find the optimal layering of the absorbing materials. The acoustic performance of the silencer was evaluated through transmission loss. To calculate the transmission loss, a finite element method-based numerical analysis program (COMSOL) was used along with genetic algorithm and simulated annealing for global optimization. The absorbing material used was polyurethane, a porous elastic material, and five practically applicable absorbing materials were selected. The placement and thickness of these materials in each layer were optimized, and the reliability of the global optimization techniques was verified through comparison with solutions from a brute force algorithm. Additionally, an application method was proposed to find the optimal solution in a short time, even for silencers with more than ten layers.

11:30

**2aEA10. Analysis of quadrupolar frequencies of the spherical gravitational wave detector by finite element modeling: A study case.** Natan Vanelli (Instituto Federal de Educação, Ciência e Tecnologia de São Paulo, Rua Iatapiru, Sao Paulo, SP 04143-010, Brazil, natan.vanelli92@gmail.com), Nadja Simao Magalhaes (Universidade Federal de São Paulo, Sao Paulo, Brazil), Sergio Turano Souza (Fatec Itaquera, Sao Paulo, Brazil), Fabio da Silva Vanelli (Instituto Federal de Educação, Ciência e Tecnologia de São Paulo, Sao Paulo, Brazil), and Carlos Frajuca (Mechanics, IFSP, Carapicuiba, SP, Brazil)

The Schenberg detector is a spherical resonant antenna gravitational wave detector with a solid sphere with a diameter of 65 cm and a weight of approximately 1.15 metric tons. It is made of a copper–aluminum alloy with 94% Cu and 6% Al. This work aims to verify the natural frequencies of the Schenberg detector structure. The authors used the finite element method to calculate the quadrupole frequency band and compare it with the value measured in tests during the construction of the antenna at University of Sao Paulo, which was 67.3 Hz. In the modal simulation with the homogenous material, the frequency band of the quadrupole modes was around 35 Hz. Many trials to improve this result were made with no significant improvement. The improvement came when the sphere was considered not homogenous, as to the cooling process of the sphere casting, the alloy that cooled faster (close to the surface sides and in the bottom) can reach a hardness 20% higher than the material that cooled slower (in the center and on the top). Including these characteristics, the band value obtained was close to the one measured during the construction of the antenna, a value of 63.2 Hz was obtained.

11:40

**2aEA11. Automated ship noise measurement system and Enhancing Cetacean Habitat and Observation program ship noise database.** Zizheng Li (JASCO Appl. Sci., 2305-4464 Markham St., Victoria, BC V8Z 7X8, Canada, zizheng.li@jasco.com) and David E. Hannay (JASCO Appl. Sci., Victoria, BC, Canada)

An automated ship Underwater Radiated Noise (URN) system has been developed and applied to process measurements of tens of thousands of commercial vessel transits, to produce the Enhancing Cetacean Habitat and Observation (ECHO) Program's Ship URN database—believed now to be the largest existing ship URN database. The system integrates environmental data (ocean current, wind speed), vessel AIS broadcasts, vessel classification metadata obtained from commercial ship databases, and acoustic data from JASCO's Underwater Listening Stations (ULS). It utilizes advanced signal processing methods (e.g., image symmetry detection, cepstrum, harmonic analysis, DEMON) to automatize analysis tasks such as estimating vessel closest point of approach and propeller rotation rates. Full wave models are applied to calculate propagation loss. Vessel noise emissions (Radiated Noise Level and Source Level) are determined in compliance with ANSI standard 12.64-2009 Grade C protocols, correcting for background noise in all frequency bands. Additionally, ShipSound offers a comparative ranking system for vessel noise emissions relative to peers of the same category. All data and metadata are stored in a database and are accessible through a full-featured web interface. In this paper, we will describe the processing system details and provide examples of measurement reports and metadata displays.

11:50

**2aEA12. Conveyor noise assessment and control.** Nick Antonio (Antonio Acoust., 2201 Highland Vista Dr., Arcadia, CA 91006, nick.antonio@antonioacoustics.com) and Destinee Luong (Antonio Acoust., Arcadia, CA)

This paper provides details of conveyor belt noise measurements and assessment of noise, with predicted sound power levels. Conveyor sources include motors, belts, rollers, hoppers and alarms, and power generation. Conveyors have both point and line sources. The paper also describes control measures to reduce noise impact.

## Session 2aED

## Education in Acoustics: Tones, Tines, and Tings–Virtual Demonstration Show by David Cotton

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

**Chair's Introduction—10:00**

### *Invited Paper*

**10:05**

**2aED1. Tones tines and tings.** David Cotton (Physics, Cardinal Newman College, Lark Hill Rd., Preston, Lancashire PR1 4HD, United Kingdom, dcotton@cardinalnewman.ac.uk)

This session will be full of ideas and demonstrations that focus on aspects of sound waves in the curriculum and beyond, e.g., how to turn a tuning fork and a magnet into a model guitar pick up! These ideas tell a story based on the development and usage of oscillation and vibration in music and communication. Includes many more ideas using lab equipment and some musical instruments. This presentation has been developed in memory of Anthony Waterhouse, supported by the Fellowship scheme offered annually through the Institute of Physics UK.

**11:05–11:45 Discussion**

WEDNESDAY MORNING, 20 NOVEMBER 2024 10:00 A.M. TO 11:45 A.M.

## Session 2aMU

## Musical Acoustics: General Topics in Musical Acoustics II

Montserrat Pàmies-Vilà, Cochair
*Dept. of Music Acoust. - Wiener Klangstil (IWK), Univ. of Music and Performing Arts Vienna, Anton-von-Webern-Platz 1, mdw - Inst. 22, Vienna 1030, Austria*

Gary Scavone, Cochair
*Music Research, McGill Univ., 555 Sherbrooke St. West, Montreal H3A 1E3, Canada*

### *Contributed Papers*

**10:00**

**2aMU1. A comparative study of flutes I.** Gordon P. Ramsey (Physics Dept., Loyola Univ. Chicago, Chicago, IL 60660, gramsey@luc.edu) and Amanda Newton (Physics Dept., Loyola Univ. Chicago, Chicago, IL)

This is part one of two presentations on the same project. Various types of flutes have been used for centuries and they are popular in many genres of music. We have done a comparative study of six different flute type instruments to determine what factors play a role in their differences in timbre. The effects of flute geometry, type of mouthpiece, body size, and material were considered on how each affected their sounds. The study included an orchestral flute, a piccolo, two Native American flutes, and two recorders. Measurements were taken at the Loyola acoustics lab and the anechoic chamber at Northwestern. This talk will present theoretical basis for our measurements, including impedance calculations, and will discuss the observations for the individual instruments. The following talk will discuss the comparative study and how our results contribute to selection of the appropriate flute to use in different genres.

10:15

**2aMU2. A comparative study of flutes II.** Amanda Newton (Loyola Univ. Chicago, 6317 N Broadway, Chicago, IL 60660, newtonamandam@gmail.com) and Gordon P. Ramsey (Physics, Loyola Univ. Chicago, Chicago, IL)

This is part two of the previous presentation on flute acoustics. This talk will expand on the individual observations of each of the six flutes to include a cross-instrument comparison. Factors used in the comparison include instrument range, material, geometry, and instrument types. Findings conclude that material had the predicted effect on tone and geometry was more significant than predicted. Additional findings and confirmation of these from the NW anechoic chamber will be discussed, as well. These comparative findings are significant as they provide insight into the physics of the sound produced in each instrument and how that contributes to the unique tone of each instrument. This is significant for musicians, including flute players and composers, as predicting and manipulating the tone of different flutes is instrumental in making music, from knowing how to make, play, and compose for flutes, including when choosing what flute instruments are applicable in certain musical settings. This talk will also discuss possible extensions of this project, including to extended members of the flute family, and more note ranges tested on each instrument.

10:30

**2aMU3. Flow measurements in a bass recorder using particle image velocimetry.** Titas Lasickas (Dept. of Music Acoust., Univ. of Music and Performing Arts Vienna, Anton-von-Webern-Platz 1, Vienna 1030, Austria, lasickas@mdw.ac.at), Giuseppe Caridi, Alfredo Soldati (Inst. of Fluid Mech. and Heat Transfer, Vienna Univ. of Technology, Vienna, Austria), and Vasileios Chatziioannou (Dept. of Music Acoust., Univ. of Music and Performing Arts Vienna, Vienna, Austria)

The timbre of wind instruments is determined by the generation of standing waves inside the instrument. These can be studied by analyzing the acoustic pressure and flow oscillations of the air column. Particle Image Velocimetry (PIV) is a powerful tool capable of non-intrusively measuring air flow inside a tube. However, only a few studies have considered the application of time-resolved PIV to measure air flow inside a wind instrument. In this study, flow is measured inside a custom-made transparent recorder tube under natural playing conditions, during steady state. By collecting detailed flow measurements, this work aims to investigate the complex interactions between the air flow and the tone hole geometry of the instrument. The tone hole geometry is varied in diameter, height, and shape during the experiments. These findings have potential implications for wind instrument makers and could lead to a deeper understanding of sound production and articulation mechanisms in wind instruments. The study presents observations and discusses the broader relevance of PIV in wind instrument acoustics research.

10:45

**2aMU4. Exploring and interpreting the spectral structure of the North Indian flute (Bansuri).** Trevor D. Smith (Eng., Univ. of Iowa, 103 South Capital St., Iowa City, IA 52241, tsmith125@uiowa.edu), Jane Kim, Daniela Perez (Eng., Univ. of Iowa, Iowa City, IA), and Ananya Sen Gupta (Dept. of Elec. and Comput. Eng., Univ. of Iowa, Iowa City, IA)

Spectrograms are powerful tools in many fields today, whether that be by a musician trying to improve their own ability with an instrument or an engineer developing a way to sense vehicles for an autonomous car. The spectrogram allows for deeper analysis than just what can be heard, giving information on its harmonic series, any frequencies below 20 Hz or above 20 kHz (range of hearing) and beyond. In this talk, we will explore the efficacy of spectral analysis to understand the acoustical structure the Indian flute (Bansuri) and provide interpretation of the melodic sounds generated by this ancient Indian instrument using spectral analysis. Specifically, we examine what harmonic features characterize the fluid melodic movements of the Bansuri and whether non-harmonic structures in the acoustic spectrogram carry pertinent musical information. We also qualitatively examine the accuracy of extracting the melodic features against non-melodic features that sound like dissonant hiss. We will discuss the potential of understanding the air column physics of this flute and relating it spectrally to similar physics of propagation in small cylindrical underwater sonar targets such as unexploded ordinances (UXO). Results based on original flute data (and potentially UXO and similar small-target sonar data) will be presented. [This work has been partially supported by the DoD Navy grant N00174-20-1-0016 and University of Iowa internal funds.]

11:00

**2aMU5. Sympathetic resonance of the pipe organ and its effects on concert Hall acoustics.** Ashley Snow (Music, Univ. of Washington, 1410 NE Campus Parkway, Seattle, WA 98195, ashsnow@uw.edu) and James P. Cottingham (Physics, Coe College, Cedar Rapids, IA)

The pipe organ stands present at most every famous concert hall and church sanctuary, yet often it serves only as decoration, rarely played. What effects, then, does the world's largest class of musical instrument have on the acoustics of concert halls that house them? With prominent metal resonators typically positioned as the immediate back wall of the stage, it is difficult to believe that the pipe organ has little sonic effect, even when it isn't being played. One hypothesis is that the pipe organ makes for some manner of auto-tune, as its pipes sympathetically resonate to the same frequencies they're tuned to, thus increasing the amplitude of frequencies that align with well-tuned 12-tone notes, and enhancing the overall musical sound of ensembles that play in concert halls with organs. It was verified experimentally that sympathetic resonance does occur in organ pipes due to musical performance, speeches, and noise, at frequencies that align with musical notes, and that overall amplitude increases when the signal matches the resonance of one or several pipes. Ongoing research is being conducted on the significance of these effects on the overall quality of musical performance to listeners in the audience.

11:15

**2aMU6. Audio performance comparison and analysis of bio-composite bassoon reed design.** Lizette M. Wong (Elect. Eng. & Music, Univ. of Texas at Austin, 204 E Dean Keeton St., Austin, TX 78712, lizettewong@utexas.edu) and Zoelle F. Wong (Aerospace Eng., Georgia Inst. of Technol., Atlanta, GA)

Bassoon reeds, commonly made with Arundo donax, often contain inconsistent audio and structural performance due to their natural degradation from exposure to saliva and humidity. They are a necessary tool to produce sound and are reported to be an expensive recurring cost. The combined cost of musical lessons and instrument upkeep creates a financial barrier for musicians. Made from polypropylene, synthetic reeds are commercially available; however, they have shown inconsistencies in the physical profile and sound. In this experiment, the audio performance of a bio-composite reed design, Fermata, is proposed and analyzed in an effort to introduce alternative reed options. Spectrograms and FFT plots were constructed after recording pitches at F2, F3, and F4 for four beats, tapering at the end and repeated five times per pitch case. Comparisons and audio analyses are made between Arundo donax, synthetic, and the Fermata reed. The current work indicates that polymer selection directly impacts overtone propagation and timbre brightness for a select range of pitch.

11:30

**2aMU7. A detailed study of the impedances of mouthpieces, flared bells, and curved pipes in brass instruments.** Miranda Jackson (Music Res., CIRMMT, Schulich School of Music, McGill Univ., Montreal, Quebec H3A 1E3, Canada, miranda.jackson@mail.mcgill.ca) and Gary Scavone (Music Res., CIRMMT, Schulich School of Music, McGill Univ., Montreal, Quebec, Canada)

The input impedance of a brass instrument provides information about quality and playing characteristics, such as intonation and response. Modeling the input impedance of an instrument with the transfer matrix and finite element methods based on its geometry is useful, as the calculations can be performed without a physical prototype. Previous work has determined that both the transfer matrix and finite element methods can determine the properties of the impedances of mouthpieces and flaring bells. These results will be more closely examined and compared with regard to relative peak locations. At present, the transfer matrix method can only be used for conical or cylindrical segments along a straight axis, so it is useful to add curvature as

a parameter of the calculation. An adaptation to the transfer matrix calculation for curved pipes will be presented. Transfer matrix and finite element calculations will be compared with each other and with impedance measurements. The ultimate goal of this work is to accurately calculate the input impedance of a realistic brass instrument, with curved pipes, valves, water keys, and slides. This will allow for improvement of existing instruments and development of new instruments and instrument parts exhibiting particular acoustical characteristics.

WEDNESDAY MORNING, 20 NOVEMBER 2024 — 10:00 A.M. TO 12:00 NOON

## Session 2aNS

## Noise: International Aircraft Noise Regulation

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

Victor W. Sparrow, Cochair
*Penn State, 201 Appl. Sci. Bldg., University Park, PA 16802*

**Chair's Introduction—10:00**

### *Invited Paper*

**10:05**

**2aNS1. International aircraft noise regulation.** Alexandra Loubeau (Appl. Acoust. Branch, NASA Langley Res. Ctr., MS 463, Hampton, VA 23681, a.loubeau@nasa.gov) and Victor W. Sparrow (Penn State, University Park, PA)

This panel session addresses the current state and challenges of aircraft noise regulations across the world. Invited experts from industry, government, and research institutions will each give a 10-min live talk, and a panel discussion with the opportunity for questions from the audience participants will follow. Presentations may include discussion of the history of aircraft noise regulations, perspectives on current regulations, recent public survey results, and their implications for future regulations. Discussion of additional future developments could include regulation of new entrant aircraft, such as urban air mobility, supersonic, and electric propulsion vehicles.

## Session 2aPA

## Physical Acoustics: Hot Topics in Physical Acoustics

Andrea P. Arguelles, Cochair
*Eng. Sci. and Mech., Penn State Univ., 212 Earth-Engr Sciences Bldg., University Park, PA 16802*

Lauren Katch, Cochair
*The Pennsylvania State Univ., 212 Earth and Engineering Sciences Bldg., State College, PA 16802*

### *Invited Papers*

**10:00**

**2aPA1. Improved performance of pulse tube refrigerators using thermoacoustics.** Ryan Snodgrass (National Institute of Standards and Technology, 325 Broadway, Boulder, CO 80305, ryan.snodgrass@nist.gov), Vincent Kotsubo, Scott Backhaus, and Joel Ullom (National Institute of Standards and Technology, Boulder, CO)

A wide variety of science is performed at temperatures near and below 4 K. Such low temperatures are commonly achieved using the pulse tube refrigerator, a type of cryocooler that cyclically compresses and expands helium gas to pump heat from the cold end. Here, we discuss three topics demonstrating that thermoacoustic analysis enables substantial gains in the understanding and performance of these refrigerators. We begin by showing that dynamic acoustic optimization of pulse tube refrigerators can lead to a tremendous increase in their cooldown speed (up to 3.5 times the status quo speed). This is a novel technique that is currently being commercialized. Second, we discuss the development of mass flow meters for the study of pulse tube refrigerator compressor efficiency. The flow meters are used to validate a thermoacoustic expression that more simply finds the flow rate using only pressure transducers. Third, we show that a large amount of cooling power may be extracted from the regenerators of these refrigerators without impacting the cooling power available at the cold end. Thermoacoustic analysis reveals that the real-fluid properties of helium are responsible for this capability.

**10:30**

**2aPA2. Dynamic phased arrays and ultrasound modulators.** Peer Fischer (Max Planck Inst. for Med. Res. & IMSEAM Heidelberg Univ., IMSEAM, Im Neuenheimer Fels 225, Heidelberg 69120, Germany, peer.fischer@mr.mpg.de)

The acoustic hologram has made it possible to shape ultrasound waves and project complex pressure patterns in 2D and in 3D, as we have also recently shown. The hologram contains tens of thousands of elements which can be used to form sophisticated pressure patterns by setting the phase and/or amplitude across the wavefront of a sound wave. It thereby offers more degrees of freedom than conventional phased array transducers (PATs). However, the hologram is generally static, whereas PATs are dynamic. PATs do not only have fewer elements, but they are also more difficult to fabricate and require complex electronics. In this talk, I will describe our efforts to combine the best of both worlds. For this, we are exploring the use of light waves as a means to control ultrasound waves. I will describe direct and indirect approaches that can be used to control and modulate holographic ultrasound fields.

**11:00**

**2aPA3. Three-dimensional ultrasound matrix imaging.** Flavien Bureau (Institut Langevin, CNRS, Paris, France), Louise Denis, Antoine Coudert (Laboratoire d'Imagerie Biomedicale, CNRS, Paris, France), Justine Robin (Phys. for Med., INSERM, Paris, France), Mathias Fink (Langevin Inst., ESPCI Paris, Paris, France), Olivier Couture (Lab. d'Imagerie Biomed., CNRS, Paris, France), and Alexandre Aubry (Instit. Langevin, CNRS, 1 rue Jussieu, Paris 75005, France, aaubry.langevin@gmail.com)

Matrix imaging paves the way towards a next revolution in wave physics. Based on the response matrix recorded between a set of sensors, it enables an optimized compensation of aberration phenomena and multiple scattering events that usually drastically hinder the focusing process in heterogeneous media. Although it gave rise to spectacular results in optical microscopy or seismic imaging, the success of matrix imaging has been so far relatively limited with ultrasonic waves because wave control is generally only performed with a linear array of transducers [1]. In this talk, we will extend ultrasound matrix imaging to a 3D geometry [2]. Switching from a 1D to a 2D probe enables a sharper estimation of the transmission matrix that links each transducer and each medium voxel. Here, we first present an experimental proof of concept on a tissue-mimicking phantom through *ex vivo* tissues and then show the potential of our reflection matrix approach for transcranial imaging, with applications to ultrasound localization microscopy of a sheep brain [3]. [1] W. Lambert *et al*., Proc. Natl. Ac. Sci. U. S. A. 117, 14645–14656 (2020) [2] F. Bureau *et al*., Nat. Commun. 14, 6793 (2023) [3] F. Bureau, L. Denis *et al*., (to be submitted).

11:30

**2aPA4. Deep-penetration acoustic volumetric printing.** Y. Shrike Zhang (Harvard Med. School, 65 Landsdowne St., PRB 286, Cambridge, MA 02139, yszhang@bwh.harvard.edu)

Over the last decades, the field of three-dimensional (3D) printing, or additive manufacturing, has witnessed tremendous progress. 3D printing enables precise control over the composition, spatial distribution, and architecture of the printed constructs facilitating the recapitulation of the delicate shapes and structures of target patterns. In particular, volumetric printing, an emerging additive manufacturing technique, builds objects with enhanced printing speed and surface quality by forgoing the stepwise ink-renewal step. Existing volumetric printing techniques almost exclusively rely on light energy to trigger photopolymerization in transparent inks, limiting material choices and build sizes. This talk will discuss our recent efforts in developing a unique additive manufacturing strategy that takes advantage of self-enhancing sono-ink design and corresponding focused-ultrasound writing technique for deep-penetration acoustic volumetric printing (DAVP). Experiments and acoustic modeling were used to study the frequency and scanning rate-dependent acoustic printing behaviors. DAVP achieves the key features of low acoustic streaming, rapid sonothermal polymerization, and large printing depth, enabling the printing of volumetric hydrogels and nanocomposites with various shapes regardless of their optical properties. DAVP also allows printing at centimeter depths through biological tissues, paving the way toward minimally invasive medicine.

WEDNESDAY MORNING, 20 NOVEMBER 2024 — 10:00 A.M. TO 11:41 A.M.

# Session 2aPP

## Psychological and Physiological Acoustics: Lightning Round Competition

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

Varsha H. Rallapalli, Cochair
*Commun. Sci. & Disorders, Northwestern Univ., 2240 Campus Dr., Evanston, IL 60208*

**Chair's Introduction—10:00**

### *Contributed Papers*

10:05

**2aPP1. Diotic narrowband noises can be perceived as intracranially off center in listeners with symmetrical audiometric thresholds.** Obada J. AlQasem (Hearing and Speech Sci., Univ. of Maryland-College Park, 8700 Baltimore Ave., College Park, MD 20740, obada.j.oj@gmail.com) and Matthew J. Goupell (Hearing and Speech Sci., Univ. of Maryland-College Park, College Park, MD)

The ability to localize sound sources in the horizontal plane depends on the interaural time (ITD) and level (ILD) difference. It is assumed that individuals with normal and symmetrical hearing thresholds perceive stimuli with zero ILD and ITD as fused centered auditory images, which is not always true. This study aimed to explore ILD lateralization biases in individuals with normal hearing and explain whether they can be explained by an interaural asymmetry in monaural loudness. An ILD lateralization task was performed where the ILDs ranged between $\pm$10 dB. The stimulus was a 1-equivalent-rectangular-bandwidth narrowband noise centered at 250–8000 Hz or a wideband noise. Preliminary results showed that the participants exhibited ILD lateralization biases more often at higher frequencies and minimal bias in the wideband condition. These biases were not caused by headphones, as the transducers were reversed between testing blocks. Biases were also consistent across up to three separate testing days. A loudness balancing task showed that participants perceived stimuli as equally loud, ruling out interaural asymmetry in monaural loudness. Audiometry also confirmed the absence of hearing threshold asymmetries. Therefore, lateralization biases in individuals with normal hearing are not explained by interaural disparities in loudness perception or hearing threshold asymmetry.

10:13

**2aPP2. Language environment analysis precision in real-world and controlled environments.** Rachael Pennock (Northwestern Univ., 2240 Campus Dr., 1-451, Evanston, IL 60208, rachaelpennock2028@u.northwestern.edu), Varsha H. Rallapalli, and Pamela E. Souza (Northwestern Univ., Evanston, IL)

Background: Adults with hearing loss experience individual auditory ecologies, encountering unique environments and listening demands. A clinician's subjective understanding of each patient's auditory ecology informs clinical decision-making. As subjective interpretation is subject to error, objective measurements would improve accurate auditory ecology characterization. The Language Environment Analysis (LENA) system is a promising tool to evaluate older listeners' environments to aid clinical decision-making. However, most LENA research has focused on adult-child conversations in quiet. This project aims to determine LENA's accuracy in capturing real-world acoustic environments. Methods: Measurements were gathered of an adult female in real-world and controlled (sound-treated booth) speech-in-noise environments. The LENA and a

Brüel & Kjaer sound level meter were placed opposite the talker. Overall levels and signal-to-noise ratios (SNR) were estimated by analyzing intensities of segment categories over varying time periods. Results: Similar noise levels and SNRs were found for analysis windows from 5 to 15 min. In the real-world environments, noise levels varied between 61 to 86 dBA, and SNR varied between $-0.9$ to $+3.0$ dB. However, values were misestimated by approximately 5 dB compared to sound level meter levels. Implications: Although they do not reach laboratory precision, LENA-informed measures could provide insight into individuals' auditory ecologies.

**10:21**

**2aPP3. Two cues are better than one: Adding interaural time differences improves spatial hearing for bilateral cochlear-implant listeners.** 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)

Bilateral cochlear-implant (BI-CI) listeners show limited sensitivity to interaural time differences (ITDs), the dominant localization cue for acoustic-hearing listeners, and mainly rely on interaural level differences (ILDs) for localization. Studies utilizing bilaterally synchronized direct stimulation capable of conveying ITDs have investigated sensitivity to ITDs or ILDs in isolation. It is, however, unclear how these two cues interact in controlled and bilaterally synchronized electrical stimulation. Therefore, this study performed an ITD-ILD cue-weighting lateralization experiment with BI-CI listeners using direct stimulation of single electrodes. Preliminary results show that BI-CI listeners display sensitivity only to ILDs with unsynchronized stimulation and equal sensitivity to both ITDs and ILDs with synchronized stimulation. Additionally, providing ITDs and ILDs together via bilaterally synchronized stimulation resulted in increased lateralization ranges and slopes, thus improved spatial hearing acuity compared to either cue alone. These results suggest providing both ITDs and ILDs via bilaterally synchronized sound processors has the potential to improve spatial hearing in BI-CI listeners. The data have implications for clinical sound processor design and stimulation strategies.

**10:29**

**2aPP4. Out-of-focus: Fuzzy cochlear implant stimulation and how to avoid it.** Dietmar M. Wohlbauer (Dept. of Otolaryngol., Head and Neck Surg., Harvard Med. School, 44 Marion St., Somerville, MA 02143, dm.wohlbauer@gmail.com), Charles Hem (Harvard Univ., Boston, MA), Caylin McCallick, Faten Awwad (Dept. of Audiol., Massachusetts Eye and Ear, Boston, MA), and Julie G. Arenberg (Dept. of Otolaryngol., Head and Neck Surg., Harvard Med. School, Boston, MA)

Cochlear Implants restore hearing in severe-to-profound hearing-impaired individuals. An array of electrodes placed in the inner ear serves as the interface to elicit sound perception via electrically presented pulses. Cochlear implant performance is reduced by channel interaction, which degrades the signal accuracy at the electrode-neuron interface. We developed and optimized a focused cochlear implant stimulation strategy that counteracts the fuzzy electric representation by reducing channel interaction thereby getting the signal back into focus. The proposed strategy combines two approaches to minimize channel interaction, dynamic focusing, and electrode deactivation, which have individually proven beneficial for speech performance. Dynamic focusing concentrates the electric fields of electrodes located far from the modiolus by applying stimulation currents on triplets of simultaneously activated electrodes, with one center electrode and two flanking returns. Deactivated electrodes reduce the negative influence of high currents when stimulating possible neural dead regions along the cochlea. Speech perception outcomes in background noise showed, that the overall performance increases when high-threshold electrodes are dynamically focused and deactivated. The suggested strategy, therefore, might be a promising approach to improve the signal accuracy in cochlear implant stimulation.

**10:37**

**2aPP5. Talker head orientation related benefit as a function of masker-talker head angle.** Vahid Delaram (Speech and Hearing Sci., Univ. of Illinois Urbana-Champaign, 901, S 6th St., Champaign, IL 61820, vahidd2@illinois.edu), Allison Trine (Speech and Hearing Sci., Univ. of Illinois Urbana-Champaign, Champaign, IL), Rohit M. Ananthanarayana (Speech and Hearing Sci., Univ. of Illinois Urbana-Champaign, Champaign, IL), Emily Buss (Dept. of Otolaryngol./HNS, Univ. of North Carolina at Chapel Hill, Chapel Hill, NC), G. Christopher Stecker (Ctr. for Hearing Res., Boys Town Natl. Res. Hospital, Omaha, NE), Margaret Miller (Ctr. for Hearing Res., Boys Town Natl. Res. Hospital, Omaha, NE), and Brian B. Monson (Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, Champaign, IL)

Understanding speech in multi-talker environments poses significant challenges as multiple speech signals compete for attention. One factor that plays a role in speech recognition in multi-talker environments is talker head orientation. In this study, we investigated the effect of talker head orientation related (THOR) spectral cues on speech recognition using eight different masker head orientation angles ranging from 0° (facing the listener) to 180° (facing away from the listener). The target talker was facing the listener in all conditions and maskers were co-located with the target. Target speech was digit triplets from a female talker, with a female two-talker masker consisting of narrative speech. A speech reception threshold was calculated for each condition as the mean of 6 reversals in a two-down one-up adaptive track. A THOR benefit was observed when maskers faced >45°. This benefit continually increased as masker head angle increased to 180°. Low-pass filtering stimuli at 8 kHz somewhat reduced the THOR benefit. The data indicate that the THOR benefit from spectral cues can be >6 dB for the highest masker head angles but could reach >10 dB when additionally accounting for masker level reductions associated with masker head rotation. [Work supported by NIH grant R01DC019745.]

**10:45**

**2aPP6. Facial expression analysis and auditory perception.** Alessandro Braga (École de technologie supérieure, Montreal, QC, Canada, alessandro.braga@etsmtl.ca), Charlotte Bigras (Univ. de Montréal, Montreal, QC, Canada), Arian Shamei (École de technologie supérieure, Montreal, QC, Canada), Sylvie Hébert (Université de Montréal, Montreal, QC, Canada), and Rachel Bouserhal (École de technologie supérieure, Montréal, QC, Canada)

The psychoacoustic assessment of loudness disorders like Hyperacusis relies on subjective reports, which are biased and impractical for non-verbal patients. Similar issues in pain evaluation have been addressed using facial expression analysis to decode the intensity and affective value of perceived pain. Thus, we are developing a system for the objective evaluation of perceived sounds from a listener's facial expression. We determine whether sensory (intensity) and affective (valence) dimensions of sound perception can be distinguished through facial expressions by employing action unit analysis and other facial feature extraction methods on our in-house dataset. This video dataset includes facial expressions in response to sounds with varying intensities and emotional valences. We train convolutional networks to decode these dimensions from both extracted features, such as AUs and raw video data. By employing feature and decision fusion methods, we are developing an automated system for the objective assessment of perceived sound. This system will pioneer the first objective method for assessing loudness disorders, enhancing patient care and potentially distinguishing between hyperacusis sub-types to facilitate personalized treatment.

**10:53**

**2aPP7. Perceptual categorization of and adaptation to human voice and musical instruments: A passive-listening study.** Zi Gao (Dept. of Psychol., Univ. of Minnesota, 75 E River Rd., Minneapolis, MN 55455, gao00196@umn.edu) and Andrew J. Oxenham (Dept. of Psychol., Univ. of Minnesota, Minneapolis, MN)

The human voice is a highly social and relevant auditory stimulus. Previous studies have observed a wide range of voice-sensitive effects, but less

is known about the role of attention and context when categorizing sounds as either voice or non-voice. To address this gap, the current study adopted electroencephalography (EEG) passive listening tasks. In Experiment 1, vowel utterances (/a/, /o/, /u/, and /i/) and instrumental tones (bassoon, horn, saxophone, and viola) were presented with equal probability in a random sequence, and different brain responses to the two categories were observed. In Experiment 2, mismatch negativity was observed for rare instrumental tones (viola) embedded in a random sequence of four different vowels, but not vice versa, suggesting that categorization of voice and non-voice could require little to no attention but may be modulated by stimulus familiarity. In Experiment 3, ambiguous voice-instrument morphs were presented in either vocal or instrumental contexts. Logistic regression models performed above chance in predicting the type of context (voice or instrument) from the responses to ambiguous morphs. The results suggest that neural signatures of both perceptual categorization (voice/non-voice) and context effects can be observed in EEG responses under passive listening conditions. [Work supported by NIH grant R01 DC012262.]

**11:01**

**2aPP8. Age-related effects on emotional reactions to sounds in individuals with misophonia.** Namitha Jain (Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, 901 S Sixth, Champaign, IL 61820, namitha3@illinois.edu), Gibbeum Kim (Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, Champaign, IL), Jae Wan Choi (Psychology, Univ. of Illinois Urbana-Champaign, Champaign, IL), Fatima T. Husain (Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, Champaign, IL), and Howard Berenbaum (Psychology, Univ. of Illinois Urbana-Champaign, Champaign, IL)

Misophonia, characterized by intolerance to specific sounds, triggers emotional reactions like disgust, anger, and distress. While misophonia typically begins in childhood/adolescence, its nature across the lifespan is understudied. Our study assesses audiological correlates and emotional reactions to sounds in misophonia across age groups. We recruited three groups: young misophonia (YM: 18–29 years, N = 14), mid-to-old misophonia (OM: 30–67 years, N = 17), and young controls (YC: 18–25 years, N = 26). All underwent hearing assessments (0.25–16kHz). Participants classified typical trigger and other affective sounds based on eliciting any misophonic reactions and rated their valence on a neutral to extremely unpleasant scale. Results show age-related effects in OM, with poorer hearing (>4 kHz) compared to YM and YC. Both groups had slower response times to trigger sounds compared to non-trigger sounds, suggesting a greater cognitive and emotional load. We found group differences on valence rating and reaction times. OM perceived sounds to be more unpleasant and responded slower to sounds relative to YM, possibly due to age or hearing loss effects. These findings suggest that misophonia may interact with auditory processing of emotions that are influenced by age and hearing acuity. Note that these results are preliminary and data collection is ongoing.

**11:09**

**2aPP9. Exploratory acoustic analysis of nonverbal affective and communicative vocalizations from non-speaking individuals with autism using cosine similarity.** Siddhant Bikram Shah (Elec. & Comput. Eng., Northeastern Univ., Huntington Ave., Boston, MA 02115, shah.siddhantb@northeastern.edu) and Kristina T. Johnson (Elec. & Comput. Eng., Northeastern Univ., Boston, MA)

Nonverbal vocalizations from non-speaking individuals are challenging to interpret without context or familiarity. This study assessed acoustic similarities between 4727 affective and communicative vocalizations (e.g., frustration, delight, request) from 8 non-speaking individuals with autism using the ReCANVo dataset. We augmented the audio with noise, pitch shifts, and tempo changes, and extracted features using pre-trained Wav2Vec2 model representations, mel-spectrograms, and low-level descriptors to train a classification network (dim 24; 85/15 train/val). We calculated a mean vector (dim 24) for each class and computed their cosine similarities. Across eight individuals and seven vocalization classes, "Social" vocalizations were most globally similar (AvgCS = 0.950), suggesting acoustic features spanning affective and communicative states, while "Dysregulated" sounds were the most distinct (AvgCS = 0.886). However, the model (Acc = 0.660, MacroF1 = 0.555) was sensitive to speaker gender and age, and class imbalances, even after augmentation. Examining the participant with the most samples (P01; n = 1687; Acc = 0.676; MacroF1 = 0.695), "Request" was most similar to "Social" (CS = 0.978), suggesting shared underlying acoustics. "Delightedi" and "Frustrated" were the least similar (CS = 0.715), supporting the hypothesis that affective vocalizations are discernable via acoustical properties. Increased sample sizes and better real-world labeling techniques will enhance model accuracy, enabling interactive augmentative technologies for this population.

**11:17**

**2aPP10. The role of envelope in rippled spectra discrimination.** Dmitry I. Nechaev (Inst. of Ecol. and Evol. RAS, Inst. of Ecol. and Evol., 33 Leninsky Prospect, Moscow 119071, Russian Federation, dm.nechaev@yandex.ru), Olga N. Milekhina (Institute of Ecology and Evolution RAS, Moscow, Russian Federation), Marina Tomozova (Inst. of Ecol. and Evo. RAS, Moscow, Russian Federation), and Alexander Supin (Inst. of Ecol. and Evol., Moscow, Russian Federation)

In previous studies, it was hypothesized that discrimination between noise with a rippled spectrum and non-rippled noise could be explained by an autocorrelation temporal processing model. Another possible mechanism is the detection of signal envelope modulation. To test this hypothesis, band-limited noise with a rippled spectrum was used as a test signal, with ripples equally spaced on a logarithmic scale. The signal's envelope spectrum had several low-frequency peaks corresponding to the intervals between ripples. The test aimed to find the maximum ripple density (rip/oct) at which listeners could detect phase reversals in the spectrum. Non-rippled noise served as a reference signal. Both test and reference signals were amplitude-modulated by a sinusoid with a frequency matching the intervals between ripples of maximum power. This amplitude modulation reduced ripple resolution by more than half for all ripple widths. Without amplitude modulation, decreasing the ripple width from 37% to 28% of the inter-ripple frequency spacing increased resolution from 40 to 70 rip/oct, with further narrowing producing a minimal effect. With amplitude modulation, narrowing the ripples resulted in incremental improvements in resolution. [Work supported by The Russian Science Foundation, Grant 23-25-00148.]

**11:25**

**2aPP11. Influence of the osseous spiral lamina on the inner ear's response to air- and bone-conducted stimulation.** Simon Kersten (Inst. for Hearing Technol. and Acoust., RWTH Aachen Univ., Kopernikusstraße 5, Aachen 52074, Germany, simon.kersten@akustik.rwth-aachen.de), Henning Taschke (formerly at: Inst. of Commun. Acoust., Ruhr Univ. Bochum, Bochum, Germany), and Michael Vorlaender (Inst. for Hearing Technol. and Acoust., RWTH Aachen Univ., Aachen, Germany)

Recent studies have provided valuable insight into the anatomy and motion of the cochlear partition, particularly emphasizing the flexible nature of the osseous spiral lamina (OSL). However, whether in studies involving air- or bone-conducted stimulation, the OSL is often treated as a rigid structure, overlooking its potential impact on the cochlear macro mechanisms. In this study, we investigate the effect of the OSL's flexibility by incorporating it as either a rigid or a flexible structure in a finite element model of the human inner ear. By applying both air- and bone-conducted stimulation, we found that the OSL's flexibility significantly influences cochlear impedances, cochlear partition stiffness, and the differential fluid flow at the oval and the round windows. These results emphasize the importance of considering the flexibility of the OSL for an improved understanding of cochlear function and interpretation of experimental data on inner ear responses. [This work was funded by the Deutsche Forschungsgemeinschaft (DFG, German Research Foundation) – Project-ID 352015383 – SFB 1330 A4.]

11:33

**2aPP12. Effects of signal bandwidth and signal-to-noise ratio on noised-masked gap detection.** Zenzele Thomas (Med. Eng., Univ. of South Florida, 4202 E. Fowler Ave., Tampa, FL 33612, thomasz16@usf.edu), David A. Eddins (Commun. Sci. and Disorders, Univ. of Central Florida, Tampa, FL), and Erol J. Ozmeral (Commun. Sci. and Disorders, Univ. of South Florida, Tampa, FL)

Speech perception relies on our ability to process fast modulations in a signal. Such temporal processing often is probed using a temporal gap detection task. Because speech-in-noise listening is a common challenge for listeners with hearing loss, we investigated the role of the periphery to process temporal gaps in the presence of noise. Three experiments were conducted using a classic gap detection threshold method, including a 3-alternative-forced-choice procedure and a 3-down-1-up adaptive tracking method to estimate thresholds. Experiments 1 and 2 tracked gap detection (in ms) for various signal bandwidths, signal-to-noise ratios (SNRs), and masker modulation depths. Experiment 3 fixed the gap duration, then tracked gap modulation depth (in dB) at various SNRs. Experiments 1 to 3 showed similar effects of SNR, with poorer SNRs increasing gap thresholds. Masker bandwidth also reduced performance in Experiment 2. Current analyses with a common modulation filter bank frontend (Dau *et al.*, 1997) performed on par as the normal hearing listeners suggesting that auditory gap detection and, by extension, temporal envelope processing, can be explained by peripheral processes. [Work supported by NIH R01 DC020514 and R01DC015051.]

WEDNESDAY MORNING, 20 NOVEMBER 2024 — 10:00 A.M. TO 12:00 NOON

# Session 2aSA

## Structural Acoustics and Vibration: Careers in Structural Acoustics and Vibrations

Christina Naify, Cochair
*Appl. Res. Lab., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758*

Alexey Titovich, Cochair
*Naval Surface Warfare Ctr., Carderock Div., West Bethesda, MD*

Benjamin S. Beck, Cochair
*Eng. Acoust., Penn State Appl. Res. Lab., P.O. Box 30, MS 3200D, State College, PA 16804*

Feruza Amirkulova, Cochair
*Mech. Eng., SAn Jose State Univ., 1 Washington Sq., San Jose, CA 95192*

**Chair's Introduction—10:00**

### *Contributed Paper*

10:05

**2aSA1. Careers in structural acoustics and vibrations.** Feruza Amirkulova (Mech. Eng., SAn Jose State Univ., 1 Washington Sq., San Jose, CA 95192, feruza.amirkulova@sjsu.edu)

The structural acoustics and vibrations (SAV) technical committee focuses on research and industry which couple fluid acoustics to structural dynamics. The committee is made up of members from industry, academia, and government whose areas of study cover a wide range of topic areas including but not limited to metamaterials, naval applications, architectural acoustics, and nondestructive evaluation. The training and career paths for those engaged in SAV research and practice is widely varied and this panel discussion will provide insight into the types of careers available which feature SAV. The panel discussion will include panelists Kathryn Matlack (The University of Illinois, Urbana-Champagne), Junfei Li (Purdue University), Amanda Hanford (Applied Research Labs, Penn State University), and Matthew Craun (Naval Surface Warfare Center, Carderock Division). Panelists will answer questions about their careers posed by panel moderators and the audience.

## Session 2aSC

## Speech Communication: In Honor of Ken Stevens' 100th Birthday

Abeer Alwan, Cochair
*66-147E Eng. IV, 420 Westwood Plaza, Los Angeles, CA 90095*

Maria-Gabriella Di Benedetto, Cochair
*DIET, Sapienza Univ. of Rome, Via Eudossiana 18, Rome, 00184, Italy*

Stefanie Shattuck-Hufnagel, Cochair
*Cambridge, MA*

**Chair's Introduction—10:00**

### *Invited Papers*

**10:05**

**2aSC1. Studying fricatives with Ken Stevens and ever since.** Christine H. Shadle (Yale Child Study Ctr., Yale Univ., 300 George St., Ste. 900, New Haven, CT 06511, shadle@haskins.yale.edu)

I began studying fricatives as Ken Stevens' Ph.D. student at MIT, following John Heinz' experiments and Ken's 1971 JASA paper. Experiments with simple mechanical models allowed source location and spectra to be measured. Using these sources in transmission-line analogues showed predicted and measured far-field spectra matched, validating the use of such models for voiceless fricatives. Parameters that captured the acoustic effects of changes in model dimensions and airflow distinguished sibilant and non-sibilant cases, and the place of the constriction. In the years since, I have used aerodynamic measurements (e.g., Rothenberg mask, hot-wires) to explore the interaction of convection and acoustic velocities in the vocal tract. These phenomena are important in voiced fricatives and whistling and are ignored in classic models. Signal processing techniques (ensemble averaging, multitaper analysis, spectral decomposition) have, e.g., demonstrated the interaction of phonation and turbulence sources in voiced fricatives. Spectral parameters allowed comparison of fricatives of different languages, or phonetic contexts; of speakers of different ages and genders, and with speech disorders. My goal has shifted from improving articulatory synthesis to defining acoustic parameters that allow production parameters to be inferred. Open questions and the many links to Ken's lab and collaborators will be described.

**10:20**

**2aSC2. Ken Steven's focus on speech production and our development of the speech inversion system.** Carol Espy-Wilson (Elec. and Comput. Eng., Univ. of Maryland, 8223 Paint Branch Dr., AVW Bldg., College Park, MD 20742, espy@umd.edu), Yashish M. Siriwardena (Elec. and Comput. Eng., Univ. of Maryland College park, College Park, MD), Suzanne Boyce (Commun. Sci. and Disorders, Univ. of Cincinnati, Cincinnati, OH), Mark Tiede (Haskins Lab., New Haven, CT), and Liran Oren (Otolaryngol., Univ. of Cincinnati, Cincinnati, OH)

Throughout his highly distinguished career, Ken Stevens studied how the relationship between the constrictions used in speech production result in the acoustic properties of the speech signal. Ken's pioneering rate in articulation stemming from the X-ray study he participated in and studies by his students in the timing and movement of articulatory gestures played a pivotal role in our work to develop a Speech Inversion (SI) system. The SI system, using deep learning, initially used ground-truth information obtained from the Wisconsin X-ray Microbeam database to estimate the articulatory trajectories for the lips, tongue tip, and tongue dorsum. Adding glottal information based on an aperiodicity/periodicity/pitch detector resulted in a significant improvement in the correlation between estimated and ground truth data. More recently, we added ground truth articulatory data about the velopharyngeal port constriction, using both nasometry and the novel technique of high-speed nasopharyngoscopy. In this talk, I will discuss the SI system and its ability to help us understand how speech production changes as a result of mental state, speech disorder, speaking rate, and accent.

**10:35**

**2aSC3. Neurocomputational analysis of impaired auditory-motor function for laryngeal motor control.** Matias Zanartu (Dept. of Elec. Eng., Univ. Tecnica Federico Santa Maria, Av. Espana 1680, Valparaiso 2390123, Chile, matias.zanartu@usm.cl)

Recent research indicates that impaired auditory-motor function may be a significant etiological factor in the development of hyperfunctional voice disorders (VH). To investigate this hypothesis, our group has introduced Laryngeal DIVA (LaDIVA), a neurocomputational framework that currently integrates a triangular body-cover vocal fold model within the physiologically validated DIVA model of speech motor control. LaDIVA is being used to investigate laryngeal motor control in terms of pitch regulation, responses to environmental noise, and voice quality. In this study, we examined the adaptation dynamics of speaking in noise among 20 participants with

typical voices and 20 with VH. Participants were instructed to utter a series of syllables under three conditions: baseline (quiet environment), Lombard (speech noise at 80 dB SPL), and recovery (quiet environment after five minutes of rest). The results revealed that participants with VH did not return to baseline after speaking under masking noise. Our model analysis indicates that participants with VH rely more heavily on auditory feedback to update the feedforward process when speaking in noise compared to those with typical voices. This increased reliance suggests that individuals with VH require more time to adapt to changes in auditory environments, which may contribute to the development of VH.

## *Contributed Paper*

**10:50**

**2aSC4. From intention to understanding and back again: How a simple message of "Catch and Pass" can build language in children.** Jennell Vick (Cleveland Hearing and Speech Ctr., 6001 Euclid Ave., Cleveland, OH 44103, jvick@chsc.org), Rebecca Mental, and Michael Rollins (Cleveland Hearing and Speech Ctr., Cincinnati, OH)

Project ELLA (Early Language and Literacy for All) is a program of Cleveland Hearing and Speech Center (CHSC) that aims to improve language and preliteracy outcomes for children from birth to age 5 and their caregivers in Cleveland communities. Project ELLA uses a public health approach that involves employing community health workers (CHWs) to increase awareness of the impact of responsive interaction for building language in infants and toddlers using a simple message: "Catch and Pass," CHWs are in the community to provide training and build trust with families and to identify and refer children who need clinical services. Project ELLA also provides no-cost speech-language therapy and support for caregivers at CHSC locations and through outreach programs. Project ELLA evaluates its impact by tracking the number of children and families served, the progress of children in therapy, the knowledge and skills of caregivers and teachers, and the established partnerships in the community. Project ELLA is a novel and innovative model that can transform the landscape of language and literacy development in Cleveland and beyond. It amplifies the basics of The Speech Chain in the language-building role of caregivers of young children.

## *Invited Papers*

**11:05**

**2aSC5. Ken Stevens and motor theories.** Patricia K. Kuhl (Inst. for Learning & Brain Sci., Univ. of Washington, Malistop 357920, Seattle, WA 357920, pkkuhl@uw.edu)

At an ASA meeting long ago, Ken Stevens said he had a confession to make, one that I wouldn't like. "I think I'm becoming a motor theorist," he said with a little laugh. I registered the surprise he expected and said, "*Traditional* Motor Theory?" He replied, "not exactly," and described the sensory and motor components of Quantal Theory. In today's talk, I'll review preliminary data from a collaboration between my laboratory and neuroscientists at Washington University (Marc Raichle, Abraham Synder) and Stanford University (Anish Mitra). We are using resting state fMRI experiments to measure spontaneous infra-slow brain activity representing information flow between sensory (Wernicke's) and motor (Broca's) brain areas in newborns, 6-, 12-, and 24-month-old infants, as well as adults, and to link information flow in early development to a child's emergent language skills. If supported, our hypothesis will help explain infants' abilities to integrate sensory and motor information early in development. Moreover, our data have the potential to substantially alter current theories, creating a sensorimotor theory of speech development, one that I believe Ken Stevens (and traditional motor theorists like Al Liberman) would welcome.

**11:20**

**2aSC6. Speech feature encoding in the human brain.** Edward Chang (UCSF, 513 Parnassus Ave., San Francisco, CA 94611, edward.chang@ucsf.edu)

The human superior temporal gyrus (STG) is a high-order auditory cortex area that is critical for speech perception. Over the past decade, cortical recordings directly from the human brain procedures have provided the most detailed information yet on the representations and computations that give rise to speech perception. Inspired by Ken Stevens' theory of distinctive features, we have discovered several defining aspects of cortical processing of speech: (1) representation of acoustic-phonetic features, (2) encoding of prosodic cues that demarcate the timing and magnitude of syllables (peak derivative of the amplitude envelope) and sentence/phrasal onsets, and (3) speaker normalized representation of vowels in 2-dimensional formant space. Perceptual studies on phoneme restoration and selective attention have demonstrated that cortical processing in the STG is directly related to our conscious perception of speech sounds and not a veridical read-out of the stimulus. Our new work seeks to address phonological specialization in the STG based on language experience, and the representation of words.

## *Contributed Paper*

**11:35**

**2aSC7. Toward the long-term evolution of the xkl speech analysis software.** Maria-Gabriella Di Benedetto (DIET, Sapienza Univ. of Rome, Via Eudossiana 18, Rome 00184, Italy, mariagabriella.dibenedetto@uniroma1.it), Luca De Nardis, Kaleem Kashif (DIET, Sapienza Univ. of Rome, Rome, Italy), Jeung-Yoon Choi, and Stefanie Shattuck-Hufnagel (MIT, Cambridge, MA)

A legacy software, "xkl," developed in the 1980's by Dennis Klatt in Kenneth N. Stevens' Speech Communication Group at the Massachusetts Institute of Technology (MIT) has been recently revamped, thanks to a joint effort between MIT and Sapienza University of Rome (https://www.sciencedirect.com/science/article/pii/S2352711023001887). This collaboration led to the development of a modern xkl, which is now compatible with modern computing platforms (Windows, macOS, Linux) (more on current software developments can be found at http://acts.ing.uniroma1.it/xkl.php). The original development was based on Motif libraries, but these libraries currently are not being updated. To ensure long-term compatibility, the xkl graphical user interface is currently being redesigned using GTK. Moreover, the software will include the possibility of computing the reassigned spectrogram,

enriching the available time-frequency representations, and facilitating the process of performing detailed analyses of the properties of speech, such as formant estimation. The xkl software has superior qualities for acoustic parameter estimation, in particular, determination of spectral peaks and their evolution in time, thanks to a flexible interface that allows the investigator to conveniently sweep through successive spectral slices. This new-xkl is currently under use in preliminary analyses of vowels in the LaMIT dataset, a set of 100 sentences in Italian (see https://www.sciencedirect.com/science/article/pii/S2352340922004772).

**11:50–12:00 Discussion**

WEDNESDAY MORNING, 20 NOVEMBER 2024 10:00 A.M. TO 12:00 NOON

## Session 2aSP

## Signal Processing in Acoustics: Tutorial on Machine Learning for Acoustics

Peter Gerstoft, Cochair
*Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093*

Michael J. Bianco, Cochair
*Marine Phys. Lab., Univ. of California San Diego, Scripps Inst. of Oceanogr., 9500 Gilman Dr., MC 0238, La Jolla, CA 92037*

Ryan A. McCarthy, Cochair
*Marine Phys. Lab., Scripps Inst. of Oceanogr., Univ. of California San Diego, 9500 Gilman Dr. #0214, La Jolla, CA 92093*

Neil Zhang, Cochair
*Univ. of Rochester, 500 Wilson Blvd., Rochester, NY 14620*

### *Invited Paper*

**10:00**

**2aSP1. Tutorial on machine learning for acoustics.** Peter Gerstoft (Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093, gerstoft@ucsd.edu), Michael J. Bianco (Clarify, San Diego, CA), Ryan A. McCarthy (Univ. of California, San Diego, La Jolla, CA), and Neil Zhang (Dept. of Elec. & Comput. Eng., Univ. of Rochester, Rochester, NY)

Acoustic data provide scientific and engineering insights in fields ranging from biology and communications to ocean and Earth science. We survey the recent advances and transformative potential of machine learning (ML), including deep learning, in the field of acoustics. ML is a broad family of techniques, which are often based in statistics, for automatically detecting and utilizing patterns in data. We have ML examples from ocean acoustics, room acoustics, and personalized spatial audio. For room acoustics, we take room impulse responses (RIR) generation as an example application. For personalized spatial audio, we take head-related transfer function (HRTF) up-sampling as examples. The tutorial will conclude with a set of Jupiter notebook examples on GitHub demonstrating ML benefits.

# Session 2aUW

## Underwater Acoustics: Instrumentation/Lab Show and Tell

Aubrey Espana, Cochair

Jie Yang, Cochair
*Appl. Phys. Lab., Univ. of Washington, 1015 NE 40th St., Seattle, WA 98105*

### *Invited Papers*

**10:00**

**2aUW1. Multi-Sensor Towbody: A down looking and side looking platform for Detection, Classification, & Localization of inert munitions.** Kevin L. Williams (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, klw8@uw.edu) and Timothy Marston (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

MuST comprises a FOCUS-3 towbody manufactured by MacArtney Underwater Technology, two sonar systems manufactured by EdgeTech, a multi-beam system from RESON, a suite of towbody orientation and motion sensors, and a shipboard handling system. The baseline Multi-Sensor Towbody (MuST) hardware and operation are described. Next, shipboard data acquisition/visualization and post mission analysis are presented. This includes examples of the data products currently being used for detection and classification. Finally, the detection/localization/classification performance is quantified using data acquired over blind test beds placed in Sequim Bay, Washington, by the Pacific Northwest National Laboratory. The test beds had 20 to 50 objects on and in the sediment within a 100 m diameter region. Both the number of objects and their positions within the region were unknown to the MuST field team. Results indicate that the hardware implementation and current software processing chain allows classification of inert munitions, with diameters at or above 81 mm, with a probability of correct classification of approximately 85%. Furthermore, the difference between ground truth and MuST geolocations were 2 meters or less for over 90% of the targets detected. [Work sponsored by ESTCP.]

**10:30**

**2aUW2. The optical passive acoustic adaptive drifter system: A multisensor dynamically drifting mid-water biosurveillance platform.** Aaron M. Thode (Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., MC 0206, La Jolla, CA 92093, athode@ucsd.edu), Eric Berkenpas, Mike Shepard (Second Star Robotics, Silver Spring, MD), Dieter Bevans (NUWC Keyport, Keyport, WA), Alison B. Laferriere, Kevin Souhrada, Matthew Alford (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA), and Lauren Freeman (NUWC Newport, Newport, RI)

Conducting unobtrusive observations of the physical, acoustic, and biological properties of the mesopelagic zone remains challenging, despite the proliferation of mobile ROV, AUV, and glider platforms. Here, we discuss the Optical Passive Acoustic Adaptive Drifter System (OPAADS), a 2 m tall Lagrangian drifter platform that incorporates a CTD, stereo video cameras, vertical and tetrahedral hydrophone arrays, and acoustic vector sensors. The platform has also suspended a 91-m vertical thermistor array and acoustic doppler current profiler. The OPAADS uses a commercial buoyancy engine to dynamically adjust its target isobar via either a preprogrammed dive sequence or real-time commands relayed by an ultra-short baseline underwater tracking and communication system. Three platforms have been built and deployed over ten times since 2022, off both southern California and off the New England Seamounts, between depths of 100 and 700 m for durations between 24 and 72 h. We discuss the sensors, operations, and observations collected by OPAADS, including data useful for (1) characterizing bioacoustic chorusing, wind-driven surface noise directionality, and turbulence dynamics; (2) conducting species identification in the deep scattering layer; (3) performing marine mammal tracking; and (4) collecting pilot data studying potential approaches for long-range underwater navigation. [Work sponsored by ONR TFO.]

**11:00**

**2aUW3. Archival, autonomously-deployed acoustic monitoring systems at the University of New Hampshire.** Thomas E. Blanford (Univ. of New Hampshire, Durham, NH 03824, thomas.blanford@unh.edu), Anthony P. Lyons, Jennifer Miksis-Olds, Jenna Hare, and Gabriel R. Venegas (Univ. of New Hampshire, Durham, NH)

The ocean is continuously changing, and multiple dynamic processes influence the underwater environment over time scales that may vary from minutes to years. As the generation, propagation, and scattering of sound in the ocean is closely tied to the environment, underwater acoustic signals also change over multiple time scales. The Center for Acoustics Research and Education at the University of New Hampshire is developing and deploying autonomous active and passive acoustic sensing platforms to conduct long-term monitoring of the environment in the Gulf of Maine. The platforms are deployed from surface vessels and rest on the seafloor, making measurements at regular intervals. The platforms vary in configuration but contain acoustic sensors (passive hydrophones and active, split-aperture echosounders at multiple frequencies, current profilers) and non-acoustic sensors (cameras, CTDs). This presentation will provide an overview of these platforms, their deployment, operation, and applications of the data to understand temporal changes in the underwater environment.

**11:30**

**2aUW4. Shallow water mid-frequency ambient noise observations off southern California.** William Hodgkiss (Scripps Inst. of Oceanogr., Univ. of California, San Diego, CA, whodgkiss@ucsd.edu) and David Ensberg (Scripps Inst. of Oceanogr., Univ. of California, San Diego, CA)

The shallow water environment is challenging for geoacoustic inversions due to site-specific complex bathymetry and sub-bottom characteristics as well as time-evolving oceanography. Shallow water mid-frequency (0.5–10 kHz) ambient noise and source tow transmission observations are presented that were collected with a 2D array (512 elements arranged in 8 vertical staves of 64-elements each all half-wavelength spaced at 6 kHz). The data were collected in August-September 2021 WSW of the Ports of Los Angeles and Long Beach and NW of Santa Catalina Island. The water depth at the array site was 311 m with the bathymetry gradually becoming deeper to the north and was ∼600–900 m deep in the east-west region for ships transiting in/out of LA/Long Beach. The mid-frequency observations included tonal source tow transmissions for characterizing propagation in the region, the radiated signatures of ships transiting the area, and the relatively quiet periods between. The 2D array geometry facilitates decomposing the acoustic field in azimuth as well as elevation. The long-term intent is to implement azimuthally and range-dependent geoacoustic inversions for the seafloor properties in this region as well as to compare shipping as sources of opportunity to the tonal transmissions.

WEDNESDAY AFTERNOON, 20 NOVEMBER 2024 — 3:00 P.M. TO 4:00 P.M.

# Session 2pID

## Interdisciplinary: Keynote Lecture

Micheal Dent, Chair
*Univ. at Buffalo, Suny, B76 Park Hall, Buffalo, NY 142260*

**Chair's Introduction—3:00**

### *Invited Paper*

**3:05**

**2pID1. Vertebrate bioacoustics: A tale of three theories.** W Tecumseh Fitch (Behav. and Cognitive Biol., Univ. of Vienna, Djerassiplatz 1, Vienna 1030, Austria, tecumseh.fitch@univie.ac.at)

The field of vertebrate bioacoustics has made great progress in understanding how animals produce vocalizations in the last two decades. Much of this progress has involved models originally developed in the context of human speech production, which were later applied to nonhuman animals. I will provide a broad semi-historical overview of this field, highlighting three prominent theories that originated in speech science and then were generalized to other species. My talk will focus on common features that are used by most vertebrates (including humans) to produce their communicative sounds, but also calls attention to numerous interesting exceptions to these general rules.

**3:45–4:00 Discussion**

## Session 2pMUb

## Musical Acoustics: VIMEO Livestream Concert: The Telematic Circlei—Celebrating Free Music performed live over the internet

Jonas Braasch, Cochair
*School of Architecture, Rensselaer Polytechnic Inst., School of Architecture, 110 8th St., Troy, NY 12180*

Chris Chafe, Cochair
*Music, Stanford Univ., CCRMA/Music, Stanford, CA 94305*

Margaret Schedel, Cochair
*Music/IACS, Stony Brook Univ., 1 wels lane, Setauket- East Setauket, NY 11733*

**7:00**

**2pMUb1. VIMEO Livestream Concert: The Telematic Circle—Celebrating Free Music performed live over the internet (https://vimeo.com/event/4551097/281fe79260).** Jonas Braasch (School of Architecture, Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, braasj@rpi.edu), Chris Chafe (Music, Stanford Univ., Stanford, CA), and Margaret Schedel (Music Dept. and Inst. of Advanced Computational Science, Stony Brook Univ., Setauket- East Setauket, NY)

In the spirit of a virtual meeting, the Technical Committee Musical Acoustics hosts a telematic concert featuring freely improvised music in the classic and jazz avant-garde tradition. In 2007, we started a Telematic Circle between Rensselaer Polytechnic Institute and Stanford University, performing free music weekly between two music seminars led by Pauline Oliveros at RPI and Chris Chafe at Stanford. This project was possible through a new low-latency INET2 internet network, Chafe's low-latency audio software, Jacktrip, and the adequacy of free music for such collaborations. Over the years, the Telematic Circle expanded to other institutions in the US and other countries. This concert will highlight a three-way connection between Rensselaer Polytechnic Institute, Stanford University, and Suny Stonybrook, featuring local musicians from all three institutions. The concert can be viewed via a VIMEO Livestream (https://vimeo.com/event/4551097/281fe79260).

## Session 3aAA

## Architectural Acoustics, Noise and Structural Acoustics and Vibration: The Unknown Unknowns: A Deep Look at Uncertainty and Precision in Architectural Acoustics

Evelyn Way, Cochair
*Maxxon Corp., 920 Hamel Rd., Hamel, MN 55340*

Michael Raley, Cochair
*PAC Int., 2000 4th Ave., Canby, OR 97013*

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

### *Invited Papers*

**10:00**

**3aAA1. Towards better understanding and communication of uncertainty in architectural acoustics and noise control.** Lily M. Wang (Durham School of Arch. Eng. and Construction, Univ. of Nebraska - Lincoln, PKI 100C, 1110 S. 67th St., Omaha, NE 68182-0816, lwang4@unl.edu)

The fields of architectural acoustics and noise control involve conducting measurements, running analyses, modeling, and making predictions about the behavior of sound in the built environment. Such tasks inherently involve uncertain aspects that affect the reliability of results. This presentation presents a general philosophy around uncertainty with suggestions on how to acknowledge and approach uncertainty, in an effort to move those involved in architectural acoustics and noise control toward better understanding and communication of uncertainty. Terminology, concepts, and pertinent standards that address uncertainty are also reviewed.

**10:15**

**3aAA2. The results and uses of a New Interlaboratory Study for ASTM E90 for frequently constructed wall types.** Evelyn Way (Maxxon Corp., 920 Hamel Rd., Hamel, MN 55340, eway@maxxon.com) and Michael Raley (PAC Int., Canby, OR)

How does one compare test reports from two different labs? How do we improve our standard test methods? How does a lab prove they get the same results over time? Every ASTM standard method is required to include a precision and bias statement. The precision statement is based on interlaboratory studies where multiple labs do repeated tests on the same sample. That data was only relevant when it is similar to the type of sample usually tested. The last interlaboratory study to establish the precision for ASTM E90 was completed over 20 years ago on low-STC steel panels. This presentation covers the initial results of a new study that included over 80% of North American accredited labs using higher-performing, commonly used, steel stud wall assemblies. The authors will present the uncertainty determined by the study along with sources of variation between the labs including opening size, microphone type, and room volumes. The presentation will also discuss how consultants can use this information to better evaluate laboratory test data, and the industry as a whole can use it to improve the reliability of acoustic testing.

**10:30**

**3aAA3. Uncertainty in room acoustics modeling.** Bruce Olson (Ahnert Feistel Media Group, 8717 Humboldt Ave., N, Brooklyn Park, MN 55444-1320, bcolson@afmg.eu) and Ana M. Jaramillo (Ahnert Feistel Media Group, Brooklyn Park, MN)

Room acoustics prediction software is a great tool for the acoustician. However, there are limitations on how reliable the predictions are and how close they can get to measured results. The first limitation of ray tracing modeling tools is the uncertainty of input data. We will also discuss uncertainty in the geometrical modeling of rooms, calculation settings, and accuracy of outputs.

**10:45**

**3aAA4. Uncertainty in heating ventilation and air condition equipment sound.** Derrick P. Knight (Trane Technolo., 824 Olympic Dr., Onalaska, WI 54650, derrick.knight@irco.com)

Uncertainty in heating ventilation and air condition equipment sound is formally defined in each test standard. Results from an industry sponsored round robin study have been disseminated in recent years. However, the uncertainty of a specific measurement is only part of the story. Manufacturers predict sound levels of more product variations than they will ever build, let alone measure. This presentation will discuss how one manufacturer deals with uncertainty in sound power data which is intended to be used by customers to design quiet buildings.

**11:00**

**3aAA5. Mechanical noise design—Discrepancies between calculations and outcomes in the field.** Daniel Choi (Newcomb & Boyd, 303 Peachtree Center Ave. NE, Ste. 525, Atlanta, GA 30303, dchoi@newcomb-boyd.com)

HVAC noise calculations that acoustical consultants typically perform utilize software that are mostly based on conceptual ASHRAE equations. Field measurements often indicate that the measured levels are sometimes close to those that are calculated, while at other times the discrepancies are greater. While conservative measures are taken to account for the discrepancies, this can potentially lead to overdesign, incurring additional costs that may not have been necessary. While considering every possible field factor is difficult, this study explores the identification of a few uncertainties that exist during design and field implementation.

**11:15**

**3aAA6. Acoustical design and uncertainty: Can we beat the MacLeamy curve.** Brandon Cudequest (Threshold Acoust., 141 W Jackson Blvd. Ste. 2080, Chicago, IL 60604, bcudequest@thresholdacoustics.com)

The MacLeamy curve is a useful framework for identifying design milestones and counteracting the cost of uncertainty. If uncertainty is inherent in aspects of design, then the MacLeamy curve suggests that risk can be minimized by identifying these uncertainties early on in the project. Even if the details are not fully formed, a contingency budget can be established until more accurate costs are known. Despite best efforts, there are moments in construction where situations become uncertain. This presentation will discuss several acoustical recommendations in the context of the MacLeamy curve and how designers can embrace the uncertainty. The discussion will be framed around a recently completed project in northern Minnesota. Its remote location made quantifying site noise a challenge; however, its proximity to a freight train required close understanding of the environmental noise impact. Additionally, the project had a number of sound isolation assemblies where no data exists: precast enclosure, double pane exterior windows, and a sliding barn door. Finally, the project auditorium had several variable elements: retractable seating, acoustic curtains, and separate theatrical curtains. The presentation will discuss how the room acoustics were modeled for these various configurations. Uncertainty is unavoidable, but there are ways to minimize its impact.

**11:30–12:00 Discussion**

# Session 3aAB

## Animal Bioacoustics and Student Council: Students and Early Career Animal Bioacoustics Talks

Marissa Garcia, Cochair
*Natural Resources and the Environment, Cornell Univ., 159 Sapsucker Woods Rd., Cornell Lab of Ornithology - K. Lisa Yang Ctr. for Conservation Bioacoustics, Ithaca, NY 14850*

Laura Kloepper, Cochair
*Dept. of Biol. Sci., Univ. of New Hampshire, 230 Spaulding Hall, Durham, NH 03824*

Xavier Mouy, Cochair
*Passive Acoust. Res. Group, NOAA, 3377 SW 28th Terrace, Miami, FL 33133*

**Chair's Introduction—10:00**

### *Contributed Papers*

**10:05**

**3aAB1. Unsupervised detection and classification of fish vocalizations in coral reefs.** Daniel Duane (NUWC Newport, 1176 Howell St., Newport, RI 02841, daniel.m.duane.civ@us.navy.mil), Nicholas Kroeger (Univ. of Florida, Gainesville, FL), Simon Freeman (Dept. of Energy, ARPA-E, Washington, D.C.), and Lauren Freeman (NUWC Newport, Newport, RI)

Typical convolutional neural network detectors for acoustic signals rely on large quantities of labeled data, which can be expensive and time-intensive to generate. Unsupervised neural network architectures allow for the automatic detection and classification of acoustic signals, which may be especially useful for classifying underwater signals that a typical human labeler may not be familiar with. Here, we used a convolutional autoencoder to automatically detect and classify fish vocalizations in a coral reef off the coast of Hawaii Island. A database of more than 2 million spectrogram segments were generated with durations of one second and frequency ranges of 30–700 Hz, corresponding to the timescale and spectrum of a significant portion of fish calls. The autoencoder was successful in recreating the features of input spectrograms, and k-means clustering was applied to the latent feature space of each sample to automatically generate classes. Visual inspection of example spectrograms within each class confirms that the autoencoder was successful in classifying fish calls, including differentiating between separate call types.

**10:16**

**3aAB2. Experimental sound exposure studies on aquatic animals: An early attempt to develop underwater bioacoustics in Iran.** Saeed Shafiei Sabet (Fisheries Dept., Faculty of Natural Resources, Univ. of Guilan, Rasht 1144, Iran (the Islamic Republic of), s.shafiei.sabet@guilan.ac.ir)

Sound-generating human activities have elevated ambient sound levels in aquatic habitats. Many aquatic animals also live in captivity for several purposes. Likewise, in aquatic habitats, there are also anthropogenic sound sources in captivity and under laboratory conditions. There is growing evidence that anthropogenic sound can have several impacts on aquatic life. In this presentation, behavioral studies exploring anthropogenic sound effects on aquatic animals that have been carried out in Iran will be reviewed (an invertebrate, the red cherry shrimp, and a vertebrate, the zebrafish). In a laboratory-based experiment, it was shown that generated sound exposure caused negative effects on sound-related spatial distribution and feeding performance in the red cherry shrimp. Zebrafish changed their swimming speed and foraging activities when exposed to sound. To increase biological relevance and behavioral interpretation, it is important to consider addressing sound components, both pressure level and particle motion, as well as designing experiments in a natural environment and in real-world conditions. Finally, limitations in underwater bioacoustics studies in Iran will be addressed.

**10:27**

**3aAB3. Insights into the demographics and spatial distribution of sperm whales at Guadalupe Island, Mexico, from passive acoustics.** William A. Deans (Scripps Inst. of Oceanogr., Univ. of California San Diego, 8635 Kennel Way, La Jolla, CA 92037, wadeans@ucsd.edu), Natalie Posdaljian, Kait Frasier, Jennifer S. Trickey (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA), Gustavo Cárdenas-Hinojosa (Comisión Nacional de Áreas Naturales Protegidas, Ensenada, Mexico), and Simone Baumann-Pickering (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA)

Sperm whales (*Physeter macrocephalus*) are cosmopolitan cetaceans exhibiting sex-based differences in ecology and distribution. Sperm whales are present at Guadalupe Island (GI), Mexico, where tourism vessels employed ultrasonic antifouling (UA) devices through 2021. We investigated sperm whale presence and demographics near the island and examined their spatial distribution using signal amplitude distributions from a single-hydrophone system. Sperm whales were detected intermittently at GI between 2018 and 2022. We used inter-click intervals to determine demographic information. Male sperm whales were present year-round; social groups of females and young were generally scarce but increased in 2022. Using Monte Carlo simulations, we evaluated different spatial distributions of each demographic group to identify the typical locations that best explained observed click received levels at the recording station. On daily and weekly scales, no significant change in sperm whale acoustic presence was observed for periods with UA, contrary to findings for goose-beaked whales (*Ziphius cavirostris*). However, because the predicted spatial distributions of sperm whale habitat indicated minimal overlap with the reach of UA, it appears that their distribution may limit their exposure. This study highlights GI as an important habitat for male sperm whales and, potentially, an emerging seasonal habitat for social groups.

**10:38**

**3aAB4. Terrestrial hearing in bearded seals (*Erignathus barbatus*).**
Noah Packard (Dept. of Ocean Sci., Univ. of California Santa Cruz, 115 McAllister Way, Sant Cruz, CA 95060, nbpackar@ucsc.edu), Jillian M. Sills (Inst. of Mar. Sci., Long Mar. Lab., Univ. of California Santa Cruz, Santa Cruz, CA), Ryan A. Jones (Dept. of Ocean Sci., Univ. of California Santa Cruz, Santa Cruz, CA), and Colleen Reichmuth (Inst. of Mar. Sci., Long Mar. Lab., Univ. of California Santa Cruz, Santa Cruz, CA)

As the Arctic becomes more accessible to human activities—and thus to anthropogenic noise sources—it becomes increasingly important to understand the auditory sensitivity of sound-reliant marine mammals, such as bearded seals (*Erignathus barbatus*). To address knowledge gaps, the in-air hearing of one adult bearded seal was measured outdoors in ambient noise conditions using psychophysical methods. 50% detection thresholds were measured for 10 frequencies (0.04–51.2 kHz) that extended across the subject's hearing range. For low to mid frequencies (0.04–12.8 kHz), thresholds were limited, or masked, by ambient noise. Thresholds obtained at higher frequencies (25.6–51.2 kHz) were sufficiently elevated above background noise to provide absolute measures of hearing sensitivity. These data reveal that the in-air, high-frequency roll off for bearded seals is in alignment with available data for related species despite more than 12 million years of evolutionary isolation. Furthermore, the masked data collected at low and mid frequencies enable an unconventional approach for estimating critical ratios, a key parameter in masking models. These findings highlight auditory similarities within the northern clade of phocid seals and improve our ability to predict potential noise effects for these species in a rapidly changing soundscape.

**10:49**

**3aAB5. Hawaiian monk seal terrestrial communication range estimates.**
Brandi Ruscher (Dept. of Ocean Sci., Univ. of California Santa Cruz, 115 McAllister Way, Santa Cruz, CA 95060, bruscher@ucsc.edu), Kirby Parnell (Mar. Mammal Res. Program, Hawaii Inst. of Mar. Biol., Univ. of Hawaii Mānoa, Kaneohe, HI), Jillian M. Sills, and Colleen Reichmuth (Inst. of Mar. Sci., Long Mar. Lab., Univ. of California Santa Cruz, Santa Cruz, CA)

The acoustic biology of Hawaiian monk seals (*Neomonachus schauinslandi*) is compelling due to their evolutionary and biogeographical separation from related species, prolonged breeding season, and endangered status. Recent auditory measurements suggest that terrestrial communication is constrained by poor hearing, but limited available information about vocal behavior precludes communication range estimates. In this study, we recorded spontaneous vocalizations from free-ranging monk seals at Kalaupapa National Historical Park (Molokai, HI) to describe spectral characteristics and provide the first source levels of the in-air, low-frequency calls emitted by this species. Sound pressure levels measured over the 90% call duration recorded 4–10 m from resting seals were used to approximate source levels at 1 m. Amplitude and spectral characteristics of recorded vocalizations were combined with monk seal hearing data and representative ambient noise levels to estimate the distances over which seals can effectively communicate with conspecifics in air. Preliminary estimates suggest that high hearing thresholds and low to moderate vocal amplitudes result in restricted communication ranges $< 500$ m. Understanding how terrestrial communication is influenced by noise in the environment will support decision-making about anthropogenic noise exposures that may be harmful to Hawaiian monk seals. [Research facilitated by NOAA's Hawaiian Monk Seal Research Program.]

**11:00**

**3aAB6. Using bone conduction brainstem auditory evoked response tests in gray seals (*Halichoerus grypus*).**
Amanda M. Moore (Dept. of Commun. Sci. & Disorders, Univ. of Cincinnati, 3225 Eden Ave., Cincinnati, OH 45267, moore4a4@mail.uc.edu), Gordon Hastie, Simon Moss, Ryan Milne (Sea Mammal Research Unit, Scottish Oceans Inst., St. Andrews, Scotland, United Kingdom), and Peter Scheifele (Dept. of Commun. Sci. & Disorders, Univ. of Cincinnati, Cincinnati, OH)

Gray seals (*Halichoerus grypus*) outer ears close when sedated, requiring appropriate methods to test hearing, circumventing the outer and middle ear (air conduction). Anthropogenic noise exposure (e.g., offshore windfarms) in the marine environment is a risk to seals, requiring appropriate methods for testing hearing. Bone Conduction Brainstem Auditory Evoked Response (BC-BAER) was investigated in six gray seals to characterize responses elicited by different acoustic stimuli, evaluate the potential of using BC-BAER to estimate hearing sensitivity, and best placement of the bone transducer. Seals were tested in December 2023 and February 2024 during ongoing research at the Sea Mammal Research Unit, University of St. Andrews, Scotland. BC-BAER measurements in six seals using broadband click and gated tone bursts stimuli had responses from 250 to 8000 Hz, the limits of the RadioEar-B71 bone transducer. Visual identification of Wave V was used to estimate hearing thresholds. Average threshold values were 45.67 dB re 20 μPa for tone bursts at 8000 Hz, the maximum sensitivity of frequencies tested. Recording of BC-BAER is a viable means of studying hearing in gray seals with the best placement of the RadioEar-B71 bone transducer being the zygomatic process compared to the mastoid.

**11:11–11:16 Break**

**11:16**

**3aAB7. Application of the oddball paradigm for investigating predictive coding in the bat auditory system.**
Victoria Lightfoot (Biology, Texas A&M Univ., 400 Bizzell St., College Station, TX 77843, vlightfoot@tamu.edu) and Michael Smotherman (Biology, Texas A&M Univ., College Station, TX)

Predictive coding is a theoretical framework for explaining how the auditory system efficiently codes the auditory scene by focusing on unexpected deviations. It is hypothesized that predictive coding plays a prominent role in the auditory processing of biosonar echo streams. At least two complementary biological mechanisms contribute to predictive coding: stimulus-specific adaptation (SSA), a reduced response to repetitive stimuli that emerges early in the ascending auditory pathway, and mismatch negativity (MMN), an enhanced response to low-probability stimuli mediated by the descending cortical feedback. Here, we utilized the oddball paradigm, an experimental design that measures the brain's sensitivity to rare (deviant) stimuli, to detect and measure SSA and MMN in the auditory brainstem response (ABR) of anesthetized free-tailed bats. We predicted that SSA would be detectable in the early ABR peaks, and that MMN, if present, should appear in the slower P0 waves reflecting cortical influences on auditory expectations. Stimuli consisted of paired tones, FM sweeps, and other naturalistic stimuli, and we tested the effects of varying stimulus probabilities from 50/50 to 90/10. Results show that both SSA and MMN were detectable and separable even in early peaks of the ABR waveform.

**11:27**

**3aAB8. Can light detection and ranging predict acoustic detection distance in heterogeneous forest environments?**
Lucas Voirin (Sciences du bois et de la forêt, Univ. Laval, 2405 Rue de la Terrasse, Québec, Quebec G1V 0A6, Canada, lucas.voirin.1@ulaval.ca), Marcel Lefebvre (Sciences du bois et de la forêt, Université Laval, Québec, Quebec, Canada), Jean-Philippe Migneron (Ecole d'architecture, Univ. Laval, Quebec City, Quebec, Canada), André Desrochers, and Marc J. Mazerolle (Sciences du bois et de la forêt, Univ. Laval, Québec, Quebec, Canada)

The estimation of the distance at which an animal can be detected is important information in wildlife monitoring. To account for the imperfect detection of individuals, frameworks like distance sampling or spatial capture-recapture build a detection function based on the distance between the source and the receiver. In acoustic monitoring, this detection distance is influenced by sound attenuation. In forest environments, vegetation significantly contributes to sound attenuation, and its effect varies with vegetation structure. Light Detection and Ranging (LiDAR) data provide fine-scale information on both horizontal and vertical structure of vegetation. These data are becoming increasingly available for large areas through public online geographic information libraries. In this study, we conducted broadcasting tests along linear transects to model the sound attenuation due to vegetation as a function of frequency and distance. The objective of this work is to estimate the attenuation coefficient based on fine-scale vegetation data to predict detection distance in an array of recorders.

11:38

**3aAB9. Ultrasonic vocalizations produced by a mouse model of autism spectrum disorder differ following social experience.** Sevda Abdavinejad (Univ. at Buffalo, Suny, 206 Park Hall, Buffalo, NY 14260, sevdaabd@buffalo.edu), Payton Charlton, and Micheal Dent (Univ. at Buffalo, SUNY, Buffalo, NY)

Mice are highly effective models for studying neurodevelopmental disorders characterized by abnormal social behaviors, such as Autism Spectrum Disorder (ASD). Because mice produce Ultrasonic Vocalizations (USVs) with many features like human speech, they have been proposed as a model for investigating neurodevelopmental disorders characterized by impairments in social communication, including ASD. The BTBR T+T+Itpr3tf/J mouse is an inbred strain displaying social abnormalities, communication deficits, and repetitive behaviors analogous to the behavioral symptoms of ASD. Here, we investigate the USVs produced by male and female BTBR mice in response to long- and short-term social experiences, including exposure to an opposite-sex mouse, a same-sex mouse, or no other mouse, for 1 h prior to the recording session. Male and female BTBR mice produce USVs which vary in features and proportions produced across social experiences compared to wild-type controls. These results add to the understanding of the BTBR mouse model for humans with ASD. Funded by NIH R01 DC016641.

11:49

**3aAB10. Hearing across the lifespan by mouse models of Alzheimer's disease.** Riley McLaughlin (Univ. at Buffalo, SUNY, 206 Park Hall, Buffalo, NY 14260, rileymcl@buffalo.edu), Payton Charlton, Sevda Abdavinejad (Univ. at Buffalo, SUNY, Buffalo, NY), Amanda M. Lauer (Otolaryngology-HNS, Johns Hopkins Univ. School of Med., Balitmore, MD), and Micheal Dent (Univ. at Buffalo, SUNY, Buffalo, NY)

Although hearing loss and dementias, such as Alzheimer's Disease (AD), are known to be closely linked, the exact nature of the link has not yet been determined. To better understand the association, we have been measuring hearing across the lifespan by several mouse models of AD. Using behavioral operant conditioning techniques, male and female mice are trained to nose poke for a chocolate milk reinforcer when they hear a 14 kHz pure tone. Using psychophysical signal detection theory, d-prime thresholds are calculated daily for each mouse subject from about 100 days of age through two years of age. Thresholds for several AD models and their wild-type counterparts generally increase across the lifespan as the mice get older. Although there is some variability across mouse subjects, thresholds from the AD models and wild-type controls are similar across the lifespan. Thus, although older humans with AD are more prone to hearing loss than those without AD, the current mouse models of AD do not recapitulate this phenotype. These results suggest an opportunity to investigate other factors contributing to the heterogeneity in hearing across the lifespan in mouse models of AD. [Work supported by NIH AG081747.]

THURSDAY MORNING, 21 NOVEMBER 2024 — 10:00 A.M. TO 12:00 NOON

## Session 3aAO

## Acoustical Oceanography: What's That Sound?

David R. Barclay, Chair
*Dept. of Oceanography, Dalhousie Univ., P.O. Box 15000, Halifax B3H 4R2, Canada*

**Chair's Introduction—10:00**

### *Contributed Papers*

10:05

**3aAO1. Bio-duck and bio-goose: Mysterious sounds from the Southern Oceans.** Ross Chapman (Univ. of Victoria, 3800 Finnerty Rd., Victoria, BC V8P5C2, Canada, chapman@uvic.ca)

This paper revisits recordings of curious sounds that were heard in waters around Australia and New Zealand some years ago. The sounds provided the New Zealand surveys were recorded on a low-frequency towed horizontal line array operating in the South Fiji Basin. They were thought to be generated by marine animals at distance from the array, but the type of creature was not identified at sea or in subsequent analysis. Owing to the limited bandwidth of the recordings, the sound was accordingly dubbed as Bio-Duck. Similar sounds recorded with higher bandwidth in Australian marine surveys were called Bio-Goose. Further characteristics of the sound waveform and spectrum are presented, along with evidence of a probable conversation between several speakers.

10:20

**3aAO2. Unidentified sounds from the endeavour hydrothermal vent field.** David R. Barclay (Dept. of Oceanography, Dalhousie Univ., P.O. Box 15000, Halifax, Nova Scotia B3H 4R2, Canada, dbarclay@dal.ca) and Brendan Smith (Dept. of Oceanography, Dalhousie Univ., Halifax, Nova Scotia, Canada)

Hydrothermal vent sites have the potential to produce transient physical, geophysical, geological, and biological sounds. A large number of sounds of unknown origin have been captured at the Main Endeavour Vent field in the North-East Pacific by an Ocean Networks Canada monitoring hydrophone. The hydrophone sits at a depth of 2,250 m, a few meters from a vent chimney.

**10:35**

**3aAO3. Listening to estuaries—Passive acoustic monitoring for environmental assessment.** Mohsen Badiey (Electr. and Comp. Eng., Univ. of Delaware, 139 The Green, Room 140, Evans Hall, Newark, DE 19716, badiey@udel.edu)

Estuaries are essential for environmental health, supporting diverse species of fish, birds, and other wildlife. They also play vital roles in the economy, recreation, and cultural heritage of their regions. The United States is home to ten major estuaries, each with unique and common features that can be effectively monitored using sound. In this talk, we present innovative methods for assessing ecosystem health by combining physical oceanography with acoustic monitoring in an estuary that serves as a major shipping passage in the northeastern United States. We explore how wind speed and direction affect sound propagation and scattering in this environment. Utilizing advanced techniques in acoustical oceanography, such as passive acoustic monitoring, active sonar systems, and underwater soundscape analysis, we present comprehensive data on various environmental parameters. Passive acoustic monitoring listens to natural and anthropogenic sounds, providing insights into marine life presence and behavior, human activities, and overall ecosystem health. Active sonar systems emit sound pulses and analyze echoes to map physical features and detect changes in water quality and sediment distribution. Underwater soundscape analysis integrates these methods to create a holistic view of the estuary's acoustic environment. Our research highlights the potential of these techniques to detect changes in estuarine ecosystems and suggests pathways for future research to enhance our understanding and protection of these crucial environments.

**10:50–12:00 Discussion**

THURSDAY MORNING, 21 NOVEMBER 2024 — 10:00 A.M. TO 12:00 NOON

## Session 3aBA

## Biomedical Acoustics: Biomedical Ultrasound Tutorials: (1) Hydrophone Measurement Methods for Biomedical Ultrasound and (2) Passive Cavitation Detection and Passive Acoustic Mapping Methods and Uses

Jeffrey A. Ketterling, Chair
*Radiology, Weill Cornell Med., Dept. of Radiol., 416 E 55th St., MR-008, New York, NY 10022*

### *Invited Papers*

**10:00**

**3aBA1. Basic hydrophone measurements for biomedical ultrasound.** Keith A. Wear (Ctr. for Devices and Radiol. Health, Food and Drug Administration, Bldg. 62 Room 2114, 10903 New Hampshire Ave., Silver Spring, MD 20993, keith.wear@fda.hhs.gov)

Acoustic pressure measurements with hydrophones are critical to characterize safety and effectiveness of diagnostic and therapeutic medical ultrasound devices. This tutorial will cover (1) fundamental interactions between ultrasound and biologic tissues that determine thresholds for safety and effectiveness, (2) underlying physics of various hydrophone designs (e.g., membrane, needle, capsule, reflection-type fiber optic, Fabry-Perot interferometric fiber optic), (3) fundamental hydrophone specifications, such as sensitivity, directivity, and effective sensitive element size, all of which are frequency-dependent, (4) pros and cons of each hydrophone design for different applications, (5) how to characterize hydrophone measurement uncertainty, and (6) deconvolution methods to remove distortions in pressure measurements due to frequency-dependent sensitivity and spatial averaging artifacts.

**10:55–11:05 Break**

**11:05**

**3aBA2. *Message in a bubble*: Methods and uses of passive cavitation detection and passive acoustic mapping.** Kevin J. Haworth (Dept. of Internal Med., Univ. of Cincinnati, Cincinnati, OH), Michael Gray (Inst. of Biomed. Eng., Univ. of Oxford, Oxford, United Kingdom), and Constantin Coussios (Inst. of Biomed. Eng., Univ. of Oxford, 54 Franklin Rd. Headington, Headington, Oxford, Oxfordshire OX3 7SA, United Kingdom, constantin.coussios@eng.ox.ac.uk)

An ever-growing body of clinical evidence demonstrates the profound impact of cavitation-based therapies on a wide variety of human pathologies. A key part of these evolving therapeutic techniques is the ability to detect, characterize, and quantify cavitation activity. In this session, we will describe the physical principles, devices, and algorithms for passive cavitation detection and mapping, with demonstrations and specific examples drawn from preclinical and clinical data. Finally, we will discuss the concept of cavitation dose and its distinct implementations for blood brain barrier opening, abdominal organ drug delivery, and histotripsy. The tutorial will include data and MATLAB code so that all attendees may play along.

# Session 3aCA

## Computational Acoustics: Innovations in Computational Acoustics I

Laura C. Brill, Cochair
*Threshold Acoust., 141 W Jackson Blvd., Ste. 2080, Chicago, IL 60604*

Ralph T. Muehleisen, Cochair
*Energy Syst., Argonne National Lab., 9700 S. Cass Ave., Bldg. 362, Lemont, IL 60439-4801*

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

**Chair's Introduction—10:00**

### *Contributed Papers*

**10:05**

**3aCA1. Simulating plane wave imaging with the fast nearfield method.** Jacob S. Honer (Elec. and Comput. Eng., Michigan State Univ., Dept. of Elec. and Comput. Eng., East Lansing, MI, honerja1@msu.edu) and Robert J. McGough (Elec. and Comput. Eng., Michigan State Univ., East Lansing, MI)

Conventional B-mode imaging suffers from limited frame rates due to compounded travel times. In contrast, plane wave imaging has emerged as a high-speed solution, offering superior frame rates while preserving resolution and contrast. Simulation plays a pivotal role in testing and refining new imaging algorithms, which is crucial for further development of plane wave imaging. However, limited support exists in current ultrasound simulation packages for plane wave imaging simulations. To address this deficiency, plane wave imaging support is incorporated into FOCUS, the "Fast Object-oriented C++ Simulator" (https://www.egr.msu.edu/~fultras-web/). This is achieved by adopting a pulse-echo imaging model that combines received signals from individual scatterers while assuming linear propagation of acoustic pressure fields within an isotropic, homogeneous, and non-dissipative medium. The fast nearfield method is deployed for the acoustic pressure field calculations, providing superior performance in terms of speed, accuracy, and memory-efficiency compared to other methods. The pulse-echo imaging model is extended to standard plane wave imaging techniques, such as electronic steering of plane waves and beamforming of the received RF data. Simulation results with a scatterer population that mimics a soft-tissue phantom featuring 5-point targets, 5 hyperechoic regions, and 5 anechoic regions of varying sizes will be demonstrated.

**10:20**

**3aCA2. Estimation of shear wave speed and shear viscosity with a simplified time-domain model for viscoelastic media.** Nicholas A. Bannon (Michigan State Univ., Dept. of Elec. and Comput. Eng., Michigan State Univ., East Lansing, MI 48824, bannonni@msu.edu), Matthew W. Urban (Dept. of Radiol., Mayo Clinic, Rochester, MN), and Robert J. McGough (Elec. and Comput. Eng., Michigan State Univ., East Lansing, MI)

Shear wave elasticity imaging is a non-invasive imaging modality that characterizes the mechanical properties of soft tissue, such as the shear wave speed or shear modulus. Present time-domain methods for shear wave parameter estimation divide the propagation distance by the propagation time to obtain an estimate of the shear wave speed. However, soft tissues are viscoelastic, and the shear viscosity frequently produces large errors in the estimated shear wave speed. To address this problem, a simplified time-domain model for shear wave parameter estimation in viscoelastic media is applied to simulated three-dimensional (3D) shear waves in a viscoelastic medium, where these 3D simulations require high performance computing to evaluate an approximate time-domain Green's function solution of Navier's equation that employs a Kelvin-Voigt rheological model. Parameters for the time-domain model are determined from an optimization procedure, then the shear wave speed is estimated from estimated time delays that account for the shear viscosity. Estimates of the shear wave speed are shown at the focal depth and in a two-dimensional region for various parameter combinations. The simplified time-domain viscoelastic model consistently yields smaller errors for the shear wave speed than cross-correlations while simultaneously providing estimates of the shear viscosity.

**10:35**

**3aCA3. Using high-performance parallel-in-time integration to model acoustics in expanding volumes.** Nathaniel W. Chapman (Comput. Sci., Central Washington Univ., 400 E. University Way, Ellensburg, WA 98926, nathaniel.chapman@cwu.edu) and Andrew A. Piacsek (Physics, Central Washington Univ., Ellensburg, WA)

Initial-value problems describing physical systems have long suffered from their inablility to be parallelized. The Parareal algorithm and its variations provide a means for parallel-in-time integration of these initial-value problems, leading to significantly decreased computation time. Further parallelization and performance improvements can be achieved using distributed networks of GPUs. Due to its potential complexities, the case of expanding volumes with a temporally constant speed of sound is considered to highlight these methods.

**10:50**

**3aCA4. Reinforcement learning applied to acoustic cloak design.** Zachary Knesek (Mech. Eng., San Jose State Univ., San Jose, CA), Vishwajeetsinh N. Varnamiya (Elec. Eng., San Jose State Univ., 1 Washington Sq., San Jose, CA 95112, vishwajeet.sinh31@gmail.com), Heet Mistry, Anthony Peters, and Feruza Amirkulova (Mech. Eng., San Jose State Univ., San Jose, CA)

This talk focuses on implementing deep reinforcement learning (RL) strategies to optimize an objective function of interest, improving our models developed earlier [1] to reduce the simulation time and the cost of design and to improve the efficiency of cloaking devices. Acoustic cloak design is achieved by minimizing the total scattering cross section (TSCS) for a non-uniform set of cylindrical scatterers, including rigid, elastic, and fluid scatterers. The optimized cloaked configurations are generated utilizing multiple scattering theory and deep RL algorithms. We extend and improve

the existing deep RL model [1] and design an acoustic cloak using the non-uniform configuration of scatterers by performing simultaneous scatterer position and radius adjustments. By automating the optimization process in a unified data-driven platform using MATLAB and Simulink, RL can significantly enhance the efficiency and effectiveness of discovering novel material configurations with desired properties. RL agent simultaneously adjusts design parameters, i.e., the position and radius of the scatterers, while taking into multiple scattering between the scatterers. Various deep RL algorithms, including deep Q-learning networks, are employed and compared to fmincon algorithms. Shah,T., Zhuo, L., Lai, P., Amirkulova, F., and Gerstoft, P., “Reinforcement learning applied to metamaterial design,” JASA 150(1), 321–338 (2021).

**11:05**

**3aCA5. A computational investigation of cavitation-induced traumatic brain injury as a result high-powered microwave exposure.** Eleanor Anderson-Zych (Mech. Eng., Univ. of Michigan, 2350 Hayward, Ann Arbor, MI 48109, eanderzy@umich.edu) and Eric Johnsen (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

Prior work by Dagro *et al*. (2021) has shown that high-powered microwave exposure can lead to rapid thermal expansion of brain tissue and has a strain-focusing effect. The rapid increase in thermal strain causes pressure waves to form in the brain and reverberate as tensile waves. This study investigates the possibility of tissue damage from these tensile waves through the growth of intrinsic cavitation nuclei. Cavitation is modeled using the Keller–Miksis equation modified to incorporate viscoelastic effects. A fourth-order variable time step Runge-Kutta scheme calculates bubble radii and wall velocities from the pressure profile. Our simulations show that the maximum hoop stretch caused by high-powered microwave exposure considered in Dagro *et al*. (2021) is less than the threshold needed to cause morphological degeneration in neural cells, as given in Estrada *et al*. (2021). By performing a parameter sweep, we find that pressures 1000 times larger than those reported by Dagro *et al*. (2021) are needed to cause injury. These correlations can help provide safety limits against brain injury due to microwave energy.

**11:20–12:00 Discussion**

THURSDAY MORNING, 21 NOVEMBER 2024 — 10:00 A.M. TO 12:00 NOON

## Session 3aED

## Education in Acoustics: Student 5-minute Elevator Pitch

Daniel A. Russell, Cochair
*Graduate Program in Acoust., Pennsylvania State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802*

Keeta Jones, Cochair
*Acoust. Soc. of Am., 1305 Walt Whitman Rd., Ste. 300, Melville, NY 11747-4300*

**Chair’s Introduction—10:00**

### *Contributed Papers*

**10:05**

**3aED1. Human listening and psychoacoustics skills through Ski-hill Graph Pedagogy Meter Fundamentals explored in music education.** Andrea Calilhanna (Faculty of Arts, Elder Conservatorium of Music, The Univ. of Adelaide, New South Wales, Sydney, New South Wales 2126, Australia, andrea.calilhanna@adelaide.edu.au)

Acoustics is central to music education, and listening should be ubiquitous in any pedagogical approach. Listening to music and learning meter theory is not always compatible with recent research. This paper provides insight into meter fundamentals education and the usual methods and offers a modern solution, Ski-hill Graph Pedagogy Meter Fundamentals. Cohn’s (2020) ski-hill graphs provide students with meter mathematics they need to apply to timing and expression of music to learn the basics of music for performance. The comprehensive contemporary meter theory offers teachers an alternative to meters understood as time signatures, one notated pulse grouped in a measure. Students require meter fundamentals pedagogy based on the sound that supports their developing awareness of the meter. Also, the meter theory requires the capacity for all meters, not just a surface level or one that focuses on Western music and notation. Ski-hill graphs offer valuable contributions to education, and the paper provides details of new ways forward in acoustics for listening in student education.

**10:10**

**3aED2. Bending beams: Effects of infill pattern and density on stiffness of 3D printed beams.** Joseph Kurek (none, 4282 Riverbirch Run, Zionsville, IN 46077, jkurek3.14@gmail.com), Nathan P. Geib, and Christina Naify (Appl. Res. Lab., The Univ. of Texas at Austin, Austin, TX)

Fused deposition modeling (FDM) is an additive manufacturing process that builds parts layer by layer, selectively depositing melted material in a predetermined path. Two of the principal 3D-print settings that can be modified with FDM are infill density and infill pattern. This project examined the impact of both of these settings on the stiffness of beams printed using

FDM. To obtain stiffness values, an accelerometer connected to an oscilloscope was first attached to beams before flicking them in order to determine their resonant frequencies. These data points, along with length, thickness, and density measurements, were then used to calculate each beam's stiffness using Euler–Bernoulli beam theory. An analysis of the results showed that both infill density and infill pattern had a significant impact on the stiffness of beams, suggesting that these print settings are important factors to keep in mind when designing parts to be 3D-printed using FDM.

## *Invited Paper*

**10:15**

**3aED3. Beyond audiograms: Simulating the complex reality of modern hearing loss.** Delbert Bray (none, 4700 Thronwood St., Portsmouth, VA 23703, dbrayjr5@gmail.com)

This presentation introduces a novel computational model designed to simulate both sensorineural and hidden hearing loss, providing a powerful tool for architects, acousticians, and researchers to better understand and address hearing disabilities. Developed using Max/MSP, the model employs a multiband downward expander to simulate sensorineural hearing loss, with thresholds adjustable based on age and common noise-induced patterns. Hidden hearing loss, an increasingly prevalent yet under-recognized condition, is modeled through the innovative manipulation of temporal envelope (ENV) and temporal fine structure (TFS) components. By allowing users to experience these hearing impairments firsthand, the model bridges the gap between audiological data and practical application in building design and acoustic engineering. This interdisciplinary approach not only enhances our understanding of hearing loss but also has potential implications for public health, incentivizing proper ear protection, and care. The model serves as a valuable educational resource, fostering empathy and driving more inclusive design practices in architecture and acoustics.

## *Contributed Paper*

**10:20**

**3aED4. Audible bandgaps in phononic sonic crystals for educational demonstration.** Nathan Scriba (Dept. of Mech. Eng., Univ. of Illinois Urbana-Champaign, 1206 W. Green St., MC 244, Urbana, IL 61801, nscriba2@illinois.edu), Elizabeth Smith (Mech. Sci. and Eng., Univ. of Illinois at Urbana Champaign, Urbana, IL), and Kathryn Matlack (Univ. of Illinois at Urbana-Champaign, Urbana, IL)

This experiment explored the use of phononic sonic crystals (PSCs) as an educational tool to demonstrate basic principles of wave propagation and phononics. Phononic crystals are periodic structures that attenuate a range of frequencies—called a bandgap—where the periodicity is on the order of the wavelength. Two PSCs were designed, constructed, and tested, to demonstrate the relationship between periodicity and bandgap frequency. The band structure of a 2D array of cylindrical rods was numerically constructed using eigenmode analysis in a finite element simulation. Based on the numerical simulations, two PSC spacings were selected for fabrication with bandgaps in the audible range of human hearing: a 3.5 cm spacing corresponding to a bandgap centered around 3200 Hz and a 6 cm spacing corresponding to a bandgap centered around 5600 Hz. A swept tone was transmitted through both PSCs and the resulting transmission curves were analyzed. The smaller spaced PSC was found to produce a bandgap from 5350z to 6000 Hz, with a 95% transmission drop; and the large PSC was found to produce a bandgap from 2800z to 3750 Hz with an 84% transmission drop. Both crystals were presented at the University of Illinois Urbana-Champaign's Engineering Open House in April 2024, and their attenuations in the bandgap frequencies were perceived audibly.

**10:25–11:00 Impromptu Contributions**

# Session 3aMU

## Musical Acoustics: Demonstrations of Measurement Techniques

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

Gary Scavone, Cochair
*Music Research, McGill Univ., 555 Sherbrooke St. West, Montreal H3A 1E3, Canada*

### *Invited Papers*

**10:00**

**3aMU1. Acoustic impedance measurements of wind instruments.** Gary Scavone (Music Res., McGill Univ., 555 Sherbrooke St. West, Montreal, Quebec H3A 1E3, Canada, gary.scavone@mcgill.ca), Champ C. Darabundit, and Miranda Jackson (Music Res., McGill Univ., Montreal, Quebec, Canada)

Measurements of the acoustic input impedance of wind instruments and their subparts can be useful for analyzing playing aspects, such as intonation, ease of response, and for comparison with numerical models. This talk will start with a discussion of commercial and custom measurement equipment options overviewing their particular features, calibration methods, advantages, disadvantages, and their associated costs. The remainder of the talk will focus on a custom-built, multi-microphone system developed in the Computational Acoustic Modeling Laboratory at McGill University, with a demonstration of its use in measuring one or two wind instruments. The authors are happy to share all hardware and software details of this system with anyone interested.

**10:20**

**3aMU2. An overview of the directivity measurement system at Brigham Young University with applications to struck percussion instruments.** Micah Shepherd (Brigham Young Univ., N249 ESC, Provo, UT 84602, mrs74@byu.edu), Hanna Pavill (Brigham Young Univ., Provo, UT), and Jacob Sampson (Brigham Young Univ., Orem, UT)

The directivity of musical instruments has long been a topic of interest for musicians and scientists alike. Over the past 15 years, researchers at Brigham Young University have engaged in high resolution directivity measurements of musical instruments using a variety of experimental setups. The current setup, a multiple capture system designed and built by now retired professor Timothy W. Leishman, features a 36 microphone arc which rotates on a turntable around the instrument of interest. This directivity measurement system (DMS) has been used most recently to measure the directivity of percussive instruments. The details of BYU's DMS will be outlined in this talk using a video tour format. Specific emphasis will be placed on measuring the directivity of struck percussion instruments. The details of an automatic striking device, which was developed to fit within the framework of a multiple capture DMS, will also be presented. The striking device is capable of holding multiple sized percussion mallets/sticks, making it a versatile tool for the study of struck percussion instruments.

**10:40**

**3aMU3. Low-cost electronic speckle pattern interferometry: Obtaining reliable signals.** Kurt R. Hoffman (Physics, Whitman College, 345 Boyer Ave., Hall of Science, Walla Walla, WA 99362, hoffman@whitman.edu), Caroline P. Peyton (Whitman College, Walla Walla, WA), and Andrew C. Morrison (Natural Sci., Joliet Junior College, Joliet, IL)

Designs for low cost electronic speckle pattern interferometry (ESPI) systems allow broad use of this technique to study the vibration or deformation of surfaces. For example, the technique allows direct imaging of the normal mode shapes of musical instruments, which can be used to assess design changes or optimize instrument fabrication. While the typical surface measurement system is quite simple in design, obtaining reliable signals is often challenging. Random phase changes due to object motion or reference signal light path can lead to decorrelations resulting in an unstable final image that washes out with averaging. Depolarization of the scattered light also diminishes the interference pattern intensity. We will be discussing the particulars of surface preparation, optical alignment, and phase control that help lead to successful ESPI image contrast and stability. We will also discuss the differences between using a stationary reference image for ESPI subtraction and a frame-by-frame difference method to obtain higher contrast interference patterns.

**11:00**

**3aMU4. Measuring body vibrations of stringed instruments.** Mark Rau (Music Res., McGill Univ., 550 Sherbrooke St. West, Ste. 500, Montreal, Quebec H3A1B9, Canada, mrau@ccrma.stanford.edu) and Gary Scavone (Music Res., McGill Univ., Montreal, Quebec, Canada)

Measurements of stringed instruments are used to interpret their vibrational characteristics, notably the input admittance and mode shapes. This talk will demonstrate methods to capture, process, and analyze stringed instrument measurements, focusing on surface vibrations. First, methods of suspending and isolating the instruments are discussed. Next, excitation methods, including the impact hammer and shaker, are demonstrated. Sensors will then be presented to measure the resulting surface vibrations, including accelerometers and laser Doppler vibrometers. Methods to view the mode shapes, including the roving hammer and scanning vibrometer measurements will be discussed. Finally, once the measurements are collected, techniques to post-process and analyze them with mode fitting will be discussed. This presentation will include video demonstrations of the previously mentioned measurement methods as applied to guitars, violins, and other stringed instruments. Best practices and the pros and cons of each method will be discussed.

**11:20**

**3aMU5. A rig for measuring violin sound radiation.** Joseph Curtin (Joseph Curtin Studios LLC, 3493 W. Delhi Rd., Ann Arbor, MI 48103, violins@josephcurtinstudios.com), Luca Jost (Tech. Univ. Dresden, Selm, Germany), and Chris Rogers (Tufts Univ., Medford, MA)

A now-standard way of measuring violin sound radiation is to tap the bridge with a miniature impact hammer while a microphone samples the resulting sound field. At least three factors complicate the process: (1) the strings exert three-dimensional forces at the bridge; (2) radiation becomes increasingly directional above 1 kHz; and (3) unless measurements are done under anechoic conditions, the acoustics of the measurement space must be considered. This video describes a rig that uses a solenoid-driven hammer, two driving directions (one horizontal at the bass corner of the bridge; the other vertical at the top-center), and twelve equatorial microphone positions 20 cm from the central axis of the violin. The resulting 24 FRFs can be used to estimate total sound output per-unit-force at the bridge. Alternatively, impulse responses from individual measurements can be convolved with recordings made on an electric violin to create realistic emulations of a normally played violin for use in psycho-acoustic testing.

**11:40**

**3aMU6. Uncertainty and repeatability in frequency response measurements.** Andrew A. Piacsek (Physics, Central Washington Univ., 400 E. University Way, Ellensburg, WA 98926-7422, andy.piacsek@cwu.edu)

Frequency response measurements are commonly used to characterize modal behavior and predict tonal qualities of musical instruments. In the case of violin family instruments, the response can be measured as bridge admittance (sometimes called mobility), which is the frequency domain ratio of the velocity of one side of the bridge to an impulsive force applied at the opposite side. As with any measurement, it is important to characterize and minimize the uncertainty of an instrument's frequency response. This is especially true for studies that investigate subtle changes in the frequency response of a single instrument due to structural adjustments (e.g., bridge material or sound post location) or environmental conditions (e.g., humidity, age or playing time). This presentation will demonstrate a method for measuring violin bridge admittance using a laser vibrometer and small impact hammer, with a focus on how the measurement is sensitive to details of the experimental setup. Methods of quantifying the variability in frequency response will be discussed along with example applications.

## Session 3aNS

# Noise: Interventions in Soundscape

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

Bennett M. Brooks, Cochair
*Brooks Acoustics Corporation, 49 N. Federal Highway, #121, Pompano Beach, FL 33062*

Brigitte Schulte-Fortkamp, Cochair
*HEAD Genuit Foundation, Ebert Straße 30 a, Herzogenrath 52134, Germany*

### *Invited Papers*

**10:00**

**3aNS1. Soundscape in application.** Bennett M. Brooks (Brooks Acoust. Corp., 49 N. Federal Highway, #121, Pompano Beach, FL 33062, bbrooks@brooksacoustics.com), Brigitte Schulte-Fortkamp (Acoust., Tech. Univ. Berlin, Berlin, Germany), and David s. Woolworth (Roland, Woolworth & Associates, Oxford, MS)

Defined as the acoustic environment understood by people in context, soundscape is a complex interplay of natural and human-made elements that contribute to the acoustical and social identity of a place. Participation in the planning and modification of an acoustic environment requires an active and creative approach and calls for a targeted engagement with the proposed designs or reorganizations. The introduction of standardized procedures and techniques is necessary to overcome the lack of defined systematic approaches to identifying interventions. Utilizing the soundscape standard will make the decision process transparent and comprehensible and thus more acceptable for authorities, policy makers, and other stakeholders. Guidance on how to implement innovative soundscape interventions through the upcoming Part 4 of the ISO Standard series 12913 will additionally increase the interest in interventions and how to achieve the intended purposes. Moreover, the application of soundscape methods have to meet the three primary public concerns: health, safety, and public welfare. In every aspect, the soundscape method is a powerful tool for understanding and actualizing our acoustical environment. Effectively balancing acoustic measurements, architectural and urban planning, and input based on the expertise of local experts will lead to a new understanding for the application of soundscapes.

**10:20**

**3aNS2. The soundscape of the Grove: The acoustic environment of a campus centerpiece.** Bryce M. Barrett (Physics and Astronomy, Univ. of Mississippi, 108 Lewis Hall, University, MS 38677, bmbarret@go.olemiss.edu), David S. Woolworth (Roland, Woolworth & Associates, Oxford, MS), and Joel Mobley (Physics and Astronomy, Univ. of Mississippi, University, MS)

A university campus is a mixed-use community where a balance between its functional and aesthetic qualities is central to its mission. The visual component of the environment is often the main concern, while the sonic component may not be carefully considered. The objective of this work was to evaluate the soundscape within the principal outdoor spaces on the University of Mississippi campus. These two spaces, known as the Grove and the Circle, are focal points of the University grounds and are relied upon to instill a positive first impression for visitors and to provide a locale for campus and community activities. In this project, the soundscape of these two spaces was evaluated through soundwalk-based surveys and sound level measurements. Five soundwalk sessions were conducted under different environmental conditions. Variations in perceptions and sound levels across the area of interest were found, and significant noise issues were identified. This study has provided meaningful information regarding the acoustic environment of the outdoor focal points of the campus and identified noise issues that should be addressed. Progress on mitigation efforts will be discussed.

**10:40**

**3aNS3. Need for an inclusive approach in soundscape research.** Arezoo Talebzadeh (Ghent Univ., 126 Tech Lane Ghent Science Park, Ghent 9052, Belgium, arezoo.talebzadeh@ugent.be)

Soundscape studies have received excessive interest in recent years, with distinct attention to standardizing research methodology and design outcomes. Soundscape is a complex adaptive system in which sound sources interact and evolve in different contexts, and multiple agents continuously influence and adapt to each other and the environment. Since soundscape is defined as a sonic environment perceived by people in context, soundscape studies must include people with diverse needs. WHO estimates the number of people with disabilities worldwide to be 1.3 billion. As urban environments become more complex, the demand for inclusive soundscape studies and design has never been more critical. Any complex system, such as soundscape, involves leveraging such systems' dynamic, interconnected, and adaptive nature to create environments where all individuals can thrive. The ultimate goal is to recognize, respect, and design human-centred auditory environments for uniqueness and variability that promote health and well-being for all. This goal can be

achieved by using an open, transparent, and co-creative process with people who have diverse needs and perspectives. By focusing on diversity, interconnectedness, adaptation, emergence, and non-linearity, designers can create soundscapes that cater to the needs of all community members, enhancing their experience and well-being and ensuring that the soundscape is inclusive, resilient, and responsive to ongoing changes and feedback.

**11:00**

**3aNS4. Hybrid VR and physical kiosks for soundscape design in urban planning.** Andy W. Chung (MOIA, Macao, Macao MO, Macao, ac@smartcitymaker.com) and Wai Ming To (Macao Polytechnic Univ., Macao, Macao)

In response to the challenge of urban noise pollution, innovative noise mitigation measures, such as acoustic windows and balconies, have been developed to effectively reduce traffic noise while allowing natural ventilation. To engage stakeholders and demonstrate the effectiveness of these innovations, a physical kiosk equipped with an acoustic window was created, along with a complementary Virtual Reality (VR) experience. This setup aims to create a near-real perception of the sonic environment, allowing users to experience noise reduction capabilities along with other sensory effects. Through this case study, we explore the practical application of VR and physical models as powerful tools for soundscape interventions and stakeholder engagement. By showcasing the integration of these technologies in urban planning, we provide insight into their potential to shape the future of soundscape design and policy development, emphasizing their role in creating healthier and more pleasant urban environments.

**11:20**

**3aNS5. Applying systems thinking in soundscape research and practice.** Francesco Aletta (University College London, Central House, 14 Upper Woburn, London N19DD, United Kingdom, f.aletta@ucl.ac.uk), Koko Zhou, Tin Oberman, Andrew Mitchell, Irene Pluchinotta (Univ. College London, London, United Kingdom), Simone Torresin (Univ. of Trento, Trento, Italy), Gunnar Cerwén (Swedish Univ. of Agric. Sci., Alnarp, Sweden), and Jian Kang (Univ. College London, London, United Kingdom)

This paper explores the future of soundscape research and practice in urban planning, emphasizing applied expertise and experience using the soundscape method. Soundscape studies recognize environmental sounds as a resource that can promote positive health effects, such as stress reduction and improved cognitive performance. However, enhancing soundscapes in urban areas may occasionally lead to unintended consequences, potentially exacerbating social inequalities. This study employs a systems thinking approach, involving participatory modelling workshops with experts to create a causal loop diagram (CLD) illustrating the complex interconnections between soundscape quality and public health. Key themes identified include noise pollution, socio-economic dimensions, environmental justice, and biodiversity. The CLD reveals feedback mechanisms and intervention points, suggesting research pathways, such as ecoacoustics, psychoacoustics, prediction models, and big data applications. These pathways aim to mitigate noise pollution, enhance biodiversity, and balance individual and community needs in soundscape design. This approach seeks to promote equitable and sustainable urban environments, advancing the discourse on soundscape research and its applications by integrating soundscape quality into urban planning and policy. Further interdisciplinary research and stakeholder engagement are essential to refine these insights and translate them into effective policies and practices.

**11:40**

**3aNS6. Friendly sounds in Children's hospital.** Antti Ikonen (Dept. of Art and Media, Aalto Univ., Otaniementie 14, Espoo 02150, Finland, antti.ikonen@aalto.fi) and Outi Ampuja (Univ. of Helsinki, Järvenpää, Finland)

The New Children's Hospital in Helsinki, Finland, is equipped with a specially designed soothing soundscape. The goal is to improve the comfort of child patients and reduce the tension when visiting a hospital and waiting for treatment and examinations. The soundscape is a combination of nature sounds and musical elements that vary depending on the location. The subtle sonic ornaments reflect the fairytale-like visuals of the hospital premises, without forgetting research on how sounds, especially from nature, evoke thoughts and memories. The soundscape is generative, so it never repeats itself exactly the same. It can be heard in the hospital's waiting areas, selected corridors, parking garage, and elevators. The impacts of the soundscape were investigated with questionnaires for the hospital staff, the child patients, and their carers. The experiences of the outpatients were investigated with individual and group interviews. According to the research, for the children the soundscape increased comfort, reduced stress, and tension, and increased the feeling of care and safety, offering a channel to nurture an emotional connection to the outside world. The responses of the outpatients highlighted the relaxing effect of the sounds. The soundscape will be updated in 2024 after careful analysis of the feedback.

# Session 3aPA

## Physical Acoustics and Engineering Acoustics: Frugal Acoustics I

Luz D. Sotelo, Cochair
*Purdue Univ., 2550 Northwestern Ave., 1900D, West Lafayette, IN 47906*

Randall P. Williams, Cochair
*Ctr. for Indus. and Med. Ultrasound, Univ. of Washington, 1013 40th Ave. NE, Seattle, WA 98105*

**Chair's Introduction—10:00**

### *Invited Papers*

**10:05**

**3aPA1. Frugal infrasound: Low-cost, low-power, low-weight, and low-effort.** Jacob F. Anderson (Geoscience, Boise State Univ., 1910 University Dr., 1535 Geosciences, Boise, ID 83725-1535, jacobanderson152@boisestate.edu) and Jeffrey B. Johnson (Geoscience, Boise State Univ., Boise, ID)

It is possible to acquire high fidelity array infrasound data at low cost. Recording infrasound economically requires sensors and acquisition systems that are inexpensive to buy or build, and efficient to transport, power, deploy, and maintain. A variety of engineering tools and design practices can help scientists economize on cost and effort from design to data collection. The design and prototyping process is streamlined by clearly defining design priorities and intended applications, selecting user-friendly components, libraries, and platforms, and using free CAD software and low-volume manufacturers for printed circuit boards. Custom enclosures built with 3-D printing and off-the-shelf watertight boxes facilitate production of sensor packages at any production scale. The cost and effort of fieldwork can be reduced by designing for portability (to reduce cost of transport and the number of field assistants needed), ease of use (facilitating use by new field assistants), and low power (to make power systems simpler, more portable, and requiring fewer maintenance visits). We describe how all of these frugal practices were implemented in the Gem infrasound logger and how they helped support new infrasound practitioners and applications that would be impractical with other infrasound systems.

**10:25**

**3aPA2. Acoustical measurements enabled by low cost electronics and digital oscilloscopes.** David A. Brown (ECE/CIE, Univ. of Massachusetts Dartmouth, 151 Martine St., UMass Dartmouth, Fall River, MA 02723, dbAcousticsdb@gmail.com) and Corey Bachand (BTech Acoust. LLC, Fall River, MA)

The advent of low-cost impedance analyzers and digital oscilloscopes have enabled the realization of cost-effective measurement systems for research, development, calibrations, and classroom/laboratory demonstrations. Traditionally, we have relied on HP/Agilent 4194 impedance anyalzers for laboratory use, but these units are heavy and expensive, originally costing about $30,000. Over the years, we have acquired and repaired many used units from ebay but recently found an alternative. Digilent offers a PC-based impedance measurement ($30) as add on to the Discovery logic Analyzer and Waveforms software, costing less than $300 that is lightweight and ideal for field or classroom use. Digital PC-based oscilloscopes and data acquisitions systems (e.g., TiePie, Cleverscope, Digilent) have also enabled the development of comprehensive low-cost acoustic calibration systems. We have developed such a system based on the TiePie Digital Oscilloscope (DIFF 1000MS/s), PC based signal generation and acquisition systems, a power amplifier, a preamplifier, and a stepper motor. The transmit signal is from a TiePie HS5-540XM 5 (500 MHz, 1-channel generator, 2-channel acquisition). The calibration and data display are controlled throught a MATLAB (GUI). Measurements of Transmit Pressure Response per Volt (TR/V or TVR), Free Field Voltage Sensitivity (FFVS), and Beam Pattern measurements and dependence on environmental pressure and temperature will be presented.

**10:45**

**3aPA3. A decade of MacGyver acoustics.** Teresa J. Ryan (Dept. of Eng., East Carolina Univ., 248 Slay Hall, Greenville, NC 27858-4353, ryante@ecu.edu), Heath Faircloth, Matthew Stengrim, Jeff Foeller (Dept. of Eng., East Carolina Univ., Greenville, NC), Diego Turo, and Joseph Vignola (Mech. Eng., The Catholic Univ. of America, Washington, D.C.)

This work presents lessons learned while aspiring to measure long distance outdoor sound propagation in an initially resource-limited program. This recounts the use of many components secured from discount stores and online vendors which are not typically incorporated in scientific research. These include inexpensive condenser microphones, portable speakers, and audio recorders more commonly seen in karaoke venues than acoustics labs. The price of using such relatively inexpensive hardware is the sweat equity required for validation and verification of these items for the research use case, to include both calibration and stability issues. Furthermore, the successful use of this equipment required developing effective protocols for use in the littoral environment and all of the threats therein:

sand, salt water, and thunderstorms. This presentation will discuss the specific hardware and factors to consider when adapting devices for research applications.

**11:05**

**3aPA4. Preventing hot bearing derailments via wireless onboard condition monitoring.** Constantine Tarawneh (Mech. Eng., Univ. of Texas Rio Grande Valley, 1201 W. University Dr., Edinburg, TX 78539, constantine.tarawneh@utrgv.edu)

The 2023 train derailment that occurred in East Palestine, OH, brought attention to the limitations of the detectors currently used in the industry. Typically, the health of train bearings is monitored intermittently through wayside temperature detection systems that can be as far as 40 miles apart. Nonetheless, catastrophic bearing failure is often sudden and develops rapidly. Current wayside detection systems are reactive in nature and depend on significant temperature increases above ambient, which, when detected, train operators have little time to react before a derailment occurs, as it did in East Palestine, OH. Multiple comprehensive studies have shown that the temperature difference between healthy and faulty bearings is not statistically meaningful until the onset of catastrophic failure. Thus, temperature alone is an insufficient metric for health monitoring. Over the past two decades, we have demonstrated vibration-based solutions for wireless onboard condition monitoring of train components to address this problem. Early stages of bearing failure are reliably detected via vibrations and acoustics signatures, which can also be used to determine the severity and location of failure. This talk will describe our work to develop and further these new methods and technologies for the rail industry as well as the impact of this research program on the training of hundreds of engineers from underrepresented backgrounds.

**11:25**

**3aPA5. Knock-knock, Who's there? Fresco.** Joseph Vignola (Mech. Eng., The Catholic Univ. of America, 620 Michigan Ave., NE, Washington, D.C. 20640, vignola@cua.edu), Nicholas T. Gangemi, Amelia Vignola, Nicholas DeLucia, Diego Turo (Mech. Eng., The Catholic Univ. of America, Washington, D.C.), Teresa J. Ryan (Engineering, East Carolina Univ., Greenville, NC), Barbara Marchetti (eCampus Univ., Washington, D.C.), and Jason Davison (Civil and Environ. Eng., The Catholic Univ. of America, Washington, D.C.)

Fresco painting, in which an artist applies dry pigments to wet plaster, dates from antiquity. A common mode of failure occurs when the decorative plaster layers separate from the masonry substrate or lower structural plaster. Our research team has for many years used scanning laser Doppler vibrometry to identify regions of delamination of frescos in the US Capitol Building as part of a conservation effort. Historically, such regions have been mapped by experienced conservators who knock on the plaster with their knuckles or small mallets and make subjective assessments of the radiated sound to identify areas needing restoration work. These acoustic signatures carry substantial information about the nature of the structure that produces the sound. Modern signal processing techniques can shed light on the underlying mechanical behavior of the system from those recorded acoustic signatures. This research will enable the use of simple, free smartphone or PC apps, and simple algorithms to put the conservator's expertise at anyone's disposal.

## *Contributed Paper*

**11:45**

**3aPA6. A practical approach to build an ultrasonic immersion test setup.** Harshith Kumar Adepu (Mech. Eng., Purdue Univ., 500 Central Dr., B36, West Lafayette, IN 47906, adepu@purdue.edu), Justin Yoosung Kim, Jacey Birkenmeyer, Partha Pratim Pandit, and Luz D. Sotelo (Mech. Eng., Purdue Univ., West Lafayette, IN)

Ultrasonic evaluation enables nondestructive defect detection and material characterization *in situ* and *ex situ*. Contact ultrasound is used for field inspections due to its portability, although it lacks resolution. Immersion testing offers higher resolution: however, these systems are more expensive and tend to be bulky. The need to balance higher resolution with portability and accessibility presents a significant gap for field testing and academic research. Our project introduces an economical and practical immersion setup. We modified a commercial 3D printer kit by incorporating plexiglass into the aluminum rails and sealing it with silicone paste to ensure watertight integrity. The original servo motors from the printer were repurposed to maneuver a custom transducer holder for precise transducer movement control. This setup achieved a resolution of 500 μm in step size for pulse-echo data collection. Constructed for about $2000 in less than a couple of days, this system exemplifies frugal engineering by delivering ultrasonic evaluations at a fraction of the cost and complexity of conventional systems. This portable setup can serve as an additional step for on-field inspections, providing greater resolution. Beyond industrial applications, it holds potential for enhancing ultrasonic education and workforce development in NDE.

## Session 3aPP

# Psychological and Physiological Acoustics: Panel on Remote Testing

Z. Ellen Peng, Cochair
*Boys Town National Res. Hosp., 555 North 30th St., Omaha, NE 68131*

Erol J. Ozmeral, Cochair
*Commun. Sci. and Disorders, Univ. of South Florida, 4202 E. Fowler Ave., Tampa, FL 33620*

**Chair's Introduction—10:00**

## *Contributed Papers*

**10:05**

**3aPP1. Comparison of Oldenburg matrix test results in laboratory and a remote testing appliffication.** Thomas Schwarz (Dept. of Med. Phys. and Cluster of Excellence "Hearing4all," Univ. of Oldenburg, Oldenburg 26129, Germany, thomas.schwarz@uni-oldenburg.de), Birger Kollmeier, Samira Saak (Medical Physics and Cluster of Excellence "Hearing4all," Univ. of Oldenburg, Oldenburg, Germany), Tobias Bruns (Fraunhofer IDMT, Oldenburg, Lower Saxony, Germany), and Lena Schell-Majoor (Med. Phys. and Cluster of Excellence "Hearing4all," Univ. of Oldenburg, Oldenburg, Germany)

The relevance of remote testing in audiology and acoustics is increasing, particularly due to the need for large datasets and ecological validity. The Virtual Hearing Clinic (VHC) is being developed as a platform to facilitate such large-scale online studies in audiological and acoustic research. This study aims to investigate the impact of using individual, uncalibrated hardware in remote testing in an unknown testing environment on the results of the Oldenburg Matrix Test. To this end, the Oldenburg Matrix Test was conducted in a pilot study with 10 otologically normal-hearing participants in a randomized order, once in the lab and once using the participants' mobile devices and headphones via the VHC outside of the lab, with an additional retest for the VHC measurement. Preliminary results show a test–retest correlation of $r = 0.58$ for the two VHC measurements. There is a deterioration in speech intelligibility scores of 2.70 dB from lab to VHC measurements. This difference appears to be a fairly constant offset with a standard deviation of 1.10 dB, indicating good validity for the remote testing approach via the VHC. However, the participant pool needs to be expanded both in number and in the range of different hearing impairments.

**10:35**

**3aPP2. Remote characterization of auditory and cognitive processing.** Esteban Sebastian Lelo de Larrea-Mancera (Psychology, Northeastern Univ., 105-107 Forsyth St. #12, Boston, MA 02115, e.lelodelarrea@northeastern.edu), Aaron Seitz (Psychology, Northeastern Univ., Riverside, CA), Frederick J. Gallun (Oregon Hearing Research Ctr., Oregon Health and Science Univ., Portland, OR), G. Christopher Stecker (Center for Hearing Res., Boys Town National Research Hospital, Omaha, NE), William J. Bologna (Speech-Language Pathol. & Audiol., Towson Univ., Towson, MD), Eric C. Hoover, Katherine N. Menon (Dept. of Hearing and Speech Sci., Univ. of Maryland, College Park, MD), Audrey Carrillo (Psychology, Northeastern Univ., Boston, MA), and Tess Koerner (Oregon Hearing Res. Ctr., Oregon Health and Sci. Univ., Portland, OR)

Previous research has demonstrated that remote testing of suprathreshold auditory function using distributed technologies can produce results that closely match those obtained in laboratory settings with a specialized, calibrated equipment. This work has facilitated internet-based recruitment of participants and the validation of various behavioral measures that provide valuable insights into auditory function. These measures include aspects of auditory processing, self-reported hearing difficulties, and cognition. In this study, we used the portable psychophysics assessment tool (PART) along with online participant recruitment to test a wide variety of audition-relevant measures in those with ($n = 92$) and without ($n = 100$) self-reported hearing loss. We assessed each measure's ability to discriminate between these two groups, as well as their reliability and multi-variate relationships. We found that several measures of auditory processing effectively differentiated between the groups. All previously tested measures showed similar test–retest reliabilities with this online approach. Additionally, multi-variate relationships, including cognition and self-reported hearing, were strikingly consistent across groups. In sum, this work further validates remote testing as a reliable general methodology.

**11:05**

**3aPP3. Enhancing adult cochlear implant utilization with remote speech perception assessments.** Katelyn A. Berg, Aaron C. Moberly (Otolaryngology, Vanderbilt Univ. Med. Ctr., Nashville, TN), and Terrin N. Tamati (Otolaryngology, Vanderbilt Univ. Med. Ctr., 1608 Aschinger Blvd., Columbus, OH 43212, terrintamati@gmail.com)

Cochlear implants (CIs) enable sound awareness and improved speech perception for adults with moderate-to-profound sensorineural hearing loss. However, CI utilization among eligible adults remains low at just 2%–13% in the United States. Remote speech perception assessments may help increase adult CI utilization by helping to reduce barriers to implantation. Specifically, remote assessments may improve patient access by lessening the time and financial burden of in-person clinical visits, particularly for adults in rural or remote locations. Furthermore, remote assessments can enhance clinical efficiency by reducing the number of in-person audiology visits (typically, six visits in the first year and every 6–12 months after that), freeing up valuable clinic time for new patients. Our research group has developed a battery of robust speech perception tests intended for remote administration, including word and sentence recognition, nonword repetition, sentence comprehension, and indexical processing tasks (e.g., talker and accent perceptions). Our findings in experienced adult CI users (> 1 year of CI use) demonstrate that performance on these tasks is consistent with prior in-person testing and correlated with conventional clinical word and sentence recognition outcomes. This presentation will discuss these findings, work from other research groups, critical next steps, and clinical implications.

**11:35–12:00 Discussion**

# Session 3aSCa

## Speech Communication: Choose-Your-Own-Adventure — Speech Production

Matthew Faytak, Chair
*Linguistics, Univ. of Los Angeles, Los Angeles, CA 95153*

### *Contributed Papers*

**10:00**

**3aSCa1. Production of chittagonian geminates.** Firoz Ahmed (Linguistics, Univ. of Florida, Gainesville, FL 32608, firozahmed@ufl.edu) and Caroline Wiltshire (Linguistics, Univ. of Florida, Gainesville, FL)

This paper investigates the production of medial geminates in Chittagonian Bangla, a dialect of Bangla/Bengali, to determine the durational differences of the geminate-singleton contrast. We also aim to discover whether there is any interaction with voicing, place, and manner on singleton and geminate duration. Chittagonian Bangla is a distinct variety of Bangla, mainly spoken in Chittagong division in Bangladesh. In this study, a wordlist of 22 true medial singleton and geminate pair wordlist was prepared, selecting stops, nasals, and laterals from four places of articulation (bilabial, alveolar, retroflex, velar); the stops included voiced and voiceless pairs. Data were recorded from two Chittagonian native participants (age 27) who were PhD students at a southeastern university in the USA. Praat was used to calculate the speech duration, and R was used for the statistical analysis and plotting. The singleton-geminate pairs showed a statistically significant difference in terms of duration. Voicing is a statistically significant factor in determining the ratio between geminate and singleton pairs, while of the places, only retroflex was a statistically significant factor. We compare our results to standard Bangla (Islam, 2016) and other languages with geminates (Al-Deaibes & Jarrah, 2023).

**10:05**

**3aSCa2. Ultrasound observations of tongue root configuration in Japanese geminated /t/.** Maho Morimoto (Chuo Univ., Tokyo, Japan, maho.morimoto.jp@gmail.com), Ai Mizoguchi (Maebashi Inst. of Technol., Maebashi, Japan), Weiyu Li (Sophia Univer., Tokyo, Japan), and Takayuki Arai (Sophia Univ., Tokyo, Japan)

Previous ultrasound studies indicate that geminated stops in some languages are produced with a more advanced tongue position compared to singletons, due to fortition (Percival *et al*. 2018 and Alamri 2022). Possible cross-linguistic variations have also been reported (Percival *et al*. 2020 and Kwon & Ahn 2024). In Japanese, the view that there is articulatory strengthening for geminates is supported through electropalatography studies (Kawahara & Matsui 2017 and Kochetov & Kang 2017), but the tongue root configuration in geminate production has not been addressed. In this study, we integrate results from three comparable ultrasound experiments to examine the tongue root positions at the offset of the singleton and geminate /t/ closure in /$C^1$ata/∼/$C^1$atːa/ pairs produced by 18 native speakers of Japanese. $C^1$ and the lexical status of the stimuli varied: nonce-words /pata/∼/patːa/, /bata/∼/batːa/, and real words /hata/ (surname) ∼ /hatːa/ "crawl-PAST." Through a within-speaker comparison of tongue contour models, we observed an advanced tongue root in geminates for /pata/∼/patːa/ in all three speakers, but in five out of the 15 speakers for /bata/∼/batːa/ and /hata/∼/hatːa/. We discuss the possibility that tongue root advancement in Japanese geminates interacts with the need to control intraoral air pressure, which may be affected by the preceding consonant.

**10:10**

**3aSCa3. Acoustic evidence for consonant cluster organization across contexts.** Christopher A. Geissler (Eastern, Slavic, and German Studies, Boston College, 370 Temple St., New Haven, CT 06511, christopher.geissler@bc.edu), Anya Sytenkova, Piper Brown, and Arthur Viegas Eguia (Linguistics, Carleton College, Northfield, MN)

C-center organization, in which onset consonant clusters exhibit temporal coordination similar to singletons, is an important theme in speech research. As an alternative to expensive and difficult articulatory research methods, Durvasula *et al*. (2021) developed a technique for identifying C-center timing using acoustic data alone. We replicate the English results of that study and investigate additional contexts. Following Durvasula *et al*. (2021), we noted the acoustic midpoints of onset consonants, took the average of these midpoints as "acoustic C-center," and used the acoustic end of the vowel as an anchor. If C-center coordination were present, C-center-to-anchor intervals should, as compared to singleton onset-to-anchor intervals, be similar in magnitude but more stable. We recorded 18 English speakers producing /s/+nasal, stop+liquid, and /s/+stop+liquid clusters along with segmentally matched non-clusters (e.g., *splay, play, lay*). Diagnostics indicated C-center-like-timing in initial clusters across segment types, as well as in tautosyllabic medial clusters; the evidence of C-center timing was not found where clusters were split across a syllable boundary. These results are consistent with established models of syllable structure and support the use of acoustic diagnostics for gestural coordination.

**10:15**

**3aSCa4. Oral and nasal airflow in nasalized laryngeal consonants.** Matthew Faytak (Linguistics, UCLA, 3125 Campbell Hall Box 951543, Los Angeles, CA 90095, faytak@buffalo.edu), Jorge Rosés Labrada (Linguistics, Univ. of Alberta, Edmonton, Alberta, Canada), Lev Michael (Linguistics, Univ. of California, Berkeley, Berkeley, CA), Myriam Lapierre (Linguistics, Univ. of Washington, Seattle, WA), Tyler T. Schnoor (Linguistics, Univ. of Alberta, Edmonton, Alberta, Canada), Tianle Yang, Mariana Quintana Godoy (Linguistics, Univ. at Buffalo, Buffalo, NY), and Beatriz Apolinario (Univ. Tecnológica del Perú, Lima, Peru)

While it has long been known that laryngeal consonants do not block nasal harmony, it has been debated whether the velum actually remains lowered during the production of glottals in nasal harmony contexts. Using newly collected nasal and oral airflow data from two Amazonian languages, Maihɨ̃ki (Tukanoan, Peru) and Piaroa (Jodɨ̈-Sáliban, Venezuela and Colombia), we demonstrate that the laryngeals [h, ʔ] are, in fact, produced as nasalized [h̃,ʔ̃] in these languages when they occur in nasal harmony spans. We observe that nasal flow is greater for both intervocalic [h] and [ʔ] in a nasal harmony context than in a non-harmony (oral) context, suggesting a lowered velum for both segments conditioned by the nasal harmony span. The glottal stops examined lack complete closure, as indicated by their nonzero oral flow; we consider the implications of this for the relationship between glottal stricture and measured nasality. [Work supported by U.S. National Science Foundation Award #1918064.]

10:20

**3aSCa5. Exploring groupings of Taiwan Mandarin Vowels through ultrasound imaging and cluster analysis.** Yen-Chen Lu (Graduate Inst. of Linguistics, National Taiwan Univ., No. 1, Sec. 4, Roosevelt Rd., Da'an Dist., Taipei City 106319, Taiwan, r11142004@ntu.edu.tw) and Chenhao Chiu (Graduate Inst. of Linguistics, National Taiwan Univ., Taipei City, Taiwan)

Previous studies have not reached a consensus on the number of phonemic vowels in Mandarin, with proposed systems ranging from four to six vowels. Since traditional top-down approaches have not led to a unified conclusion, this study employs a novel method using cluster analysis on ultrasound lingual images of Taiwan Mandarin vowels, hoping this bottom-up approach offers new insights into the debate. A total of 2700 tokens of consonant-vowel (CV) and isolated vowel (V) syllables were recorded from a female native speaker of Taiwan Mandarin. Preprocessed ultrasound images from vowel midpoints were analyzed using K-means clustering. To determine the optimal number of clusters, the elbow method and KneeLocator from the kneed package were used. Although KneeLocator suggested four clusters as optimal, the elbow plot did not show a clear elbow point. This mirrors the inconclusive results of previous studies on the Mandarin vowel system. Similar to the elbow plot, the clustering visualization using Principal Component Analysis (PCA) and Uniform Manifold Approximation and Projection (UMAP) did not show distinct clustering patterns, indicating potential limitations of K-means clustering for our data. Future research may benefit from exploring alternative clustering algorithms and preprocessing techniques, such as noise reduction in ultrasound images. This preliminary study shows the potential of using cluster analysis on ultrasound images for phonetic research and highlights on the challenges in defining Mandarin vowel inventory.

10:25

**3aSCa6. Vocal fold vibration associated with inspiratory phonation—An acoustic and electroglottographic study.** Xiuli Song (Duquesne Univ., 600 Forbes Ave., Pittsburgh, PA 15282, songx@duq.edu), Manwa L. Ng (Speech and Hearing Sci., Univ. of Hong Kong, Pokfulam, Hong Kong), and Yang Chen (Duquesne Univ., Pittsburgh, PA)

Unlike normal phonation (EP), inspiratory phonation (IP) involves speech production using ingressive airflow. It is often used in clinical voice assessment and treatment. The present study examined the voice and glottal vibratory characteristics associated with IP by means of acoustic and electroglottographic (EGG) analyses. Acoustic and EGG signals of the vowel /i/ produced by 44 young vocally healthy adults (22 men and 22 women) using IP and EP were simultaneously recorded and later analyzed using Praat. EGG parameters included contact quotient (CQ), contact quotient perturbation (CQP), contact index (CI), relative contact rise time (RT), slope of the contacting phase ($Slope_{con}$), slope of the de-contacting phase ($Slope_{decon}$), amplitude, $\%Jitter_{EGG}$, $\%Shimmer_{EGG}$, and $HNR_{EGG}$. Acoustic parameters included $F0_{AC}$, $\%Jitter_{AC}$, $\%Shimmer_{AC}$, and $HNR_{AC}$. During IP, mean $F0_{EGG}$, $\%Jitter_{EGG}$, and $\%Shimmer_{EGG}$ increased, while $HNR_{EGG}$decreased. Mean $F0_{AC}$, $\%Jitter_{AC}$, and $\%Shimmer_{AC}$ increased during IP, with no significant difference between IP and EP for $HNR_{AC}$. CQP, CI, and RT increased, while $Slope_{con}$ and amplitude decreased during IP, with no significant difference between IP and EP for CQ and $Slope_{decon}$. Results indicated marked differences in voices and glottal vibration between IP and EP.

**10:30–10:50 Discussion**

THURSDAY MORNING, 21 NOVEMBER 2024 — 10:00 A.M. TO 12:00 NOON

## Session 3aSCb

## Speech Communication: Choose-Your-Own-Adventure — Speech Production and Technology

Maria Kondaurova, Chair
*Psychological and Brain Science, Univ. of Louisville, Louisville, KY 40292*

### *Contributed Papers*

10:00

**3aSCb1. Acoustic changes in consonants produced with a facemask.** Feiyun Jiang (Duquesne-China Health Inst., Duquesne, 600 Forbes Ave., Pittsburgh, PA 15282, jiangf@duq.edu), Yang Chen (Duquesne-China Health Inst., Duquesne, Pittsburgh, PA), and Manwa L. Ng (Speech and Hearing Sci., Univ. of Hong Kong, Pokfulam, Hong Kong)

The widespread use of surgical facemasks in the post COVID-19 era has raised concerns about their impact on speech communication. As many previous studies focused only on vowel production, the present study aims to explore how surgical mask affects production of consonants. Four stops (/$p^h$, p, $t^h$, t/) and three fricatives (/s, ɕ, f/) produced by 61 adults (31 males and 30 females) who were native speakers of Mandarin Chinese were obtained. Spectral moment and other acoustic measures including Centre of Gravity (CoG), Standard Deviation (SD), skewness, and kurtosis, and spectral peak, duration, and voice onset time (VOT) were analyzed by using Praat software. Preliminary findings indicated higher CoG values, lower SD, skewness, and kurtosis, and VOT when wearing a facemask. A repeated-measures ANOVA revealed significant effects of facemasks on acoustic features, suggesting that facemasks tended to alter the acoustic properties associated with Mandarin consonants, potentially reducing speech intelligibility.

10:05

**3aSCb2. The impact of telepractice on voice production in normal-hearing children.** Maria V. Kondaurova (Psychological & Brain Sci., Univ. of Louisville, 317 Life Sciences Bldg., Louisville, KY 40292, maria.kondaurova@louisville.edu), Alan F. Smith (Otolaryngology-Head/Neck Surg. and Commun. Disorders, Sect. of Speech-Language Pathol., Univ. of Louisville, Louisville, KY), Yesenia Rodriguez Cruz (Psychological & Brain Sci., Univ. of Louisville, Louisville, KY), Irina Kondaurova, and Qi Zheng (Bioinform. and Biostat., Univ. of Louisville, Louisville, KY)

Telepractice is increasingly being used to evaluate voice quality and speech signal processing in children with and without hearing loss. The study examined whether telepractice (referred to as tele) affects voice production in normal-hearing children. Ten children (mean age = 10.4 years, age range 8–12 years) participated in four (2 in-person, 2 tele) weekly visits, order counterbalanced. At each visit, the clinician asked children to read sentences, hold a standardized conversation, and provide subjective measures of vocal effort. Fundamental frequency (Hz) and smoothed cepstral peak prominence (CPPS) (dB) were significantly higher in the children's utterances during tele compared to in-person communication irrespective of the task type or visit number. For intensity (dB SPL), a significant interaction for session type (in-person, tele) and visit number (first, second) was found regardless of the task type. Thus, intensity was higher in tele compared to in-person conditions during the first in-person versus tele visits only. A positive relationship between intensity and CPPS was found. No effect of telepractice on subjective measures of vocal effort was identified. The results suggest that vocal characteristics in normal-hearing children were affected during remote compared to in-person communication. This may impact the telepractice delivery of therapeutic intervention in children.

10:10

**3aSCb3. Acoustic and neural representation of recognizing different pragmatic intentions from speech prosody in high-functioning ASD children.** Xiaoming Jiang (Inst. of Linguistics, Shanghai Int. Studies Univ., 1550 Wenxiang Rd., Shanghai 201620, China, xiaoming.jiang@shisu.edu.cn), Yi Li (Inst. of Linguistics, Shanghai Int. Studies Univ., Shanghai, China), Xiquan Ma (Shanghai Children's Med. Ctr., Shanghai Jiao Tong Univ. School of Med., Shanghai, China), Shuyi Zhang (McGill Univ., Montreal, Quebec, Canada), Yilin Zhao, Yuanhui Li, Jinyang Chen, Zhuyun Wu, and Yixuan Zhu (Inst. of Linguistics, Shanghai Intl. Studies Univ., Shanghai, China)

Children with autism spectrum disorder (ASD) have difficulties detecting others' intentions and attitudes from nonverbal cues in social communication. However, their ability to detect pragmatic functions from speech is less understood. This study involved high-functioning ASD children and typical controls listening to speech with different prosodies conveying attitudes towards discussion content (e.g., confident or desired speech) or the listener (e.g., dominant or friendly speech). Participants judged the attitudinal prosodies while their cortico-hemodynamic responses were monitored with functional near-infrared spectroscopy (fNIRS). Analysis showed enhanced cortical activity in the middle/superior temporal gyrus (MTG/STG) and dorsolateral prefrontal areas during content-oriented prosody, and in the dlPFC, inferior frontal gyrus, frontopolar area, and orbitofrontal regions during listener-oriented speech in ASD children compared to the typical development (TD) group. Behaviorally, ASD children showed higher accuracy than TD when hearing listener-oriented speech but similar performance for content-oriented speech. Representational similarity analysis (RSA) indicated that ASD children used both pitch and intensity cues for listener-oriented prosodies but only intensity for content-oriented prosody. This study highlights the altered neurobehavioral profiles of speech perception in children with ASD and underscores the importance of fNIRS in assessing pragmatic functions in high-functioning ASD children.

10:15

**3aSCb4. Predicting the consequence of hearing loss for recognition of vocally expressed emotion using modulation spectral features.** William Martens (Linguistics, National Acoust. Lab., Macquarie Univ., Level 4, Australian Hearing Hub, 16 University Ave., New South Wales 2109, Australia, bill.martens@mq.edu.au) and Erin M. Picou (Hearing and Speech Sci., Vanderbilt Univ. Med. Ctr., Nashville, TN)

Adults with hearing loss demonstrate difficulty recognizing vocal emotion expressed by others, and yet little is known about what stimulus parameters predict the difference in emotion classification performance between normal hearing (NH) listeners and hearing impaired (HI) listeners who use hearing aids. For a controlled test of sensitivity to prosodic cues enabling discrimination between six classes of vocally expressed emotion, a set of 40 short utterances was prepared that was counterbalanced for the effects of talker sex by including 20 pairs of utterances devoid of lexical cues to emotion, wherein a male and female version of each utterance was matched in prosodic pitch contours and modulation spectral features. Observed sensitivities on a six-alternative emotion-classification task differed significantly between a group of 32 NH and 30 HI listeners. Regression analysis of obtained sensitivity data revealed that the differences in classification performance between groups of NH and HI listeners could be predicted partially by their audiometric thresholds for tones at two low frequencies (250 and 500 Hz); however, a popular measure of modulation spectral features (MSFs) in the short speech stimuli also aided in predicting the difference in sensitivity to prosodic cues between groups. [Work supported by Sonova AG, Stäfa, Switzerland.]

10:20

**3aSCb5. Overcoming biases in state-of-the-art automatic speech recognition for young children with speech disorders.** Vishal Shrivastava (Commun. Sci. and Disorders, Northwestern Univ., Frances Searle Bldg., Room 2-331, 2240 Campus Dr., Evanston, IL 60208, shrivastava_vishal@outlook.com) and Marisha Speights (Commun. Sci. and Disorders, Northwestern Univ., Evanston, IL)

State-of-the-art models like Whisper and GPT-4o face significant challenges in recognizing and processing child speech, particularly disordered speech, due to the limited availability of annotated child speech data and inherent demographic biases. Our research aims to bridge this gap by adapting these models for more accurate classification and recognition of disordered vs. non-disordered child speech. Using the SEED corpus, we leveraged advanced data augmentation, transfer learning, and parameter-efficient fine-tuning, achieving significant Word Error Rate (WER) reductions: from 57.5% to 10.3% for disordered speech and from 33.6% to 6.04% for non-disordered speech. Our novel approach also addresses demographic biases, ensuring equitable ASR performance across diverse groups. By overcoming current methodological constraints and emphasizing fairness and precision, our research not only advances ASR capabilities but also establishes a benchmark for inclusive technological advancement. These developments are crucial for practical applications in speech therapy and education, showcasing our commitment to impactful research in this domain. Future plans include enhancing model robustness by incorporating self-supervised learning techniques and exploring domain adaptation to ensure adaptability across varied speech environments. This comprehensive approach represents a significant step forward in signal processing, providing robust solutions to longstanding challenges in child speech recognition.

10:25–10:50 Discussion

## Session 3aSCc

## Speech Communication: Choose-Your-Own-Adventure — Perceiving Speech

Jie Yang, Chair
*Communication Disorders, Texas State Univ., Round Rock, TX 78665*

### *Contributed Papers*

**10:00**

**3aSCc1. The role of language proficiency on the perception of speech and song.** Reem Idris (Psychology, Univ. of Toronto, Missisauga, 250 Webb Dr., Mississauga, Ontario L5B 3Z4, Canada, reem.idris@mail.utoronto.ca), Anna Czepiel, and Christina Vanden Bosch der Nederlanden (Psychology, Univ. of Toronto, Missisauga, Mississauga, Ontario, Canada)

During everyday communication, background noise can affect our ability to comprehend speech accurately. Although many studies have investigated perception of speech in noisy environments, it is less clear how our perception in noisy environments could be enhanced by song, which could be beneficial due to melodic/rhythmic predictability. Language proficiency may also impact perception of song and speech. The current study aimed to clarify the unique struggles that people with low language proficiency face. We used the Test of Adolescent and Adult Language - Fourth Edition (TOAL-4) to divide 48 participants into low and high-English proficiency groups. Participants' speech-in-noise performance was assessed across four blocks of sung and spoken sentences. As predicted, performance improved for both song and speech over the course of the experiment and was higher in high-proficiency participants than low-proficiency English speakers. However, contrary to our predictions, participants found it harder to comprehend song than speech. This did not interact with language proficiency (high versus low). Overall, our findings suggest that, regardless of language background, song does not benefit perception in noise, challenging previous research that highlight the benefit of melodies and rhythms in facilitating speech perception in noisy or difficult listening environments.

**10:05**

**3aSCc2. Individual differences in the production of Utterance-Final Intonational Prosodies in English.** Sishi Fei (Dept. of Linguistics, The Univ. of Hong Kong, 9.30, Run Run Shaw Tower, Pokfulam Rd., Hong Kong, Pokfulam 999077, Hong Kong, sishi_fei@163.com)

Speech prosody conveys meaning through features like duration, pitch, and intonation. Subtle changes in utterance-final intonations (e.g., *It's raining.* [flat intonation] as a statement to *It's raining?* [rising intonation] as a question) reflect different speaker intentions and impact communication. However, this sound-to-meaning mapping can vary substantially between speakers due to socio-cultural factors. This study investigates individual differences in producing utterance-final prosodies in English, focusing on gender and first language. We analyze utterances from native English speakers and Chinese English learners (an equal number of females and males in each group). Fundamental frequency (F0) and duration are extracted using a customized Praat script. Linear mixed-effects models are employed on these two acoustic correlates, with gender (male versus female) and first language (English versus Mandarin Chinese) as between-subject factors, and utterance type (statement versus question) as the within-subject factor. We anticipate significant effects of gender and first language, with female speakers exhibiting higher average F0 values and longer durations than male speakers. Native English speakers will show clearer distinctions in F0 and duration between statements and questions compared to Chinese learners. These findings can provide valuable insight into talker variability in speech acoustics.

**10:10**

**3aSCc3. Acoustic characteristics of English intonation in Mandarin-English bilinguals.** Jie Yang (Commun. Disorders, Texas State Univ., 200 Bobcat Way, Round Rock, TX 78665, thyjessie@utexas.edu)

In addition to segmental level contrast, Mandarin Chinese uses two levels of prosodic contrast, tone at word level and intonation at utterance level, to convey linguistic meaning. Mandarin speakers need to adjust fundamental frequency (f0) to maintain intelligibility at both word- and utterance-level. For English speakers, f0 changes at utterance level for intonation were not confined by word-level tonal requirements. The present study investigated the acoustic characteristics of English intonation produced by Mandarin-English bilinguals at different developmental stages. Six- and nine-year-old Mandarin-English bilingual children and adults (seven per group) completed the speech production tasks. Stimuli were three types of carrying sentences (statement, question with and without inversion) in English. Carrying sentences end with monosyllabic target words that are phonetically similar in Mandarin (e.g., [di], 帝 Dee). Average f0 and magnitude of f0 change were measured within word and over the entire utterance, and compared among age groups. Results indicated (1) average f0 was the highest in questions without inversion, followed by questions with inversion. Statements showed the lowest average f0. (2) Magnitude of f0 change was larger in speakers with more English exposure and in older children. Sentence type, language experience, and developmental age appear to influenced average f0 and magnitude of f0 change.

**10:15**

**3aSCc4. (Non)assimilation of consonant clusters in phonological conditioning: An acoustic study.** Gifty Osei-Bonsu (Applied Linguistics, Univ. of Education, Winneba - Ghana, Winneba, Winneba Box 25 Winneba C/R, Ghana, gosei-bonsu@uew.edu.gh) and Charlotte F. Lomotey (Applied Linguistics, Univ. of Education, Winneba - Ghana, Winneba, Ghana)

In phonological conditioning, phonemes become assimilated to match the properties of adjacent phonemes. At the syllable coda morpheme boundary in the English language, these morphemic elements express grammatical features of marking number (plural) and genitive *(singular and plural possession)* in regular nouns, and the 3$^{rd}$ person singular and past tense in regular verbs. These morphemes create consonant clusters that are expected to assimilate for easy production (or linguistic economy). Unfortunately, it has been revealed that non-native speakers do not always assimilate the consonants to reflect this conditioning. This study investigates the (non)assimilation of consonant clusters in phonological conditioning from an acoustic perspective. It uncovers the patterns of phonological conditioning that are derived from (non)assimilation of consonant clusters in Ghanaian English and the factors that account for such patterns using the Speech Learning Model (SLM) by Flege (1995) and the Markedness Differential Hypothesis by Ekman (1977). Six (6) hours of data recorded from 50 Ghanaians were subjected to auditory and acoustic analyses. Results indicate that Ghanaian speakers of English sometimes assimilate these phonological endings correctly but mostly deviate from inner circle expectations. Typically, while they assimilate the voiceless endings, the voiced endings most often pose a challenge to them. Keywords: assimilation, consonant clusters, phonological conditioning, morphophonemic orthography, markedness

10:20

**3aSCc5. Effect of phonetics training in the perception of the four-way stop contrast by native English-speaking learners of Hindi.** Sreeparna Sarkar (Univ. of Pennsylvania, 260 South Main St., Newark, DE 19711, sree@udel.edu)

The Hindi four-way stop contrast poses a challenge in speech production and perception for native English-speaking learners of Hindi. This study tests the perceptual accuracy of the four-way stop contrast by native speakers of English and if it improves if they are explicitly taught about the phonetic details of the stops including their articulatory mechanisms and the resulting acoustic signals. Data was collected from two sets of participants 1) learners who were provided the phonetics training (n=11, male=5, female= 6) and 2) learners who were not provided the phonetics training (n=12, male=6, female 6). Participants were native speakers of American English and learners of Hindi. Participants from both groups responded to 270 audio stimuli of CVs; the Cs were the different stop types and the V was /a/, e.g.: /ta/, /pa/, /$t^h$a/, /$g^h$a/. Participants had to select the CV they heard from a selection of 6 choices shown on their computer screen simultaneously. Results show a higher perceptual accuracy (70.37% accuracy) in the group that received the phonetics training compared to the one that did not (63.37% accuracy). This 7% difference was significant according to a linear mixed effects model (t value = 8.977, p value $<0.05$) suggesting a significant effect of phonetics training in the perception of the four-way stop contrast. An important future direction of this study is to test whether this perceptual accuracy is also reflected in speech production.

10:25

**3aSCc6. Representing Tone in Onomatopoeic Expressions.** Joey K. Zimmerman (none, N/A, Piedmont, CA N/A, jkzimmerman@berkeley.edu)

My research seeks to clarify what specific acoustic properties that humans use to represent sounds in onomatopoeia, in the specific domain of tone. A wealth of research (Tanaka *et al*. 1995, Yamauchi *et al*. 2003, Yamauchi and Iwamiya 2005, and Matsui 2020, etc.) has find that, when representing pure tones as onomatopoeia, native Japanese speakers will use F2 as a proxy for f0. This has also been found to be true of native Mandarin speakers (Matsui 2020). Yet, it has not been determined whether this is also true of speakers of a non-tonal langauge, like English, and whether this association is one sided. My ongoing study will be conducted on 30 native English speakers and 30 native Mandarin speakers. In the first part, which reuses the methodology of existing research, participants will be asked to listen to pure tones of 62.5, 125, 250, 500, 1000, 2000, 4000, and 8000 Hz and represent them as an onomatopoeia. In the second part, which will be administered only to Mandarin speakers, participants will undergo a 2-AFC task where they must categorize tokens from a tonal minimal pair. All of the tokens have been high-pass filtered to obscure tone and have F2 altered to be higher or lower than normal. Preliminary results indicate that English speakers do use F2 as a proxy for f0.

**10:30–11:00 Discussion**

THURSDAY MORNING, 21 NOVEMBER 2024 — 10:00 A.M. TO 12:00 NOON

## Session 3aSCd

## Speech Communication: Choose-Your-Own-Adventure — Speech Potpourri I

Sarah Bakst, Chair
*SRI, Menlo Park, CA*

### *Contributed Papers*

10:00

**3aSCd1. Acoustic voice assessment reliability in varying reverberation times.** Ahmed Yousef (Dept. of Commun. Sci. and Disorders, Univ. of Iowa, 119 Wendell Johnson Speech and Hearing Ctr., Iowa City, IA 52240, aysef@uiowa.edu) and Eric Hunter (Dept. of Commun. Sci. and Disorders, Univ. of Iowa, Iowa City, IA)

Objective: Acoustic voice and speech assessment is a non-invasive and cost-effective tool for the clinic and the laboratory. However, the effect of room environment can impact the voice and speech metrics. This study investigates the threshold levels of simulated reverberation times above which acoustic measures become unreliable for voice quality assessment. Methods: Fifteen male and female subjects were recorded producing the sustained /a:/ vowel three times in a sound booth. Using Audacity software, the recordings were mixed with various simulated reverberation levels (T30). Acoustic measures related to voice quality assessment—such as shimmer, jitter, Cepstral Peak Prominence (CPP), and Harmonics-to-Noise Ratio (HNR)—were estimated for both original and reverberation-affected recordings. Statistical analyses were performed to determine significant differences in these measurements across the different reverberation levels. Results/Conclusions: The analysis demonstrated that CPP and jitter were more robust against high reverberation levels, showing high T30 threshold values, while shimmer and HNR were more vulnerable to reverberant rooms with low T30 threshold values. The outcomes recommend specific reverberation thresholds in clinics and recording environments for accurate acoustic measurements. Identifying these reverberation limits aids in optimizing recording environments and establishing standard room conditions for reliable acoustic voice assessments.

**10:05**

**3aSCd2. Impact of stimulus difficulty on forced alignment performance in L2 English learners: A focus on L1 Mandarin.** Gabin M. Mobétie (Biomed.i Information Processing Lab., Ecole de Technologie Supérieure (ETS) Montréal Canada, 3887 rue Saint-Dominique, Montréal, Quebec H2W 2A2, Canada, gmobetie@ensc.fr), Eija Aalto (Biomed. Information Processing Lab., Ecole de Technologie Supérieure (ETS) Montréal Canada, Edmonton, Finland), Walcir Carsodo (Linguistics, Concordia Univ., Montréal, Quebec, Canada), Lucie Menard (Linguistics, UQAM, Montréal, Quebec, Canada), and Catherine Laporte (Biomed. Information Processing Lab., Ecole de Technologie Supérieure (ETS) Montréal Canada, Montreal, Quebec, Canada)

This study investigates the performance of forced alignment algorithms and tools for second language (L2) English speech, more specifically L2 English speech produced by L1 Mandarin speakers. A recent study has shown that while the aligner's performance is less consistent with L2 speakers than with native speakers, current forced alignment methods are still accurate enough to be useful for studies of L2 English speech. However, no previous study has investigated whether the difficulty of the spoken content influences accuracy. In the present study, sentences designed to present different levels of difficulty to L1 Mandarin speakers were recorded. Participants included four L1 Mandarin speakers and one L1 English speaker as a control. All participants were associated with a subjective accentedness rating scale by two experienced listeners. A dataset containing recordings was created, which underwent preprocessing, including noise filtering and segmentation by silent periods. Statistical analysis revealed that alignment performance is correlated with accentedness but not with sentence difficulty. The study identified specific phonemic errors and possible differences in variance at different difficulty levels, highlighting the challenges faced by non-native speakers. Forced alignment methods appear to be well suited for phonetic analysis included with L1 Mandarin and L2 English contexts.

**10:10**

**3aSCd3. Which cross-language acoustic model to choose: Assessing tonality and phonemic inventories effects.** Hongchen Wu (School of Modern Languages, Georgia Inst. of Technol., Atlanta, GA) and Yixin Gu (School of Modern Languages, Georgia Inst. of Technol., Atlanta, GA 30313, ygu321@gatech.edu)

Cross-language forced alignment techniques, i.e., using a high-resource language acoustic model to automatically align audio files with transcripts for another language, present a promising and feasible approach for expediting the acoustic analysis of low-resource languages (Chodroff, Ahn, and Dolatian 2024). However, the factors influencing the effectiveness of cross-language forced alignment remain underexplored. Mandarin and Japanese share more similarities in phonemic inventories and syllable structures, while Japanese and English are both non-tonal languages. Comparing the alignment results generated by Mandarin and English acoustic models on Japanese audio data can help us explore whether tonality and phonemic inventory similarity have an equal influence on cross-language alignment performance. The present study used 30,975 Japanese audio files from Common Voice datasets as input and Montreal Forced Aligner Mandarin, English, and Japanese acoustic models to generate forced alignment output. In the 1.5 million data points across different comparison pairs, we found that when performing a cross-language alignment on a non-tonal language, a non-tonal language acoustic model is the optimal choice, but a tonal language acoustic model could also work decently if assigning vowels to a falling tone. These findings suggest tone matters in cross-language alignment and offer methodological insights for implementing cross-language alignment in low-resource languages.

**10:15**

**3aSCd4. A phonetic basis of accent bias in speaker identification technology.** Sarah Bakst (Speech Technology and Research Lab, SRI, 333 Ravenswood Ave., Menlo Park, CA 94025, sarah.bakst@sri.com), Ebony Pearson (Speech Technol. and Res. Lab., SRI, Menlo Park, CA), Luciana Ferrer (Computer Science Inst., Univ. of Buenos Aires - CONICET, Buenos Aires, Argentina), Mitchell McLaren (Speech Technology and Research Lab, SRI, Brisbane, Queensland, Australia), and Aaron Lawson (Speech Technol. and Res. Lab., SRI, Menlo Park, CA)

Speaker identification (SID) technology aims to determine whether the speech in a recording of an unknown speaker matches that of speaker known to the SID system. If the unknown speaker comes from a community that is underrepresented in the SID training data (e.g., accent), the SID model is more likely to confuse the unknown speaker for other speakers in that speaker community: group-level characteristics are mistaken for individual identifiers (all speakers with that accent seem like the same speaker to SID). Previous solutions include reweighting training data to create balance across speaker groups. While effective, such solutions are insufficient because they require foreknowledge of all potential speaker groups to be tested. The present study seeks a solution by first understanding how group-level characteristics are represented in SID models, focusing on dialect diversity within first-language English speakers. We hypothesize that filter properties (e.g. vowel formants) are more heavily represented in SID models than other uniquely-identifying spectral components. Preliminary results show that speaker clusters based on vowel measures alone are *dissimilar* to clusters formed by the SID model representations (embeddings) extracted from the same recordings—except for certain underrepresented accents (African American Language), suggesting SID representations of speech differ by speaker group.

**10:20**

**3aSCd5. Frame-wise short-term Fourier transform is no longer the best practice for speech analysis.** Hideki Kawahara (Ctr. for Innovative Res. and Liaison, Wakayama Univ., 930 Sakaedani, Wakayama 640-8510, Japan, kawahara@wakayama-u.ac.jp), Kohei Yatabe (Tokyo Univ. of Agric. and Technol., Tokyo, Japan), and Ken-Ichi Sakakibara (Health Sciences Univ. of Hokkaido, Ishikari-gun, Japan)

We propose to apply a fast implementation of the discrete Fourier transform (FFTW, Frigo, and Johnson 1998) to the whole speech segment to analyze instead of using short-term Fourier analysis (STFT) using frame-wise truncation and windowing. The discrete Fourier transform preserves all the information in the time-domain signal and provides its frequency-domain representation. We also propose to use a Gaussian windowing function without finite-length truncation. Instead of truncation, we let the machine epsilon effectively give the window an infinite domain. The advancement of computational power more than one billion times than half a century ago (e.g., Leiserson *et al*. 2020) made this excessive combination a practical and possibly a better alternative to current frame-based STFT practice. We introduce three tool sets to illustrate the benefit of this implementation. The first is acoustic environment assessment tools for better acquisition of speech material (Kawahara *et al*., 2023). The second is an objective evaluation tool for pitch extractors and a reference pitch extractor tuned by the tool. The third one is a real-time acoustic environment simulation tool to test their effects on speech production. We open-sourced the tools and placed tutorials on our YouTube channel.

**10:25**

**3aSCd6. Frequency and context prominence effects on English allophone perception: Identification of /r/ and /θ/.** Naomi Gurevich (CSD, Purdue Univ. Fort Wayne, 2101 E. Coliseum Blvd., Fort Wayne, IN 46805, gurevicn@pfw.edu) and Daniel Aalto (Communication Sciences and Disorders, University of Alberta, Edmonton, Alberta, Canada)

Purpose: External evidence from multiple cross-linguistics studies suggests the importance of acoustic clarity of speech sounds in words can vary for decoding speech (Gurevich & Kim, 2023b). Dependence on acoustic salience is modulated by the structure and function of words (Bell *et al*., 2003; Jun, 2011; Jurafsky *et al*., 2001; Scharenborg *et al*., 2016). This perception study of English examines the functional importance of positional

allophones to intelligibility. Methods: Speech produced by 21 speakers reading five randomized lists of 308 words with high coverage of English allophones (from Gurevich & Kim, 2023a) was presented to 11 naïve listeners for identification using a Latin-square design. The identification accuracy of the highest and the lowest frequency sounds were compared in their most and least prominent positions. Results: A total of 257 perceptual judgements were analyzed. Word-initial prevocalic /r/ and /θ/ had 97% and 75% accuracies, respectively; the pre-consonantal /r/ and inter-consonantal /θ/ had 95% and 43% accuracies, respectively. Conclusions: The results experimentally corroborate the expected hierarchy where higher frequency allophones in prominent contexts show higher accuracy compared to lower frequency allophones in less prominent contexts. Investigating allophones in additional languages will explore the effect of phonology on perception.

**10:30–11:00 Discussion**

THURSDAY MORNING, 21 NOVEMBER 2024 10:00 A.M. TO 12:00 NOON

## Session 3aSCe

## Speech Communication: In Honor of Ken Stevens — 100th Birthday

Stefanie Shattuck-Hufnagel, Chair
*Massachusetts Inst. of Technology, Cambridge, MA 02139*

### *Contributed Papers*

**10:00**

**3aSCe1. Perceptual learning of plain-ejective contrast.** Jack Mahlmann (Dept. of Linguistics, Univ. of Toronto, 100 St George St., Toronto, Ontario M5S 3G3, Canada, jack.mahlmann@mail.utoronto.ca) and Yoonjung Kang (Dept. of Language Studies, Univ. of Toronto Scarborough, Toronto, Ontario, Canada)

A recent cross-linguistic study finds that the presence of post-burst silence versus aspiration is a primary cue for ejective-plain stop contrast while VOT plays a minor role (Percival, 2024). This study explores whether English listeners can acquire this release cue after a short exposure and distributional training: "natural correlation" or "inhibition" (Kondaurova and Francis, 2010). Target words were created from Q'anjob'al /tu/ and /t'u/, distinguished by both VOT (short versus high) and release (aspiration versus silence). In the exposure phase, listeners heard stimuli along with pseudo-orthographic forms, <tu> or <t*u>. In the training phase, listeners categorized stimuli and were given feedback: the "natural correlation" group ($n = 31$) was trained on the same stimuli as the exposure phase; for the "inhibition" group ($n = 29$), only the release cue distinguished the stops and VOT varied from short to long for both stops. The pre-test and post-test, completed before and after the training phase, show that the release cue, already a primary cue prior to training for most listeners, especially with longer VOT, became stronger after training. No group difference was found. The novel release cue likely makes ejectives poor exemplars of English stops, therefore, easy to distinguish from plain stops (Best *et al.* 2001).

**10:05**

**3aSCe2. Asymmetrical effects of speaking rate on Japanese vowels and consonants.** Hironori Katsuda (Univ. of Toronto Scarborough, 1265 Military Trail, Toronto, Ontario M1C 1A4, Canada, katsuda1123@gmail.com) and Yoonjung Kang (Univ. of Toronto Scarborough, Toronto, Ontario, Canada)

Previous research suggests that vowels are more responsive to speaking rate variation than consonants in production. However, this tendency often derives from limited stimuli or broad comparisons that may overlook differences among subcategories like stops and fricatives. Additionally, how listeners adapt to the asymmetrical effects of speaking rate remains largely unknown. We conducted an online experiment with production and perception tasks using Japanese, where many segments have phonemic length. We examined whether different segmental targets, specifically five vowels (/i, e, a, o, u/) and five consonants (t, k, s, m, n/), differ in their sensitivity to speaking rate changes. In the production task, participants produced minimal pairs of nonce words contrasting in length (e.g., /kemp**i**na/, /kemp**iː**na/) within a carrier sentence both at fast and slow speaking rates, following production prompts. In the perception task, they categorized duration continua embedded in fast and slow speech as phonemically short or long. Converging evidence suggests a distinction between vowels and stops versus fricatives and nasals, with the former group exhibiting greater sensitivity to speaking rate changes in production. This production pattern was also generally reflected in the perception results. We discuss potential reasons behind this asymmetry and explore its implications.

**10:10**

**3aSCe3. From analysis-by-synthesis to predictive coding and beyond: The Martingale Dynamics of Human Speech.** Gordon Ramsay (Dept. of Pediatrics, Emory Univ., Marcus Autism Ctr., 1920 Briarcliff Rd. NE, Atlanta, GA 30329, Georgia, gordon.ramsay@emory.edu)

Ken Stevens is best remembered for his seminal contributions in developing theories of speech perception based on detection of acoustic features emerging from quantal properties of speech. Equally influential, however, is his earlier proposal for speech perception through analysis-by-synthesis, whereby hypotheses generated by an internal model of speech production are evaluated against sensory input. These ideas are recapitulated in recent theories of predictive coding, which also rely on monitoring discrepancy between simulations of potential action and actual sensation. Analysis-by-synthesis and invariant feature detection are often considered as alternative top-down and bottom-up approaches to explaining perception. Both viewpoints can, however, be reconciled within mathematical theories of optimal estimation, which offer two alternative, equivalent solutions to the problem of extracting information about the state of a dynamical system from partial observations. Both solutions rely on internal models, but one involves calculating the prediction error, as in analysis-by-synthesis, whereas the other involves calculating the relative salience of the sensory field, akin to landmark detection. This presentation provides a historical overview of analysis-by-synthesis, from cybernetics to predictive coding, and also reviews parallel developments in martingale dynamics that unite analysis-by-synthesis with landmark detection.

**10:15**

**3aSCe4. Do acoustics predict tongue shape in American English rhotics?** Amanda Eads (New York Univ., 665 Broadway, New York, NY 10012, are326@nyu.edu), Josephine Oh (Purdue Univ., West Lafayette, IN), Marcela Lara (New York Univ., Poughkeepsie, NY), Jonathan L. Preston (Syracuse Univ., Syracuse, NY), and Tara McAllister (New York Univ., New York, NY)

American English /ɹ/ is one of the most common and treatment-resistant speech deviations in children with residual speech sound disorder (RSSD; Flipsen, 2015) and has also been the focus of considerable basic science research on articulatory-acoustic relations (Zhou *et al*., 2008). English /ɹ/ can be produced using multiple tongue shapes that are commonly grouped into the broad categories bunched/tip-down and retroflex/tip-up. These shapes have a similar acoustic signature across the first three formants (Mielke *et al*., 2016) and are not readily perceptually distinguished (Twist *et al*., 2007). Previous work in adults has demonstrated that these shapes are acoustically distinct at the level of the 4th and 5th formants (F4 and F5; Espy-Wilson & Boyce, 1994; Zhou *et al*., 2008). The current study will investigate these acoustic-articulatory relationships using ultrasound and audio data from 36 children with no history of speech-language-hearing challenges and 30 children with RSSD affecting /ɹ/ who completed an ultrasound biofeedback treatment study. The /ɹ/ interval was segmented from the acoustic signal of each utterance and ultrasound frames in the rhotic interval were coded in a customized program. This provides ground truth articulatory labels for the data, thereby allowing for examination of relationships between F4 and F5 frequency values and bunched and retroflex tongue shapes.

**10:20**

**3aSCe5. Is articulatory skill connected to phonetic and phonological learning in monolinguals and bilinguals?** Dianna Diaz (CUNY Brooklyn College, 2900 Bedford Ave., Brooklyn, NY 11210, dianna.diaz72@bcmail.cuny.edu), Nicole Nieves, and Gabriella Diaz (CUNY Brooklyn College, Brooklyn, NY)

While research has revealed positive consequences of bilingualism on cognitive function, the findings are difficult to replicate and thus remain controversial. Emerging areas of research where a consistent bilingual advantage HAS been identified include studies on phonetic and phonological learning (PPL), defined as the ability to learn novel sounds or features of unfamiliar accents effectively. In the current study. we explore articulatory skill as a potential mechanism underlying the differences in PPL between monolingual and bilingual populations. We administer a PPL task (following Spinu *et al*. 2023, 2020) and investigate articulatory skill by training 10 English monolingual and 10 English-Spanish bilingual participants to produce unfamiliar speech sounds, i.e., secondarily palatalized labiodental fricatives, e.g., the word "hoof" is produced as [hufj]. The target consonants in the participants' baseline and experimental productions are compared in terms of duration, F1 and F2 of preceding vowels, and spectral peak during the frication portion. While data collection is in progress, we hypothesize that (1) bilinguals will outperform monolinguals on both tasks, (2) PPL correlates with articulatory skill, and (3) proficiency moderates the results across the board. Our findings will shed light on a lesser explored mechanism potentially underlying monolingual-bilingual differences in PPL.

**10:25**

**3aSCe6. Three-dimensional finite element acoustic analysis of bent vocal tracts.** Debasish Ray Mohapatra (Elec. and Comput. Eng., Univ. of British Columbia, 2366 Main Mall, Human Commun. Technol. Lab., Vancouver, British Columbia V6T 1Z4, Canada, debasishray@ece.ubc.ca) and Sidney Fels (Electr. and Comput. Eng., Univ. of British Columbia, Vancouver, British Columbia, Canada)

The bent air column in a vocal tract can significantly impact its acoustic characteristics by altering formant frequencies. However, existing vocal tract models typically assume the tract to be a straight three-dimensional (3D) tube with varying cross-sectional areas. Although wave propagation in realistic vocal tracts has been previously studied, the acoustic effect of the tract curvature on resonance frequencies has yet to be thoroughly explored. In this work, we use a state-of-the-art 3D finite element (FE) wave solver to characterize the effect of bending in acoustic ducts, akin to vocal tracts, by comparing their transfer functions up to 14 kHz. The duct geometries are modified to match vocal tracts for vowels /a/ and /u/. For a comparative analysis, we empirically adjust the degree of bending for a portion of the duct to increase its geometrical complexity. Our result shows that the bent air column does not significantly affect acoustic output for ducts with uniform cross-sections. However, transverse modes appear at higher frequencies for bent vocal tract geometries, i.e., ducts with varying cross-sectional areas. The comparison of transfer functions also shows that above 7 kHz, there is a notable shift in formant frequencies for such duct geometries.

**10:30**

**3aSCe7. Phonetic segment switching based on differential-encoding of the vocal tract resonances.** Brad H. Story (Speech, Language, and Hearing Sciences, Univ. of Arizona, 1131 E. 2nd St., P.O. Box 210071, Tucson, AZ 85721, bstory@email.arizona.edu) and Kate Bunton (Speech, Language, and Hearing Sci., Univ. of Arizona, Tucson, AZ)

Previous work has shown that speech segments can be encoded by specifying relative acoustic events along a time axis that consist of directional changes of the vocal tract resonance frequencies called resonance deflection patterns (RDPs) [JASA, 146(4), 2522–2528]. These events are transformed via acoustic sensitivity functions, into time-varying modulations of the vocal tract shape. RDPs specifying /b, p/, /d, t/, and /g, k/, for example, would typically be coded as [−1 −1 −1], [−1 1 1], and [−1 1 −1], respectively, where the vectors indicate, from left to right, the targeted directional shift of the first, second, and third resonances of the vocal tract. The specific aim of this study was to determine if listeners' recognition of speech synthesized in the RDP paradigm aligns with the RDP settings. That is, if RDPs are conceived as a bank of switches, can word recognition be guided by changes in the pattern of switch settings. A synthesized version of "abracadabra" was used as the baseline word. Variations were generated by systematically changing the RDPs for consonants in the initial "abra" and in the final "dabra" to generate new targets such as "aglacabadra." Listener responses to these variations will be reported.

**10:35**

**3aSCe8. Tongue bracing for improved control of vocal-tract model.** Takayuki Arai (Information and Commun. Sci., Sophia Univ., 7-1 Kioi-cho, Chiyoda-ku, Tokyo 102-8554, Japan, arai@sophia.ac.jp), Chisaki Nakagawa, Ryohei Suzuki, Chandler Earp, and Shinya Tsuji (Sophia Univ., Tokyo, Japan)

We have developed a series of physical models of the human vocal tract and sound sources for different purposes, including education, speech and language pathology, scientific studies, and many other applications. Some of the models have dynamic aspects of speech production and can simulate articulatory movements. When the tongue of a dynamic model approaches the palate, tongue bracing helps to form a midline groove on the surface of the tongue, especially for high vowels. Recently, we have fine-tuned the design of our version of Umeda and Teranishi's (UT) model to simulate continuous speech production. For the series of UT models, a set of plates or blocks line up next to each other to simulate the lower lip and tongue. The movements of these plates/blocks simulate the articulatory gestures. In the new model, the lateral and midline portions of each block for the tongue can be controlled independently. This allows us to achieve tongue bracing

and eventually minimizes errors between the target and the actual positions of the blocks with quick motions.

**10:40**

**3aSCe9. Whistling in alveolar fricatives: Evidence from two speech modes.** Luis M. Jesus (Univ. of Aveiro, Portugal, Escola Superior de Saúde (ESSUA), Universidade de Aveiro, Campus Universitário de Santiago, Agras do Crasto, Aveiro 3810-193 Aveiro, Portugal, lmtj@ua.pt)

The tongue constriction observed in alveolar fricatives, forming an unstable jet and boundary layer at the teeth, can be accompanied by a front cavity feedback mechanism that reinforces this instability, generating a narrow-bandwidth peak as observed in whistling. This study presents new evidence of a generalised use of this mechanism in speech, based on acoustic recordings of eight female speakers from the same dialectal region in Portugal. Sustained fricatives /s, z/, six disyllabic words, six sentences and a phonetically balanced text, were produced in voiced and whispered speech modes. Variation of the multitaper Power Spectral Density of not whistled/whistled fricative pairs was assessed using Functional Principal Component Analysis. The first of two principal components explained 93.3% of the /s/ variance and 90.4% of the /z/ variance. Results also revealed the following regarding the first shape descriptor used to model the curves: /s/ – sd = 969.55, $R^2 = 0.170$, $p = 0.000$, /z/ – sd = 931.41, $R^2 = 0.105$, $p = 0.000$; and the second shape descriptor: /s/ – sd = 257.32, $R^2 = 0.009$, $p = 0.248$, /z/ – sd = 299.57, $R^2 = 0.051$, $p = 0.007$. Whistled fricatives were observed both in voiced and whispered speech, so this does not seem to be a mechanism used to compensate for a weaker source strength as typically observed in whispered speech. Nevertheless, reconstructed not whistled/whistled curve pairs were significantly different.

**10:45–11:15 Discussion**

THURSDAY MORNING, 21 NOVEMBER 2024 — 10:00 A.M. TO 12:00 NOON

## Session 3aSCf

## Speech Communication: Choose-Your-Own-Adventure — Speech Potpourri II

Ivy Hauser, Chair
*Univ. of Texas at Arlington, Arlington, TX*

### *Contributed Papers*

**10:00**

**3aSCf1. Acoustic convergence to English sibilants is lexically-specific: Evidence for exemplar models.** Ivy Hauser (Linguistics, Univ. of Texas Arlington, 701 Planetarium Place, Box 19559—132 Hammond Hall, Arlington, TX 76019, ivy.hauser@uta.edu)

Convergence occurs when talkers alter pronunciation toward speech they hear. Exemplar models provide one account of convergence, as recently heard exemplars are activated during speech production (Goldinger, 1998). Exemplars of speech sounds are stored with their lexical item, predicting that convergence should be lexically-specific. This means that phonetic values should shift towards the model/interlocutor for each word, even if that entails increasing values for some words and decreasing them for others. To test this, a laboratory shadowing study was conducted using English /s/. 34 English speakers produced 80 /s/-initial words (balanced for lexical frequency and following vowel rounding) with fillers, then were passively exposed to model speech with the same /s/-initial words produced by a female talker, then produced the words again post-exposure. Spectral center of gravity (CoG) for /s/ was extracted using Praat (Boersma, 2001). Most participants produced baseline CoG values that were both lower and higher than the model for different words. When the data are analyzed by baseline relative to the model, a significant convergent pattern emerges: words initially produced with higher CoG shifted down after exposure and words initially produced lower shifted up. Participants converged to the model's acoustic values for CoG and direction of shift was lexically specific. This indicates that talker- and word-specific acoustic variability is stored and utilized, as predicted by exemplar models of speech sound representation.

**10:05**

**3aSCf2. Visual pattern completion predicts adaptation to acoustically degraded speech.** Julia Drouin (Univ. of North Carolina at Chapel Hill, 321 South Columbia St., 3105 Bondurant Hall, Chapel Hill, NC 27599, julia_drouin@med.unc.edu) and Charles Davis (Duke Univ., Durham, NC)

Listeners recognize acoustically degraded speech using incomplete auditory cues. Under predictive processing frameworks, partial auditory input is mapped to linguistic representations via a process of pattern completion. Previous research demonstrated that while listeners vary in their ability to recognize degraded speech, performance improves when a written transcription is presented with the auditory signal. Individual variation in recognizing degraded speech suggests that speech learning may rely on other cognitive processes. Building on work characterizing predictive processing as pattern completion, we examined the relationship between domain-general pattern recognition and degraded speech learning. Normal-hearing listeners (n = 85) completed a visual pattern recognition task and a degraded speech learning task with written transcriptions. Listeners were trained and tested on degraded speech pre- and post-training using a retrieval-based transcription task. Results showed training significantly improved listeners recognition of degraded speech. Critically, individual performance on the visual pattern completion task predicted the magnitude of improvement for novel sentences; tendency towards pattern completion, as opposed to pattern separation, predicted post-training performance for novel sentences. The results implicate pattern completion as a domain-general learning mechanism that may facilitate speech adaptation in challenging contexts, which may hold rehabilitation value for listeners with hearing loss.

**10:10**

**3aSCf3. Assessing individual speaker state and team based speech communication dynamics within the APOLLO lunar missions: Leveraging fearless steps APOLLO community resource.** Zahra Omidi (CRSS: Center for Robust Speech Syst., Univ. of Texas - Dallas, Richardson, TX) and John H. Hansen (CRSS: Center for Robust Speech Syst., Univ. of Texas - Dallas, 800 W Campbell Rd., Erik Jonsson School of Eng., EC32, P.O. Box 830688, Richardson, TX 75080-3021, John.Hansen@utdallas.edu)

In naturalistic speech communications, assessing individual speaker state and collaborative team dynamics is useful for problem solving in time sensitive tasks. The NASA Apollo program represents a massive longitudinal effort for team problem-solving. For the last decade, CRSS-UTDallas has worked to recover this data, creating the Fearless Steps Apollo (FS-APOLLO) community resource. This collection of 150000 h of digitized naturalistic audio and associated meta-data from advanced speech diarization/recognition is being released for speech science, technology, and historical preservation. This current effort focuses on comparing three Apollo missions to assess speaker stress/emotion/state and team communication dynamics. Apollo (A11, A7, and A13) missions were chosen, with 5 h of audio across five NASA team channels for each mission. A comprehensive speech and speaker analysis system was developed for independent measurement of stress, vocal intensity, Lombard effect, and emotion metrics. An integrated speech/speaker diarization pipeline was first applied to separate speech blocks. Next, this system was used to analyze Apollo team conversations, employing emotion recognition and task stress profiling to capture the social-linguistic-task based aspects of group interactions. Individual and team results will be presented from this study. Through this effort, we aim to create new opportunities for speech scientists and researchers to collaborate on developing effective metrics for speaker assessment.

**10:15**

**3aSCf4. Considering word token versus word type in the interpretation of Pillai scores: The case of the Cantonese AM/P∼OM/P merger.** Holman Tse (Literature, Language, and Writing, Saint Catherine Univ., 2004 Randolph Ave., Mail #4097, Saint Paul, MN 55105, hbtse110@stkate.edu)

Stanley & Sneller (2023) have recently demonstrated the importance of large sample sizes in the interpretation of Pillai Scores in sociophonetic research. In this paper, I address whether larger is necessarily better if larger means more tokens of the same word types. The case study focuses on the AM/P∼OM/P merger in Cantonese, which involves /o/ becoming /ɐ/ in prelabial contexts. The list of OM/P words is very small, though a few of these words are high frequency. The data come from spontaneous speech samples from 41 participants from the HLVC Corpus (Nagy 2011). Pillai Scores were calculated for each individual speaker based on both a smaller sample (five tokens maximum of the same word type, $n = 1163$) and a larger sample (more than five tokens of the same word type, $n = 1659$). Applying Stanley & Sneller's (2023) threshold value formula, results show 10 merged speakers with the smaller sample but five merged with the larger sample. Overall, these results show that while some researchers may have previously overestimated the spread of this merger (Bauer & Benedict 1997), increasing sample size without increasing word type diversity may lead to underestimation of the extent of the merger.

**10:20**

**3aSCf5. Internal /r/ epenthesis as gestural overlap in a memphis English emergent sound change.** Sydney M. Norris (Univ. of British Columbia, 2300 Lower Mall, Vancouver, British Columbia V6T1Z4, Canada, sydneynorris@live.com), Maddy Walter, Melissa Villasenor (Univ. of British Columbia, Vancouver, British Columbia, Canada), Maya Phoenix (York Univ., Guelph, Ontario, Canada), and Bryan Gick (University of British Columbia, Vancouver, British Columbia, Canada)

Temporal overlap of speech movements has occasionally been invoked as a post-hoc explanation for the historical emergence of a complex speech sound. An extreme case of this concerns so-called "internal" r-epenthesis in English dialects, limited to lexicalized pronunciations such as "warsh," "Warshington," "squarsh," and "garsh" in the Midland US, as well as "erster" and "berl" (for "oyster" and "boil") in many coastal dialects [Gick, 1999, Phonology 16]; Gick (p. 33, 50–51) hypothesizes an originating process for these forms in which the three component movements of English /r/ (i.e., lip-rounding, anterior tongue-raising and pharyngeal constriction), all present in the surrounding "rising" VC or VV sequences, temporally overlap, resulting in the "percept of an /r/." However, this hypothesis has been untestable as only two such examples are attested, both representing long-frozen sound changes. We report an apparently relevant in-progress sound change in Memphis English (ME), a dialect exhibiting the Southern Vowel Shift [Fridland, 1999, Lang. Var. Change 11]. Videos of younger ME speakers are analyzed in which internal r-epenthesis is replacing the falling diphthong [ju] in words, such as "human," "Cuba," "music," etc. Preliminary observations support a temporal overlap hypothesis; acoustic/structural analysis will be presented and implications discussed.

**10:25**

**3aSCf6. Generational dynamics in velar palatalization: An acoustic study of Tohoku Japanese.** Naoya Watabe (Univ. of Tokyo, 3-8-1, Komaba, Meguro-ku, Tokyo 1538902, Japan, w.naoya2821@gmail.com), Chuyu Huang (Nagoya Gakuin Univ., Nagoya, Japan), Ai Mizoguchi (Maebashi Inst. of Technol., Maebashi, Japan), Hiroto Noguchi (Tokyo Med. and Dental Univ., Tokyo, Japan), Ayako Hashimoto (Tokyo Kasei Gakuin Univ., Tokyo, Japan), Mafuyu Kitahara (Sophia Univ., Tokyo, Japan), Megumi Kimoto (Kobe Univ., Kobe, Japan), and Sanae Matsui (Sophia Univ., Chiyoda, Tokyo, Japan)

This study examines the contextual and generational differences in consonantal palatalization of Tohoku Japanese compared to Tokyo Japanese. Typically, both Japanese dialects transform alveolar obstruents to alveo-palatals before /i/. Moreover, previous research has shown that Tohoku Japanese also exhibits palatalization of velar stops /k, ɡ/. However, this palatalization ceases to appear among the younger generations despite the scant acoustic evidence. This study aims to: (i) describe the occurrence of velar palatalization in Tohoku Japanese and (ii) analyze variations between Tohoku and Tokyo Japanese by socio-demographic factors including age and gender. Twenty-five speakers from Tohoku and twenty-one from Tokyo participated in a production task using target words containing /k/ in the /ki/ and /ka/ contexts, and /tɕ/ as the palatalization baseline. The results of Center of Gravity (CoG) revealed significant interactions between the target consonant, dialectal group, and age group in the palatalization patterns. Younger speakers of Tohoku Japanese show a similar CoG distribution that resembles those of Tokyo speakers, particularly in the pairwise comparison of /k/ in /ki/ and /ka/. This empirically suggests a converging shift among young Tohoku speakers toward the standard variation of Japanese, supporting a diminishing regional variation correlated with consonantal context and influenced by generational change.

**10:30–11:00 Discussion**

# Session 3aSP

## Signal Processing in Acoustics: Explainable Artificial Intelligence

Bernice Kubicek, Cochair
*Elec. and Comput. Eng., Univ. of Iowa, 103 South Capitol St., Iowa City, IA 52242*

Ananya Sen Gupta, Cochair
*Dep. of Elec. and Comput. Eng., Univ. of Iowa, 103 S Capitol St., Iowa City, IA 52242*

**Chair's Introduction—10:00**

### *Invited Papers*

**10:05**

**3aSP1. Goal reasoning for intelligent parameter adaptation in active sonar.** Jill K. Nelson (Elec. and Comput. Eng., George Mason Univ., 4400 University Dr., MSN 1G5, Fairfax, VA 22030, jnelson@gmu.edu)

Active sonar systems include a variety of parameters that can be dynamically tuned to improve system performance. In practice, however, parameters are typically set and fixed over long periods based on expected environmental conditions, prior performance, etc. Dynamic tuning of system parameters by sonar operators is impractical due to both the short time frame between adaptations and the complex relationship among the parameters, goals, and system performance. In intelligent active sonar, the system tunes parameters based on a set of goals and evaluation of how well those goals are being met. This approach significantly reduces the cognitive load on the operator, who can set high-level goals that are interpreted by the system and translated into low-level parameter adjustments. Goal-driven autonomy (GDA), for example, detects discrepancies between predictions and observations and generates context-specific explanations when significant discrepancies are observed. These explanations may indicate that new goals need to be introduced or existing goals de-activated. The explanation generation and goal formation aspects of GDA provide insight into how the intelligent system is reasoning about its actions and allow an operator to have direct input to the intelligent system when desired.

**10:12**

**3aSP2. Can explainable AI increase trust in seabed predictions from ship noise?** Tracianne B. Neilsen (Physics and Astronomy, Brigham Young Univ., N251 ESC, Provo, UT 84602, tbn@byu.edu)

The recent increase of artificial intelligence (AI) systems has led to several questions, including "Can AI assessments or predictions be trusted?" and "How can people be encouraged to accept AI-based recommendations?" These questions have led to the rise of explainable AI. This short talk presents a scenario in which AI could provide information about the seabed to a sonar operator. In this scenario, received ship noise is input to a deep learning model trained to predict a seabed class. The predicted seabed class is then used to calculate transmission loss (TL) as a function of range. These TL curves from the predicted seabed class are compared to those obtained using the seabed information stored in a database. This comparison could allow the sonar operator to evaluate the applicability of the seabed database for their present location. Adoption of this type of AI tool depends on the attitude of the sonar operator about the AI and its predictions. Ideas are given for how this attitude could be improved by adoption of explainable AI techniques. [Work supported by the Office of Naval Research grant #N00014-22-12402.]

**10:19**

**3aSP3. Metamers in deep learning for synthetic aperture sonar image classification and their significance in understanding deep networks.** Isaac Gerg (Computer Vision, Kitware, Inc., 1712 US-9 #300, Clifton Park, NY 12065, isaac.gerg@gergltd.com)

Metamers in deep learning are images that appear different but elicit the same feature vector response at the penultimate layer. These images can be easily discovered using optimization methods built into existing deep learning training frameworks. Typically, metamers look nothing like the training data, despite having feature vectors that match exactly with existing training samples. This phenomenon emerges as a function of network layer depth, where initially dissimilar representations converge to a representation of an existing training image. Conversely, simple transforms like image despeckling may have the opposite effect: initial representations similar to a training sample diverge as a function of layer depth until they diverge so much that they become misclassified. In this discussion, we explore this phenomenon, provide practical examples, and aim to stimulate conversation about this intriguing behavior of deep networks.

**10:26**

**3aSP4. Improving transparency and adaptability of undersea object recognition systems.** Luke DeYoung (Johns Hopkins Univ., Appl. Phys. Lab., 11100 Johns Hopkins Rd., Laurel, MD 20723, Luke.DeYoung@jhuapl.edu), Alexander Swaters, and Dave Barsic (Johns Hopkins Univ. Appl. Phys. Lab., Laurel, MD)

Side-scan sonars are commonly deployed on autonomous underwater vehicles for seafloor imaging and search applications. In recent years, deep neural networks have become a common solution for object recognition problems with these systems, achieving impressive performance. However, adaptability and interpretability of these systems remain a challenge. The models used in these systems are typically tuned for recognition of specific categories of objects and, due to the labor required to create segmentation mask labels, are often limited to bounding box prediction. Meta's Segment Anything Model (SAM), released in 2023, provides a solution to add instance segmentation capabilities to existing object recognition systems. SAM may be fine-tuned on side-scan sonar data to predict separate object and acoustic shadow masks, allowing for automatic and robust measurement of undersea objects. In addition, this fine-tuned variant of SAM may then be integrated with existing object recognition algorithms tuned for side-scan sonar image analysis to transform them into instance segmentation algorithms. The end result is a highly descriptive recognition system with customizable reporting behavior based on the dimensions of the object(s) of interest.

**10:53**

**3aSP5. Data preprocessing for seabed characterization with machine learning.** Zoi-Heleni Michalopoulou (Dept. of Math. Sci., New Jersey Inst. of Technol., 323 M. L. King Boulevard, Newark, NJ 07102, michalop@njit.edu)

Data received when sound propagates in oceanic waveguides exhibit a behavior characterized by the physics of the propagation medium. These data can be presented as input to machine learning algorithms for environmental characterization, in our problem seabed identification. A key factor for the success of the task at hand is how the data are employed within the classification algorithms and whether raw signals or qualitative and quantitative features extracted from those are used at the input level. In this work, data are studied both in the frequency and time domains and their behavior is analyzed. It is observed that employing simple characteristic attributes extracted from recorded signals is equivalent to identifying seabed signatures, leading to effective and efficient sediment classification. [Work supported by ONR.]

**11:00**

**3aSP6. Feature interpretation in underwater acoustics.** Gopu R. Potty (Ocean Eng., Univ. of Rhode Island, 215 South Ferry Rd., Narragansett, RI 02882, gpotty@uri.edu)

Deep Learning algorithms such as Convolutional Neural Networks (CNN) have been widely used recently in underwater acoustics applications including target recognition, passive acoustic monitoring of marine mammals, parameter estimation, etc. In many of these applications, spectrogram images are used to train the algorithms to identify the unique features. CNNs suffer from the black box effect due to their large number of parameters and developing methods to transform this black box algorithm into an interpretable one is an active area of research. This talk will highlight the challenges in interpretation of machine learnt features based on acoustic physics of propagation. How such knowledge of propagation physics may be incorporated into neural networks (e.g., through physics informed Neural Networks) would be explored. [Work supported by Office of Naval Research.]

**11:07**

**3aSP7. Explainable AI for trustworthy image analysis.** Jordan E. Turley (Johns Hopkins Appl. Phys. Lab., 11100 Johns Hopkins Rd., Laurel, MD 20723, Jordan.Turley@jhuapl.edu), Jeffrey A. Dunne, and Zerotti Woods (Johns Hopkins Appl. Phys. Lab., Laurel, MD)

The capabilities of convolutional neural networks to explore data in various fields has been documented extensively throughout the literature. One common challenge with adopting AI/ML solutions, however, is the issue of trust. Decision makers are rightfully hesitant to take action based solely on "the computer said so" even if the computer has great confidence that it is correct. There is obvious value in a system that can answer the question of why it made a given prediction and back this up with specific evidence. Basic models like regression or nearest neighbors can support such answers but have significant limitations in real-world applications, and more capable models like neural networks are much too complex to interpret. We have developed a prototype system that combines convolutional neural networks with semantic representations of reasonableness. We use logic similar to how humans justify conclusions, breaking objects into smaller pieces that we trust a neural network to identify. Leveraging a suite of machine learning algorithms, the tool provides not merely an output "conclusion," but a supporting string of evidence that humans can use to better understand the conclusion, as well as explore potential weaknesses in the AI/ML components. This paper will provide an in-depth overview of the prototype and show some exemplar results. [Work supported by the Johns Hopkins University Applied Physics Laboratory.]

**11:14**

**3aSP8. Uncertainty-aware classification of underwater acoustic signals using Bayesian deep learning.** Marko Orescanin (Comput. Sci., Naval Postgraduate School, 1 University Circle, Monterey, CA 93943, marko.orescanin@nps.edu)

This talk presents our work on applying Bayesian deep learning methods to provide interpretable and uncertainty-aware classification of underwater acoustic signals. We demonstrate how Bayesian convolutional neural networks can accurately classify both passive sonar and synthetic aperture sonar data while quantifying prediction uncertainty—a critical capability for autonomous systems. By examining estimated aleatoric and epistemic uncertainties, we gain insight into the model's decision-making process and areas of low confidence. We leverage predictive entropy and calibrated uncertainties to identify challenging samples, explain misclassifications, and reveal patterns in model behavior. Our results show how uncertainty correlates with factors like range, aspect angle, and seafloor texture in ways that align with acoustic propagation physics. We also discuss how active learning strategies guided by epistemic uncertainty can

efficiently train models while providing further model interpretability. This work establishes Bayesian deep learning as a promising approach for developing explainable and reliable underwater acoustic classifiers, addressing the need for transparency in AI-driven underwater sensing applications.

**11:21–11:41 Discussion**

THURSDAY MORNING, 21 NOVEMBER 2024 — 10:00 A.M. TO 12:00 NOON

## Session 3aUW

## Underwater Acoustics: Invited Speaker Talk Outside the Field of Underwater Acoustics

David Dall'Osto, Chair
*Appl. Phys. Lab. at the Univ. of Washington, 1959 NE Pacific St., Seattle, WA 98195*

**Chair's Introduction—10:00**

### *Invited Paper*

**10:05**

**3aUW1. On the role of model resolution for the prediction of ocean sound speed variability in submesoscale-rich flows.** Annalisa Bracco (School of Earth and Atmospheric Sci., Georgia Inst. of Technol., 311 Ferst Dr., Atlanta, GA 30332, annalisa@eas.gatech.edu)

I will summarize a body of recent work performed to investigate the role that mesoscale and submesoscale circulations have on the prediction of ocean sound speed. I will use a suite of regional simulations performed at horizontal resolution varying between 500 m and 1 km, with 30 to 200 vertical levels. Regions of interest are the DeSoto Canyon in the Gulf of Mexico, where strong density gradients fuel submesoscale instabilities year around, and the Atlantis II Sea Mount in the North Atlantic, where flow-topographic interactions constrain the dynamics. The simulations partially resolve submesoscale dynamics with different degrees of realism. I will briefly introduce how eddies, fronts, salt fingering, and lee waves can affect circulation and sound propagation at different depths in the water column, and present challenges associated with their representation in ocean models.

**11:00–12:00 Discussion**

# Session 3pAA

## Architectural Acoustics: Potpourri Acoustics

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

### *Contributed Papers*

**3:00**

**3pAA1. An analysis of spatial impulse response measurements and their ability to validate spatial features within an acoustic model.** John Latta (Univ. of Nebraska - Lincoln, 1110 S. 67th St., Omaha, NE 68182-0816, jlatta@wjhw.com) and Lauren Ronsse (Univ. of Nebraska - Lincoln, Omaha, NE)

Conventional impulse response measurements are commonly used to validate omnidirectional metrics within ODEON models. ODEON models can also provide further analysis of the spatial properties of spaces. However, the spatial properties of a model cannot be validated using exclusively conventional measuring techniques. Instead, spatial impulse response measurements must be performed to properly validate the spatial components of an ODEON model. Conventional impulse and spatial impulse response measurements performed at the Recital Hall in the Strauss Performing Arts Center at the University of Nebraska at Omaha were used to create and validate acoustic metrics of an ODEON model of the same space. These spatial impulse response measurements contained visualizations of directional energy over time to assess reflections. ODEON produces several graphics comparable to those produced by the measurements. Across both visualization results, early reflections are nearly identical. However, at late energy reflections, larger discrepancies arise. This is likely due to either ODEON, the spatial measurement processing software, or the ambisonics microphone used to capture the spatial measurements. The accuracy of early energy data across both methods allows spatial impulse response measurements to be used in acoustic design, as a method of validating acoustic models or examining the changes in real-world reflections.

**3:15**

**3pAA2. Passive hearing abilities of dyslexia children: Investigating the rumble frequency of skull bones.** Ibrahim Y. Elnoshokaty (Acoustics, ENOSH Sci. Ctr., 22 Villa Mohamed Mahmoud Kassem St., Elhegaz Sq., Cairo, Heliopolis 11431, Egypt, ibrahim@enoshmink.com), Howida Elgabaly, and Manal Omar (Faculty of Postgraduate Childhood Studies, Ain Shams Univ., Cairo, Egypt)

Dyslexia is a specific learning disorder characterized with impairment of auditory frequency discrimination, impaired perception of amplitude, and modulation. The relation between sound discrimination and the passive hearing and sound conduction through the skull bone resonance cavities is still unclear. Methodology: 31 children diagnosed by dsm-5 as Dyslexia, sample collected from the dyslexia unit in( snsc) affiliated to, Asu.,along July 2022 to July 2023 with age range from 6 to 14 years, and both sexes were included. Material and apparatus: A calibrated 8 microphones for record and analysis at 63 Hz to 16 kHz via customised skull bone resonance cavity helmet. while seated in a room with RC25 chamber in a quiet laboratory Results: 31 dyslexic children age range 6_14, 22 males, and 9 females. normal bone resonance resulted in 8 children, while abnormal bone resonance was in 23 children. Two categories of abnormalities of the sound: (a) damping frequency, at a low/mid frequency ranging between 120 and 1000 Hz in 8 children. (b)rumble frequency, damping at a frequency ranging between 120 and 500 Hz, and further increase to range 1000 to 8000 Hz in 15 children. The sinuses sound abnormalities were in frontal and mastoid equally $n = 21(67.7\%)$, followed by maxillary $n = 19(61.29\%)$, more dominant on the right side. Conclusion: Investigating the skull bone resonance cavities can have good reflection on the sinuses condition, where further investigations will be mandatory and clinical intervention of the sinuses disease is to be essential.

**3:30**

**3pAA3. Experts from multiple fields exchange perspectives on optimizing the acoustic environment in healthcare.** Bonnie Schnitta (SoundSense, LLC, 39 Industrial Rd., Unit 6, PO Box 1360, Wainscott, NY 11975, bonnie@soundsense.com), Joanne Solet (Harvard Medical School, Cambridge, MA), Francis M. Pitts (Lomonaco & Pitts, Architects P.C., Troy, NY), Sean Harkin, Nicholas Roselli (SoundSense, LLC, New York, NY), Omar Hamza (Cambridge Health Alliance, Cambridge, MA), and Paul Acre (None, Greenbrier, AR)

This diverse panel of experts will discuss the impact and challenges associated with acoustics in healthcare from multiple perspectives: design, regulation, clinical care, research, and engineering, and will entertain questions from participants.

**3:45–5:00 Discussion**

# Session 3pAB

## Animal Bioacoustics: Climate Change and Animal Bioacoustics

Edward J. Walsh, Chair
*VA Loma Linda Healthcare Syst., Loma Linda, CA 92357*

**Chair's Introduction—3:00**

### *Contributed Papers*

**3:05**

**3pAB1. Binaural echo error detection gives self-supervised learning to improve landmark classification.** Roman Kuc (Elec. Eng., Yale, 511 Becton, 15 Prospect St., New Haven, CT 06511, roman.kuc@yale.edu)

Binaural processing is typically associated with source localization using interaural time and level differences. This paper describes its role for improving landmark classification from an echo sequence. A brain-inspired system explores the blind human echolocation problem of differentiating two foliage targets with different sized leaves using audible echoes. A biomimetic sonar views each target by producing left and right-ear monaural echo waveforms whose target-specific frequency power spectra are classified using template matching. Binaural processing implements error detection by ignoring views that give contradictory monaural classifications. Binaural classification using consistent monaural classifications also produces self-labeled data that update monaural template estimates to continually improve target classification accuracy through additional target encounters. Monaural classification error probability starts at 0.225 after supervised learning with 10 encounters with each known target and reaches 0.134 after an additional 90 self-supervised template updates. The binaural classification error probability starts at 0.032 with an average of 1.49 binaural views required to achieve consistent monaural classifications and reaches 0.019 with an average of 1.32 views.

**3:17**

**3pAB2. Ocean soundscapes reveal climatic and economic oscillations.** Vanessa M. ZoBell (Scripps Inst. of Oceanography, Univ. of California San Diego, 812 Ocean Surf Dr., Solana Beach, CA 92075, vmzobell@ucsd.edu), Natalie Posdaljian, Kieren Lenssen, Sean Wiggins, John Hildebrand, Simone Baumann-Pickering, and Kait Frasier (Scripps Inst. of Oceanography, Univ. of California San Diego, La Jolla, CA)

An accumulation of 26 859 days of acoustic recordings (spanning over 74 years at six sites) was processed and analyzed to investigate the interannual, seasonal, and diel patterns within six ocean soundscapes. Biological sound levels (18-25 Hz and 40–45 Hz) quantified acoustic energy from fin and blue whales, ambient sound levels (63-Hz and 800-Hz one-third-octave) quantified energy from ships and wind, and broadband spectra (15 Hz to 1 kHz) identified changes across time and space. Biological sound levels varied seasonally and correlated with large-scale climatic patterns and long-term ocean fluctuations. During marine heatwaves, baleen whale associated sound levels decreased in southern sites and increased in sites adjacent to the California Current. Ship sound levels at high-traffic sites reflected economic events such as the financial crisis, labor shortages and negotiations, and changes to port flows. Wind speeds and associated sound levels reflected on-shore/off-shore relationships and decreased during the morning hours. Understanding marine soundscapes aids in understanding the ocean's ecological health amidst the ever-changing impact of climate change.

**3:29**

**3pAB3. Are offshore windfarms killing whales?** Michael Stocker (Ocean Conservation Res., P.O. Box 559, Lagunitas, CA 94938, mstocker@ocr.org)

A noticeable rise in mysticete mortalities in and around New Jersey concurrent with the increase in activities associated with offshore windfarm development has excited speculation that the windfarm development is at cause of the strandings. Various competing and sometimes countervailing interests have taken up an array of narratives on what is occurring, so inquiry into the topic is fraught with speculation and motives. This presentation will examine the conditions, circumstances, and motives of the 2022–2023 Jersey Shores "Unusual Mortality Event" (UME) with the objective of stimulating open conversation about this event—toward avoiding reoccurrences of similar events, and similar discussions.

**3:41**

**3pAB4. Identifying narwhal vocalizations to assess marine conservation areas in the changing Arctic.** Juliana Rodrigues Moron (School of Earth and Ocean Sci., Univ. of Victoria, 1622 Camosun St., Victoria, BC V8T 3E6, Canada, jrodriguesmoron@uvic.ca), Stan Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, British Columbia, Canada), and William D. Halliday (Wildlife Conservation Soc. Canada, Whitehorse, Yukon, Canada)

Climate-change related increases in sea surface temperatures and associated declines in sea ice in the Arctic are driving phenological shifts in habitat use of several species of marine mammals, which are also facing higher noise levels from increasing human activities. While some protected areas have been established, their effectiveness for species of interest, such as narwhal, have not been assessed. Sound is important for communication and navigation in marine mammals, which makes passive acoustic monitoring an appropriate tool for studying these species, being particularly effective in remote and harsh Arctic environments. Recordings collected in the Disko Fan and Davis Strait Conservation Areas (southern Baffin Bay), known habitat for narwhals, are analyzed using a new method for identifying narwhal whistles in long-term passive acoustic datasets. We identify narwhal whistles by assuming that any whistles produced around the same time as narwhal echolocation clicks (manually identified on spectrograms) were produced by narwhal. These whistles are then used to train a new deep learning detector to process long-term passive acoustic datasets for narwhal presence. This is an important step toward understanding the impacts of climate change on the distribution of this species and the effectiveness of conservation areas.

**3:53**

**3pAB5. Assessing Northern Gulf of Mexico Sperm Whale's (*Physeter macrocephalus*) seasonal patterns using passive acoustic monitoring.**
Naomi M. Ciampaglio (Physics, Univ. of Louisiana, 219 Rosemary Pl., Lafayette, LA 70508, naomi.ciampaglio1@louisiana.edu) and Natalia Sidorovskaia (Physics, Univ. of Louisiana at Lafayette, Lafayette, LA)

Sperm whales (*Physeter macrocephalus*) show high sexual dimorphism and sexually regulated group structures. In the Northern Gulf of Mexico (NGoM), seasonal population demographics and the group structures have not been closely analyzed. Previous studies suggest that NGoM sperm whale population is dominated by females and juveniles, However, mature males have been known to have a presence as well. Utilizing Environmental Acoustic Recording System buoys, continuous acoustic recordings were acquired from June 2018 through June 2020. The buoys were deployed off the coast of Louisiana and the data were analyzed using a sperm whale click detector algorithm developed by our research team (Tiemann *et al*., 2011) as well as CABLE (Cachalot Automatic Body Length Estimator) software (Beslin *et al*., JASA 2018) for detecting sperm whale presence and estimating body lengths. Preliminary results showed that sperm whales are present all year with higher acoustic activity during the winter. The data also indicate a higher female and juvenile presence. Future investigations focus on understanding monthly and daily behavior of the NGoM sperm whale groups. The results will contribute to estimating NGoM sperm whale time budget. [This research was made possible in part by grants from The Gulf of Mexico Research Initiative and BOEM.]

**4:05**

**3pAB6. Under-ice ambient noise and its masking effect on marine mammals in the Arctic.**
Najeem Shajahan (Oceanography, Dalhousie Univ., Wildlife Conservation Soc. Canada, Room 5611, Oxford St., Halifax, Nova Scotia B3H4R2, Canada, najeemtkm@gmail.com), William D. Halliday, and Stephen Insley (Wildlife Conservation Soc. Canada, Whitehorse, Yukon, Canada)

Climate change is changing the pattern of sea ice in the Arctic, which increases underwater ambient sound levels during the ice-covered season. The Arctic is home to multiple marine mammal species, even during the winter, which rely on the underwater acoustic environment for acoustic communication. Thus, climate change may be impacting the ability of these animals to communicate. Under-ice ambient noise is primarily generated by wind acting on the ice surface, rapid temperature changes, and impulsive transient signals resulting from ice cracking and ridging. To characterize under-ice ambient noise, acoustic data were collected from several locations in the western Canadian Arctic in this study. To determine the relationship between the received noise level, and wind speed and ice concentration, correlation analysis was used. For the detection and analysis of broadband transient events in the acoustic data, a spectrogram image processing approach was used. The spectral characteristics of transient events were generated using measurements and a random pulse train model. Furthermore, the wind and ice generated noise components were combined to predict the total ambient noise field in an ice-covered ocean using a wavenumber integral model. Finally, the model together with measurements was used to assess how ice-generated noise can cause communication masking in marine mammals such as ringed seals and bearded seals.

**4:06**

**3pAB7. Syllable element typology in grasshopper sparrow warble song.**
Bernard Lohr (Dept. of Biol. Sci., Univ. of Maryland Baltimore County, 1000 Hilltop Circle, Baltimore, MD 21250, blohr@umbc.edu) and Rebecca Hill (Dept. of Biol. Sci., Univ. of Maryland Baltimore County, Baltimore, MD)

Song structure in birds often varies geographically, and understanding the patterns of this variation can give us insights into how vocal behaviors act as culturally transmitted traits. The warble song of grasshopper sparrows contains syllable types that may be shared between individuals both within and across populations. We investigated how syllable structure varied geographically in this song type by developing a library of syllable types using recordings from several North American populations, representing three distinct subspecies. We made initial assignments to this library based on syllable duration, bandwidth, and acoustic structure. Then we quantified our visual assessment of syllable types using a cross-correlation analysis—threshold values for syllable type assignments were identified by discontinuities in the histograms of cross-correlation similarity scores. Our sorting procedure classified $>90\%$ of all syllables. We found that there were many common syllable types across all populations. However, the proportions of many syllable types, and first-order Markov transitions from one syllable type to another, often differed across populations and, especially, across subspecies. Grasshopper sparrow warble song syllables appear to stem from a library of common syllable types, but populations and subspecies differ in the use and ordering of those syllables.

**4:07**

**3pAB8. Acoustic vector sensing reveals ecologically modulated sympatry of blue and fin whales.**
John P. Ryan (Research, MBARI, 7700 Sandholdt Rd., Moss Landing, CA 95039-9644, ryjo@mbari.org), Paul Leary, Kevin B. Smith (Naval Postgraduate School, Monterey, CA), Jarrod Santora (Fisheries Ecology Div., NOAA, Santa Cruz, CA), Andrew DeVogelaere, Chad King (Monterey Bay National Marine Sanctuary, Monterey, CA), Francisco Chavez (Research, MBARI, Moss Landing, CA), Vanessa M. ZoBell (Biol. Oceanography, Scripps Inst. of Oceanography, Solana Beach, CA), Tetyana Margolina, John E. Joseph (Naval Postgraduate School, Monterey, CA), William Oestreich, Kelly Benoit-Bird (Monterey Bay Aquarium Research Institute (MBARI), Moss Landing, CA), and Jeremy A. Goldbogen (Hopkins Marine Station, Stanford Univ., Pacific Grove, CA)

Earth's largest animal species, blue and fin whales, share a similar life history strategy involving annual long-distance migration between foraging and breeding habitat. The difficulty of monitoring spatial distributions of both species simultaneously has limited understanding of their sympatry within foraging habitats that sustain them. In addressing this observational challenge, a valuable method is acoustic vector sensing of the sounds produced by the whales. Using three years of continuous acoustic vector sensor data, we examine habitat occupancy of blue and fin whales in the central California Current Ecosystem. While fin whales called almost exclusively from offshore deep-water habitat, blue whales called predominantly from shelf and slope water habitat—indicating an allopatric tendency. However, stronger sympatry occurred during the third year, as blue whales spent a greater proportion of time in offshore deep-water habitat. This year was distinguished by the strongest wind-driven coastal upwelling, the furthest offshore extension of upwelled waters, and the greatest abundances of krill that are prey of both whale species. upporting the recovery of endangered blue and fin whale populations requires understanding where and how they live, and acoustic vector sensing is effective for this purpose in ecological research and resource management.

THURSDAY AFTERNOON, 21 NOVEMBER 2024 3:00 P.M. TO 4:00 P.M.

## Session 3pAO

## Acoustical Oceanography: Program Managers Roundtable

Lauren Freeman, Chair
*NUWC Newport, NUWC, Naval Undersea Warfare Ctr., 1176 Howell St., Newport, RI 02841*

**Chair's Introduction—3:00**

### *Invited Paper*

**3:05**

**3pAO1. Program Manager's roundtable.** Lauren Freeman (NUWC Newport, NUWC, Naval Undersea Warfare Ctr., 1176 Howell St., Newport, RI 02841, lauren.a.freeman3.civ@us.navy.mil)

Join us for a one-hour virtual discussion with program managers from major US funding agencies including the Office of Naval Research and National Science Foundation, with experience funding programs in ocean acoustics, underwater acoustics, or sea-going research. Panelists will cover pre-planned questions on when and how to reach out to a program manager, how to write an effective planning letter and proposal, and current available opportunities for different career levels (fellowships for students, early career awards, and funding opportunities for ocean research/tech development); as well as comment on their own career paths. Questions will be collected in advance as well as taken during the session by the moderator. This is an opportunity to both learn what it takes to be a program manager, and strategies and tips for learning about funding opportunities and responding effectively.

THURSDAY AFTERNOON, 21 NOVEMBER 2024 3:00 P.M. TO 4:30 P.M.

## Session 3pBA

## Biomedical Acoustics: Show Me Your Lab Crib!

Timothy L. Hall, Cochair
*Univ. of Michigan, 2200 Bonisteel Blvd., 1107 Gerstacker Bldg., Ann Arbor, MI 48109*

Ellen Yeats, Cochair
*Biomed. Eng., Univ. of Michigan, 2200 Bonisteel Blvd., Ann Arbor, MI 48109*

### *Invited Paper*

**3:00**

**3pBA1. LabTAU: Laboratory of therapeutic applications of ultrasound.** Maxime Lafond (LabTAU - INSERM U1032, 151, cours Albert Thomas, Lyon 69424, France, maxime.lafond@inserm.fr), Adrien Rohfritsch (LabTAU - INSERM U1032, Lyon, France), Thomas Payen (EDAP TMS, Vaulx en Velin, Rhône, France), and Cyril Lafon (LabTAU - INSERM U1032, Lyon, France)

The LabTAU is a household name in the history of biomedical ultrasound. Our philosophy is to transfer our research from the bench to the patient. As such, we are striving to be at the forefront of emerging techniques while nurturing clinical and industrial partnerships. Our research focuses on four main areas: high-energy ultrasound, low-energy ultrasound, bubbles cells and particles, and wave and

instrumentation. After briefly presenting the geographical context of the laboratory, we tour the facility, showcasing our main experimental platforms by using examples of active research across the spectrum from fundamental physics and technology to clinical developments.

## *Contributed Papers*

**3:15**

**3pBA2. Physical acoustics lab at the university of oxford.** James Kwan (Dept. of Eng. Sci., Univ. of Oxford, Parks Road, Oxford OX1 3PJ, United Kingdom, james.kwan@eng.ox.ac.uk) and Ronald A. Roy (Dept. of Eng. Sci., Univ. of Oxford, Oxford, United Kingdom)

We want to welcome everyone to have a look at our lab's space where the academic magic happens. Our group boasts some unique bespoke acoustic equipment and experimental methods that cover topics including physical acoustics, ultrasound enhanced drug delivery, and sonochemistry. In the video, we will walk through the lab and showcase some of our brilliant students and the experimental techniques they are developing.

**3:30**

**3pBA3. Biomedical Acoustics Simon Lab (BASiL) tour at penn state university.** Jacob C. Elliott (Graduate Program in Acoust., Penn State, Research West, State College, PA 16801, jce29@psu.edu), Ferdousi Sabera Rawnaque, Grace M. Wood, and Julianna C. Simon (Graduate Program in Acoust., Penn State, University Park, PA)

Welcome to the Biomedical Acoustics Simon Lab (BASiL)! Located in the newly constructed Engineering Collaborative Research and Education (ECoRE) building at Penn State, you can find the BASiLisks in their natural habitat. Our lab conducts a wide range of research at the intersection of diagnostic and therapeutic ultrasound, and as such, we have a large variety of equipment at our disposal for experimental use. In this video, we will give a tour of our brand new lab space, complete with demos of ultrasound field mapping, treatment monitoring, visualizing cavitation in cells, and more. In addition, the BASiLisks will give a brief overview of our new and exciting research developments. We are looking forward to showing you around our lab and hope that you can join us! [Ongoing work in the lab supported by NSF CAREER 1943937 and NIH R01EB032860.]

## *Invited Papers*

**3:45**

**3pBA4. Lights, camera, vaporization: Coupling acoustics and microscopy to study acoustic droplet vaporization.** Mitra Aliabouzar (Univ. of Michigan, 1301 Catherine St., 3226A Medical Science Bldg. I, Ann Arbor, MI 48109, aliabouz@umich.edu) and Mario L. Fabiilli (Univ. of Michigan, Ann Arbor, MI)

Within the Department of Radiology at the University of Michigan, we have integrated mechanics, material science, and biomedical acoustics to develop ultrasound-assisted therapies and diagnostic techniques. What is our approach? Using ultrasound-responsive droplets. These droplets phase-transition from liquid to gas bubbles when exposed to ultrasound, a process termed acoustic droplet vaporization (ADV). We harness this transformation for a wide array of biomedical applications, such as delivering therapeutic payloads, modulating hydrogel properties, and characterizing material properties. In this video, we will demonstrate how to make monodisperse droplets using microfluidic techniques and prepare tissue-mimicking hydrogels containing these droplets for optical characterization. Resolving both the ADV-bubble response in an ultrasound field at clinically relevant frequencies, as well as the associated cellular and biological responses in real-time, requires high temporal and spatial resolution. To achieve this, we couple an ultra-high-speed camera (10 million frames per second) and time-lapse confocal fluorescence microcopy (1 frame per second). We will walk you through the process: aligning optical and ultrasound foci, coupling the transducer with samples on the microscope stage, aligning laser for high-intensity back illumination, synchronizing ultrasound and the camera, and finally, triggering ultrasound to generate ADV-bubbles

**4:00**

**3pBA5. Advancing neuroscience with cutting-edge ultrasound technology.** Yu Xuan Chin (Dept. of Biomed. Eng., Washington Univ. in St. Louis, 4370 Duncan Ave., St Louis, MO 63110, chin.s@wustl.edu) and Hong Chen (Washington Univ. in St. Louis, St. Louis, MO)

Housed in one of the largest neuroscience buildings in the United States, the Chen Ultrasound Lab is at the forefront of technology innovation that bridges neuroscience with ultrasound engineering. The Chen Ultrasound Lab has pioneered technologies, including sonodelivery, sonobiopsy, and sonogenetics. Our lab has designated areas for the development, calibration, and production of ultrasound transducers, ensuring accuracy and precision for our in-house manufactured transducers. Our ability to rapidly prototype is attributable to our mature 3D printing technology. The Chen Ultrasound Lab is also equipped with preclinical and clinical ultrasound systems, including neuronavigation-guided and MR-guided focused ultrasound devices, crucial to our large animal studies with pigs and nonhuman primates as well as on-going clinical studies. Our dedicated animal surgery and behavior testing suite strengthens our research capability. Experience firsthand how our ultrasound crib contributes to pioneering discoveries in the field of neuroscience and medicine by joining us in our video tour that explores our state-of-the-art lab.

## *Contributed Paper*

**4:15**

**3pBA6. Surface acoustic wave-driven enhancement of enzyme-linked immunosorbent assays: ELISAW.** Lei Zhang (Univ. of California, San Diego, SME Room 320, 7835 Trade St., Ste. 100, San Diego, CA 92121, lez003@ucsd.edu), Yuxiong Liu (Washington Univ. in St. Louis, St. Louis, MO), Shuai Zhang (UCSD, San Diego, CA), Cecile Floer (Univ. de Lorraine, Nancy, France), Srikanth Singamaneni (Washington Univ. in St. Louis, St. Louis, MO), and James Friend (Mech. and Aerospace Eng., Univ. of California San Diego, La Jolla, CA)

Enzyme-linked immunosorbent assays (ELISAs) are widely used in biology and clinical diagnosis. Relying on antigen–antibody interaction through diffusion, the standard ELISA protocol can be time-consuming, preventing its use in rapid diagnostics. We present a time-saving and more sensitive ELISA without changing the standard setup and protocol, using surface acoustic waves (SAWs) to enhance performance. Each step of the assay is improved principally via acoustic streaming-driven advection. Using SAWs, the time required for one step of an example ELISA is reduced from 60 to 15 min to achieve the same signal intensity. By extending the duration of SAW exposure to 20 min, the binding amount can be significantly improved over the 60 min, 35 °C ELISA without SAWs. When introducing SAWs to the bead-based ELISA, the time required for binding can be further reduced to 2 min due to increasing depletion zone by acoustic streaming on beads surface. The sensitivity of ELISA can be significantly increased with lower LoD by combination of SAW stimulation and ultrabright fluorescent nanoscale labels. By significantly increasing the speed and sensitivity of ELISA, its utility may be improved for a wide range of point-of-care diagnostics applications.

THURSDAY AFTERNOON, 21 NOVEMBER 2024 — 3:00 P.M. TO 5:00 P.M.

# Session 3pCA

# Computational Acoustics and Computational Acoustics: Innovations in Computational Acoustics II

Laura C. Brill, Cochair
*Threshold Acoust., 141 W Jackson Blvd., Ste. 2080, Chicago, IL 60604*

Ralph T. Muehleisen, Cochair
*Energy Syst., Argonne National Lab., 9700 S. Cass Ave., Bldg. 362, Lemont, IL 60439-4801*

Jennifer Cooper, Cochair
*Johns Hopkins Univ., Applied Physics Lab., 11100 Johns Hopkins Rd., Mailstop 8-220, Laurel, MD 20723*

**Chair's Introduction—3:00**

## *Contributed Papers*

**3:05**

**3pCA1. Computing cochlear pressure exposures from weapon systems using free-field to cochlea transfer functions.** Courtney Mattson (Appl. Res. Assoc., 7921 Shaffer Pkwy., Littleton, CO 80127, cmattson@ara.com), Santino Cozza (Appl. Res. Assoc., Albuquerque, NM), Theodore F. Argo (Appl. Res. Assoc., Littleton, CO), and Timothy J. Walilko (Appl. Res. Assoc., Albuquerque, NM)

Overpressures generated by weapon systems can induce cochlear damage. Historically, the risk of injury has been assessed using free-field pressure waveforms. However, this approach fails to consider the modifying effects of human tissue and bone. To address this limitation, we developed transfer functions that convert free-field pressure signatures into approximations of the exposures that reach the cochlea. These transfer functions were derived from simultaneous free-field and cochlear pressure data recorded during shock tube exposures at varying peak pressures (1, 4, 8, and 12 psi) with and without hearing protection devices (HPDs). Our findings indicate that high frequencies ($> \sim$1000 Hz) are amplified by the human skull. Weapon system pressure signatures are processed using the transfer functions to approximate the intracochlear pressure differential between the scala vestibuli and scala tympani. The pressure-dependent transfer function is interpolated between the collected data based on the peak pressure of the incident waveform. The transformed data shows that the protective effects of HPDs vary based on the incident waveform. Protection tends to be effective at high frequencies, while low frequency pressures often exceed a 140 dB limit. Future studies may identify optimal HPDs for weapon operators that mitigate the harmful effects of blast exposures.

**3:20**

**3pCA2. Computational ultrasound testing for non-destructive evaluation of multilayered structures: 3D modeling of longitudinal wave propagation signal responses.** Mounir Tafkirte (Physics, Lab. of Metrology and Information Processing, Faculty of Sci., Ibn Zohr Univ., B.P 8106, Cité Dakhla, Agadir, Agadir 80000, Morocco, mounir.tafkirte@edu.uiz.ac.ma), Adil Hamine, Hicham Mesbah, and Mohamed Ettahiri (Physics, Lab. of Metrol. and Information Process., Faculty of Sci., Ibn Zohr Univ., Agadir, Morocco)

Non-destructive evaluation (NDE) using ultrasonic imaging (UI) is essential for detecting defects in complex multilayered structures, which are commonly encountered in aerospace, automotive, and medical fields. A thorough understanding of material properties, wave propagation, and dispersion behavior is crucial for accurate defect detection. The Transfer Matrix Method (TMM) provides a comprehensive approach by modeling the entire UI process, including the control system, ultrasonic transmitter, wave propagation (both bulk and guided modes), and receiver response. This study focuses on predicting the backscattered ultrasonic signal during longitudinal wave propagation in a multilayer structure immersed in water, considering normal incidence and specific frequencies. TMM, employing a quadrupole formalism, integrates stress and velocities at layer boundaries and models the multilayer structure as a transfer matrix derived from individual layer matrices. This approach allows for the calculation of reflection coefficients across a wide frequency spectrum. TMM generates detailed distance-time and distance-frequency representations that illustrate the propagation of various longitudinal modes in configurations, such as plexiglass/water/glass under direct (PWG) and reverse (GWP) insonation. Comparisons between PWG and GWP distance-time planes may be affected by layer arrangement and properties, which influence the reflection coefficient, highlighting the system's sensitivity to layer order even with similar thicknesses and materials.

**3:35**

**3pCA3. Geometry of 3D ray field and pulse propagation.** Nikolai E. Maltsev (Retired, 1467 Leaftree Cir, San Jose, CA 95131, nick_e_maltsev@yahoo.com)

Field computations in ray approximation perform mapping in 6D space of ray parameters (a,b,t) and spatial coordinates (x,y,z), where (a,b,t) are launch angles in the source and t-eikonal. In the case of 2 spatial dimensions, mapping occurs in 4D space. This work investigates properties of such mappings, which allow a robust search of eigen-rays and modeling of pulse propagation

**3:50**

**3pCA4. Shaping sound field through acoustic metasurfaces: An optimisation method for correcting loudspeaker emissions.** Letizia Chisari (Informatics, Univ. of Sussex, Falmer, Brighton BN1 9RH, United Kingdom, l.chisari@sussex.ac.uk), Andy Philippides (Informatics, Univ. of Sussex, Brighton, United Kingdom), and Gianluca Memoli (Metasonixx Ltd., Brighton, United Kingdom)

In a loudspeaker system, the acoustic wave front modulation can be achieved using active control systems that adopt algorithms to modulate the acoustic wave front emitted, adapting to the environmental conditions or to the specific requirements of the sound source. To be efficient, a large number of loudspeakers and the digital processing of the signal are required. Acoustic metasurfaces are sub-wavelength structures that can manipulate sound, allowing the control of, e.g., the shape or direction of the wave front. In this work, we aim to reduce the number of speakers by coupling acoustic metasurfaces at the outlet of a single transducer. To design the metasurface for this task direct methods, which rely on unit cells geometrical parameters retrieved by finite element simulations, are traditionally used. This approach has to be repeated iteratively and demands significant efforts to investigate the enormous number of possible metamaterial combinations. More recent methods include topology optimization, physics-informed neural networks, and machine learning. These methods frequently limit their focus on optimization for unit cell structures and do not provide a design or strategy for the whole metasurface. In this work, the metasurface phase profile is instead retrieved by combining a finite element method to generate the search space and a gradient-free optimization algorithm to retrieve the best metasurface configuration required to satisfy a desired pressure distribution pattern in a 2D listening area.

**4:05–5:00 Discussion**

# Session 3pEA

## Engineering Acoustics: Frugal Acoustics: Panel Discussion

Randall P. Williams, Cochair
*Ctr. for Indus. and Med. Ultrasound, Univ. of Washington, 1013 40th Ave. NE, Seattle, WA 98105*

Luz D. Sotelo, Cochair
*Purdue Univ., 2550 Northwestern Ave., 1900D, West Lafayette, IN 47906*

### *Contributed Papers*

**3:00**

**3pEA1. Low-cost low-power underwater acoustic data acquisition and processing system.** Julian Blanco, Richard L. Campbell (Appl. Ocean Sci., North Stonington, CT), and Christopher M. Verlinden (Appl. Ocean Sci., 9500 Gilman Dr., La Jolla, CA 92093-0701, cmverlin@ucsd.edu)

There is a need for low-cost acoustic recording and processing instruments. Cheaply available and highly capable off-the-shelf microcontrollers and analog-to-digital converters (ADC's) allow rapid prototyping of novel systems to achieve minimum performance requirements without overhead. Combined with low-cost manufacturing capabilities for pre-assembled printed circuit boards (PCBs), acoustic recording hardware and processing have become more accessible. Prototypes can be designed and delivered for less than \$100 USD per iteration. Using this methodology, we developed the Self Contained Acoustic Recording Instrument (SCARI). While consuming less than ~40 mW, two channels of 48 ksps acoustic data can be recorded to .wav files on a microSD card with a SAMD51 microcontroller, an SGTL5000 ADC, and differential pre-amplifier on a compact PCB. Data can be saved to a microSD card or streamed via USB to be saved/processed by an embedded computer, with resources left over for simple real-time processing.

**3:12**

**3pEA2. A low-cost ultrasonic absorption spectrometer for liquids.** Michelle R. Crouse (Chem. & Biochem., California State Univ., Dominguez Hills, 2012 Bataan Rd., A, Redondo Beach, CA 90278, mruth.sci@gmail.com), Małgorzata Musiał (Mater. Meas. Lab., National Inst. of Standards and Technol., Boulder, CO), Jacob Pawlik, Bryan Bosworth, Nathan Orloff, Aaron Hagerstrom (Commun. Technol. Lab., National Inst. of Standards and Technol., Boulder, CO), Jason A. Widegren (Mater. Meas. Lab., National Inst. of Standards and Technol., Boulder, CO), Angela C. Stelson, and Robert Lirette (Commun. Technol. Lab., National Inst. of Standards and Technol., Boulder, CO)

Ultrasonic absorption spectroscopy can probe intermolecular interactions that inform research into processes for chemical engineering and pharmaceuticals. The only commercial ultrasonic spectrometer costs over one hundred thousand dollars, putting it out of reach for many institutions. We designed an inexpensive ultrasonic absorption spectrometer comprised of off-the-shelf components and parts from a rapid prototyping service. We employed the through-transmission method for measuring absorption, using thirty-one pulse measurements at varying distances. These measurements significantly improved measurement accuracy when compared to both a fixed path technique and the pulse-echo method that rely on only two measurements. We measured pure water to correct for diffraction effects and used repeated water measurements to propagate the uncertainty. To validate the spectrometer, we measured salt solutions, polyvinyl alcohols, and celluloses. We then compared the data to results obtained from a commercial spectrometer and they were well within the margin of error for both devices. Notably from these measurements, we observed a relaxation peak near 1 MHz in scandium sulfate, which was only reported once before with the use of a resonance method. The system we present here offers an inexpensive alternative to a commercial ultrasonic absorption spectrometer that is accessible to university researchers and students.

**3:24**

**3pEA3. Here's my two cents… Low-cost, acoustically transparent containers allow quantification of cavitation energy *in vitro*.** Darcy M. Dunn-Lawless (Inst. of Biomed. Eng., Univ. of Oxford, Botnar Inst. for Musculoskeletal Sci., Windmill Rd., Oxford OX3 7LD, United Kingdom, darcy.dunn-lawless@magd.ox.ac.uk), Abigail Collins, Constantin Coussios, and Michael Gray (Inst. of Biomed. Eng., Univ. of Oxford, Oxford, United Kingdom)

As cavitation-based therapies continue to enter the clinic, there is growing demand for methods to quantify the energy released by cavitating bubbles. Passive Acoustic Mapping (PAM) can reconstruct the energy and distribution of cavitation from multi-sensor recordings of bubble emissions, but its accuracy is impaired *in vitro* by the aberrating presence of a sample container between the cavitating media (nuclei/cells/tissues) and detectors. To our knowledge, the effects of these vessels on PAM have never been studied. Additionally, the typical need for sterility and a large number of samples makes low cost essential for sample containers in cavitation experiments. Here, we characterize the effects of common laboratory vessels in the range 3–14 MHz via an acoustic reciprocity experiment, then describe the design and testing of a novel container with improved acoustic transparency. The new device reduced worst-case magnitude and phase errors by 13.7 dB and 6.6 radians respectively, compared to ordinary 2 ml centrifuge tubes. We will also present quantitative measures of container effects on PAM energy measurement and localization. The new containers are manufactured from 100 micron polymer film via vacuum forming, are quick and easy to make in any shape in a normal laboratory, and cost US\$0.02 each.

**3:36**

**3pEA4. An inexpensive teaching laboratory apparatus for measuring the speed of sound and more.** Andrew C. Morrison (Natural Science, Joliet Junior College, 1215 Houbolt Dr., Joliet, IL 60431, amorriso@jjc.edu)

One of the laboratory experiments in our general education acoustics class is a measurement of the speed of sound activity. The apparatus for this activity consists of a pair of ultrasonic transducers, a signal generator, and an oscilloscope. Each transducer is mounted on a custom base that is designed to rest on top of a standard meter stick which lays flat on the lab bench. The bases are arranged on the meter stick such that the transducers face each other. One of the transducers is driven by the signal generator at its resonance frequency, creating a sound wave which is detected by the other transducer. By monitoring the relative phase of the transmitted and detected sound wave signals, an accurate measurement of the wavelength of

the sound can be obtained. From the wavelength measurement and the drive frequency students easily calculate the speed of sound in the room. With the advent of 3D printing and the availability of inexpensive signal generators and oscilloscopes, this apparatus is affordable not only for colleges but also to many high schools. Furthermore, the apparatus can be also used to explore other wave phenomena, such as reflection, absorption, and interference.

**3:48–5:00 Discussion**

THURSDAY AFTERNOON, 21 NOVEMBER 2024 3:00 P.M. TO 5:00 P.M.

## Session 3pED

## Education in Acoustics: Where Do You Get Your Inspiration?

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

**Chair's Introduction—3:00**

### *Contributed Papers*

**3:05**

**3pED1. Training for fitting hearing protection devices.** William J. Murphy (Stephenson and Stephenson, Research and Consulting, 5706 State Rte. 132, Batavia, OH 45103, wmurphy@sasrac.com)

Hearing protection devices, particularly formable protectors, can be difficult to fit into the ear canal. Research in hearing protector fit testing has begun to focus on methods that can facilitate training for earplug users. The graph that has inspired my research was was created following a field study with oil and gas workers. The graph indicates the attenuation workers achieved before training versus after training. The graph has been used by other hearing science researchers.

**3:10**

**3pED2. Applying physics education research to teaching.** Jack A. Dostal (Dept. of Phys., Wake Forest Univ., P.O. Box 7507, Winston Salem, NC 27109, dostalja@wfu.edu)

A fundamental paper that guides my thinking about teaching is Lillian McDermott's 2001 Oersted Medal Lecture entitled "Physics Education Research - The Key to Student Learning." It was published in AJP and given at the American Association of Physics Teachers meeting that year. It highlights the value of taking a scientific approach to improve student learning. It also demonstrates the benefits of active engagement in the classroom and the limits of traditional instruction. Teaching effectively can be trained; it is more than just an art. McDermott highlights core principles of physics education research and puts those principles in context of physics examples (simple electric circuits and single/double slit diffraction). Her phrase, "Teaching by telling is an ineffective mode of instruction for most students," is a mantra that I have always kept in mind when making decisions about how to proceed in a course.

**3:15**

**3pED3. Syllabus—More than just a piece of paper with course rules.** Daniel A. Russell (Graduate Program in Acoust., Pennsylvania State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, dar119@psu.edu)

I have never liked "syllabus day"—the first day of the semester when I'm supposed to go over the syllabus that students never read. Several years ago, I began writing course syllabi as newsletters, complete with color and images, and instead of discussing the syllabus in class I assign a quiz that requires students to read and engage with the syllabus content. When I saw the book "*Syllabus: the Remarkable, Unremarkable Document That Changes Everything*" by Germano and Nicholls (2020), I picked it up and read it, thinking it would give me a few ideas for making my syllabus documents even better. However, to my surprise, this book was much more than tips for getting students to read a boring course contract. Instead it challenged me to re-think how I design an entire semester-long course in light of what I want my students to have learned by the end of the semester and what I need to do in order to get them there. In this lightning-round talk I'll summarize a few of the main takeaways that I have adopted since reading "*Syllabus*."

**3:20**

**3pED4. Case study: Developing an acoustics program in Ecuador Through Capstone Projects.** Carlos Yoong (Wood PLC, 2020 Winston Park Dr. #700, Oakville, Ontario L6H6X7, Canada, carlos.yoong@woodplc.com), Galo Durazno, Guido Abril, and Fausto Maldonado Galarza (Escuela Superior Politécnica del Litoral, Guayaquil, Ecuador)

The Coastal Polytechnic School in Guayaquil, Ecuador is the leading engineering school in the country. Recent efforts have been targeted towards the creation of an acoustics program that enables students apply basic acoustics knowledge in real-life problems that the Faculty of Mechanical Engineering currently has. With the lack of courses in acoustics, various capstone projects have been conducted by the students under the supervision of experts in acoustics. Most of the basic training in acoustics theory is given during the first weeks of the project and then the student is exposed to field work by taking measurements utilizing different test procedures to obtain acoustical metrics. Then, a subsequent analysis is performed under the supervision of various experts. These capstone projects will be the stepping stone for the first programs in industrial acoustics in the City of Guayaquil and will help newly graduates understand the importance of acoustics throughout their careers. The end goal is to have a well-established program with relevant undergraduate courses in acoustics.

**3:25–5:00 Impromptu Contributions**

## Session 3pMU

## Musical Acoustics: General Panel Discussion on Music Acoustics Topics

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

### *Contributed Paper*

**3:00**

**3pMUa1. General panel discussion on musical acoustics topics.** Andrew A. Piacsek (Physics, Central Washington Univ., 400 E. University Way, Ellensburg, WA 98926-7422, andy.piacsek@cwu.edu)

A round-table discussion with presenters from both General Topics sessions in Musical Acoustics. This session provides an opportunity for participants to ask detailed follow-up questions and to stimulate discussion on a variety of topics.

THURSDAY AFTERNOON, 21 NOVEMBER 2024 3:00 P.M. TO 5:00 P.M.

## Session 3pNS

## Noise: A World of Vehicle Noise

Jeanine Botta, Cochair
*The Quiet Coalition, 720 East 31st St., Apartment 7G, Brooklyn, NY 11210*

Daniel Fink, Cochair
*none, The Quiet Coalition, P.O. Box 533, Lincoln, MA 01733*

**Chair's Introduction—3:00**

### *Invited Papers*

**3:05**

**3pNS1. An update on horn-based vehicle signaling, and newer sound effects for convenience, nostalgia, and entertainment.** Jeanine Botta (The Quiet Coalition, 720 East 31st St., Apt. 7G, Brooklyn, NY 11210, jeaninebotta@aol.com) and Daniel Fink (The Quiet Coalition, Concord, MA)

A presentation at the 2013 International Congress on Acoustics explored vehicle horn-based lock alert use and the conflicts it created among neighbors. From 2007 to 2012, more than 60% of vehicles manufactured for the North American market left factories with horn-based alerts to signal locking and unlocking. Most were set to honk at the second press of the key fob's lock button, although settings could be changed to flashing lights or no audible or visual alert. Automakers addressed questions about legality in user manuals, stating that owners should observe local laws. By 2013, internal combustion engine vehicles had become quieter, and electric and hybrid vehicles seemed to offer a chance to reduce noise further. Instead, automakers adopted horn-based features in electric and hybrid vehicles, and added sounds for convenience, nostalgia, and entertainment, using external speakers. Reviewing auto industry data, noise ordinances, federal, state, and local regulations, and case law, we examine legal and regulatory aspects of non-emergency horn signals and vehicle sound effects.

**3:20**

**3pNS2. Feeling heard: Conflict resolution in vehicle noise disputes.** Tricia Schafer (Tricia Schafer Dispute Resolution PLLC, 6929 N Hayden Rd., Suite C4-199, Scottsdale, AZ 85250, tricia@vitalitylaw.com)

Neighborhood noise disputes often arise due to differing lifestyles, varying schedules, or simply different tolerances to noise. While these conflicts can strain relationships and disrupt daily life, mediation provides a constructive way to resolve them amicably. With regard to automobile noise, such as audible remote keyless entry, what might seem trivial to one person can be deeply distressing to another, leading to frustration, stress, and even health issues over time. Addressing these concerns promptly and fairly is crucial to maintaining a harmonious neighborhood environment. Mediation is a process designed to facilitate constructive dialogue and negotiation between parties in conflict. This paper will emphasize strategies to bring disputing parties to the table, how to prepare for a mediation, selection of a mediator, and what to expect during the mediation process.

**3:35**

**3pNS3. Lessons from US noise camera pilot programs as an alternative to direct policing. Author Tanya Marie Bonner.** Tanya M. Bonner (WaHI-Inwood Task Force on Noise, 560 West 165th St., Apt. 913, New York, NY 10032, tanya.bonner@gmail.com)

An analysis of the four cities with the only existing noise camera pilot programs in the United States was conducted to determine the effectiveness of the programs (reduction in noise complaints, number of citations issued, meeting of program target goals), how well it addresses excessive noise pollution caused by vehicles, and the impact on vulnerable communities (indications of greater presence and enforcement in minority communities, and support from local communities). Methods of analysis included collecting data on the use of, general geographic locations of, and outcomes of the pilot programs since their inception. Qualitative data were also gathered from community residents, environmental health experts, community activists, and city officials to assess the implications of expanding such programs as a form of hands-off policing.

**3:50**

**3pNS4. Quiet Florida.** Mary L. Tatigian (none, 4360 7th Ave. SW, Naples, FL 34119, mtatigian@hotmail.com)

Quiet Florida There are laws and regulations governing noise in states, counties, cities, and towns. Without community action and engagement with elected representatives, however, there may not be enforcement of existing laws. That was the situation in Collier County, Florida, when a group of citizens became concerned about the adverse health effects of aviation noise, and noise from illegally modified vehicle exhausts in and around the resort town of Naples. Creating an organization called Quiet Florida in March 2021, citizens reached out to elected officials, got on the meeting agenda, and informed the county commissioners of the negative health impacts of noise pollution. The next step was the formation of the Southwest Florida Noise Task Force, comprised of a county commissioner, the Collier County Sheriff Department, Naples City Police, the Florida Highway Patrol, the county Transportation Management Department, Florida Legislative Aide, and Quiet Florida. The Task Force is sending letters to 300 automotive repair shops and car dealerships, and using print, broadcast, and social media to inform businesses and the public of the negative health impacts of transportation noise and that modified exhaust systems are illegal. A detailed discussion of Quiet Florida's efforts will be presented. Mary Tatigian Quiet Florida

**4:05**

**3pNS5. Conducting an orchestra to reduce vehicle noise.** Ingrid Buday (Advocacy, No More Noise Toronto, 105 The Queensway, Ste. 2110, Toronto, Ontario M6S 5B5, Canada, ingrid.buday@gmail.com)

As with conducting an orchestra, directing the "sections" of vehicle noise pollution advocacy and hitting the right notes can be a challenge. Noise comes from a variety of sources and can be perceived differently by different people. And while the conductor tries to reconcile incompatible voices, the notes of regulation and legislation can be heard in the background, further complicating the piece. In Canada, No More Noise Toronto has initiated change at City Hall in a very short time. The grassroots campaign has been addressing high-volume vehicle noise by collecting data, critiquing existing processes, and encouraging Toronto citizens to participate, which has not been done previously. Using meters and crowdsourced methods, No More Noise Toronto seeks to understand noise "from the bedroom window," which validates citizens' noise reporting stories. Reviewing findings related to processes that lead to inaccurate data, barriers to involvement in the civic engagement process, and challenges created by frustration and feelings of apathy, we discuss actions taken and practices put into place as we've promoted a collaborative approach with stakeholders to find solutions and common ground in the shared goal of protecting the health of Toronto citizens.

## *Contributed Papers*

**4:20**

**3pNS6. Mitigation of flow-acoustic coupling across circumferential quarter-wave resonators within an intake silencer.** Pranav Sriganesh (Dept. of Mech. and Aerospace Eng., The Ohio State Univ., 930 Kinnear Rd., Columbus, OH 43212, sriganesh.1@osu.edu) and Ahmet Selamet (Dept. of Mech. and Aerospace Eng., The Ohio State Univ., Columbus, OH)

An acoustic resonator subjected to mean flow across its opening is susceptible to high-amplitude tonal noise generation. This noise source is established by flow-acoustic coupling, a phenomenon where, under certain flow conditions, shear layer instabilities at the opening of a resonator couple with acoustic resonances, leading to the periodic formation of strong vortices. The current work explores methods to suppress flow-acoustic coupling across circumferential quarter-wave resonators within a unique silencer designed to attenuate high-frequency noise from turbocharger compressors. Experiments conducted on a flow bench with the baseline silencer reveal multiple tones due to flow-acoustic coupling across the quarter-wave resonators of different lengths. Three-dimensional computational fluid dynamics

is first employed to study the mechanism of noise generation due to flow-acoustic coupling. Having successfully predicted the acoustic signature of the baseline silencer, the computational study is then extended to investigate the use of flow deflectors at the opening of each resonator to mitigate flow-acoustic coupling. Finally, the improved design is fabricated and evaluated experimentally on the flow bench to verify the suppression of the undesirable tones.

**4:35**

**3pNS7. Directivity patterns for road traffic auralization.** Christian Dreier (Inst. for Hearing Technol. and Acoust., RWTH Aachen Univ., Kopernikusstrasse 5, Aachen 52074, Germany, christian.dreier@akustik.rwth-aachen.de), Simon Kamps, and Michael Vorlaender (Inst. for Hearing Technol. and Acoust., RWTH Aachen Univ., Aachen, Germany)

Auralizations enable the simulation of sound fields in complex environments. For example, they can be used to investigate the perceptual aspects of sound emitted by vehicles. As input for the signal processing chain, sound source models must adequately represent the acoustic spatial and spectro-temporal characteristics of real vehicle emissions. An *in-situ* measurement method for determining the velocity-dependent directionality of moving sound sources is presented. The work presents results for different vehicle types. Finally, the implementation of the directional characteristics into an urban sound auralization model is discussed in order to simulate the sound transmission in anurban environment including complex propagation effects.

**4:50–5:00 Discussion**

THURSDAY AFTERNOON, 21 NOVEMBER 2024 — 3:00 P.M. TO 4:35 P.M.

## Session 3pPP

## Psychological and Physiological Acoustics: Psychological and Physiological Acoustics II

Andrew J. Oxenham, Chair
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**Chair's Introduction—3:00**

### *Contributed Papers*

**3:05**

**3pPP1. Assessing voice familiarity in normal-hearing adults—Preliminary results.** Fuh-Cherng Jeng (Hearing, Speech, and Language Sci., Ohio Univ., 1 Ohio University Dr., Grover Ctr. W231, Athens, OH 45701, jeng@ohio.edu), Maddie R. Porter, Molly G. Taylor, Chao-Yang Lee, Sydney W. Bauer, and Kalyn McDonald (Hearing, Speech, and Language Sci., Ohio Univ., Athens, OH)

The ability to recognize familiar voices is crucial for effective communication, with infants showing early sensitivity to their mother's voices. Despite extensive research in auditory perception, understanding how normal-hearing adults distinguish between familiar and unfamiliar voices remains limited. This study aimed to investigate the behavioral responses of normal-hearing adults to familiar versus unfamiliar voices. Building upon existing literature and relevant models, we hypothesized that reaction times would correlate with individuals' ability to recognize familiar versus unfamiliar voices and that shorter reaction times would occur when additional speech cues were embedded in the acoustic stimulation. Eleven adults participated, exhibiting perfect accuracy in identifying their mother's voices and rejecting voices of female strangers. Reaction times were significantly shorter ($\chi^2 = 8.985$, $p = 0.003$) when participants responded to their mother's voices compared to those of female strangers, suggesting a cognitive advantage in processing familiar voices. Additionally, reaction times were significantly shorter ($\chi^2 = 19.032$, $p < 0.001$) for disyllable words compared to monosyllables, indicating differential processing mechanisms for linguistic stimuli. Our study contributes to understanding the cognitive processes involved in voice recognition and speaker identification. Clinical implications include potential interventions for voice recognition disorders and strategies for memory enhancement.

**3:15**

**3pPP2. Speech-on-speech masking by voice-gender differences and spatial separation cues in simulated single-sided deaf listeners.** Caroline Cuthbertson (Dept. of Otolaryngol., HNS and Communicative Disorders, Univ. of Louisville, 529 S. Jackson St., Ste. 416, Louisville, KY 40202, caroline.cuthbertson@louisville.edu), Joesphine Kinder, Phillip Friggle, and Yonghee Oh (Dept. of Otolaryngol., HNS and Commun. Disorders, Univ. of Louisville, Louisville, KY)

Many previous studies have reported that a listener's binaural sensitivity can influence speech segregation performance in multi-talker environments. This study aimed to investigate how monaural hearing changes the masking release performance, especially by voice-gender differences and spatial separation between talkers. Speech recognition thresholds were measured in twenty normal-hearing listeners with simulated single-sided deafness. Target speech was fixed in front (0°), and two maskers were presented either symmetrically or asymmetrically (toward the healthy or deafened ear) in four spatial configurations (maskers at 0°, 30°, 60°, or 90°). The measurements were repeated in both same-gender and different-gender target-masker combinations. The results showed that the maximum masking release (15-20 dB with equal weighting between voice and spatial cues) occurred at $> 60°$ target-maskers separation when the maskers were

presented in the deafened ear. For both symmetric maskers and the maskers in healthy ear conditions, 5 to 12 dB of voice-difference benefits were exhibited, but no spatial benefit was observed (-5-0 dB SRM). The findings in this study suggest that listeners with single-sided deafness could benefit from speech-on-speech masking with the use of both voice and spatial cues when the masker talkers were spatially separated from the target talker, toward the deafened ear.

**3:25**

**3pPP3. Effects of spatial asymmetry and voice-gender differences between talkers on speech-on-speech masking performance in adults with normal hearing.** Joesphine Kinder (Dept. of Otolaryngol., HNS and Commun. Disorders, Univ. of Louisville, 529 S. Jackson St., Ste. 416, Louisville, KY 40202, josephine.kinder@louisville.edu), Phillip Friggle, Caroline Cuthbertson, and Yonghee Oh (Dept. of Otolaryngol., HNS and Commun. Disorders, Univ. of Louisville, Louisville, KY)

Speech segregation can be improved by talkers' spatial separation, known as spatial release from masking (SRM). This study aimed to investigate how a listener's SRM performance is affected by spatial asymmetry and voice-gender differences between talkers. With twenty young normal-hearing adults, speech-on-speech masking performance was measured as the threshold target-to-masker ratio (TMR) needed to understand a target talker in the presence of masker talkers. Target speech was fixed in front (0°), and two maskers were presented either symmetrically or asymmetrically (toward the right or left) in four spatial configurations (maskers at 0°, 30°, 60°, or 90°). The measurements were repeated in both same-gender and different-gender target-masker conditions. The amounts of SRM were computed from the TMR thresholds for each condition. The results showed that the SRM was increased by up to 10 dB when the same-gender maskers were symmetrically separated from the target, and the SRM was improved by up to 20 dB in the asymmetric target-masker condition; however, these spatial benefits were reduced by 10–15 dB when the voice-gender difference cue was presented. The findings in this study suggest that the ability to use spatial cues for speech segregation can be influenced by both target-masker asymmetry and voice-gender differences.

**3:35**

**3pPP4. Effects of a low-intensity two-talker speech masker on real-time spoken word recognition processes in children with normal hearing.** Kelsey Klein (House Inst. Foundation, 1127 Wilshire Blvd., Ste. 1620, Los Angeles, CA 90017, kklein@hifla.org)

To comprehend speech, listeners much efficiently initiate lexical access (i.e., recognize word form), activate semantics (i.e., recognize word meaning), and suppress lexical competition. This study used a Visual World Paradigm eye-tracking task to examine how the presence of a low-intensity two-talker speech masker may disrupt these real-time processes in 19 10- to 12-year-old children with normal hearing. Each child completed the task (1) in quiet and (2) with the masker presented at a level individually determined to ensure high accuracy (mean SNR = 4.5 dB). Although accuracy was high while listening with the masker (mean accuracy >95%), children showed slower lexical access and semantic activation, relative to listening in quiet. Speed of semantic activation in the masker condition was not predicted by lexical access speed, suggesting that delayed semantic activation while listening to masked speech is not simply the result of delayed lexical access; rather, the presence of a two-talker speech masker may disrupt children's ability to process word meaning. Participants also showed increased and prolonged lexical competition when listening with the masker, suggesting that a low-intensity masker may reduce children's ability to resolve competition and commit fully to the target word, even when they ultimately recognize the word accurately.

**3:45**

**3pPP5. Lyric intelligibility of musical segments for older individuals with hearing loss.** William M. Whitmer (Hearing Sci. - Scottish Section, Level 3, New Lister Bldg., Glasgow Royal Infirmary, Glasgow G31 2ER, United Kingdom, bill.whitmer@nottingham.ac.uk), David McShefferty (Hearing Sci. - Scottish Section, Glasgow, United Kingdom), Michael A. Akeroyd (School of Med., Univ. of Nottingham, Nottingham, United Kingdom), Scott Bannister (School of Music, Univ. of Leeds, Leeds, United Kingdom), Jon P. Barker (Speech and Hearing Research Group, Univ. of Sheffield, Sheffield, United Kingdom), Trevor J. Cox, Gerardo Roa, Bruno Fazenda (Acoust. Res. Ctr., Univ. of Salford, Salford, United Kingdom), Jennifer L. Firth (School of Med., Univ. of Nottingham, Nottingham, United Kingdom), Simone Graetzer (Acoust. Res. Ctr., Univ. of Salford, Salford, United Kingdom), Alinka Greasley (School of Music, Univ. of Leeds, Leeds, United Kingdom), and Rebecca Vos (Acoust. Res. Ctr., Univ. of Salford, Salford, United Kingdom)

Understanding lyrics is a major barrier to enjoying music for people with hearing loss. To improve lyric understanding through machine-learning, metrics need to be informed by the experiences of the target population. Currently, there are no data on lyric-recall ability of older individuals with hearing loss. Twelve older participants with mostly mild-sloping hearing loss listened to and recalled 100 segments of popular music that varied in genre, duration and word count. In each trial, participants heard a randomly chosen segment twice with a 5-s interstimulus interval over headphones at an A-weighted level of 65 dB plus individualised frequency-dependent non-linear gain. The proportion of words heard correctly varied greatly across samples, from 0 to 100%, and varied as a function of genre, similar to past studies with different populations. Individual intelligibility across samples was correlated with age; sample intelligibility across individuals, however, was not correlated with word count or rate. Improving sung-lyric intelligibility is a different challenge from spoken-speech enhancement for those with hearing loss. Not only does the enhancement need to be considered within the overall enjoyment of the music, but the variation in results—as seen in the current study—reflects variations in genre, orchestration and vocal quality in sung music.

**3:55**

**3pPP6. The role of genre association in Sung Dialect categorization.** Maddy Walter (Univ. of British Columbia, 2613 West Mall, Vancouver, British Columbia V6T 1Z4, Canada, maddyw37@student.ubc.ca), Sydney Norris, Sabrina Luk, Marcell Maitinsky, Jahurul Islam, and Bryan Gick (Linguistics, Univ. of British Columbia, Vancouver, British Columbia, Canada)

Genre-associated sociocultural cues may influence dialect recognition when listening to music. Previous work identifies genre-specific sociolects [Coupland, 2011, J. Socioling. 15]; singers make genre-dependent production changes [Gibson, 2019, U.Canterbury Diss.]; and accent is perceived with greater ease in song compared to speech [Mageau *et al*., 2019, Phonetica 76]. However, the role of genre associations in dialect categorization has not been sufficiently addressed. Our previous work suggests that genre identification may play a larger role in this than auditory speech cues [Walter *et al*., 2023, J. Acoust. Soc. Am. 154]. We investigate this further using a dialect-identification task with improved "spoken" stimuli. Participants heard sung clips (original instrumental-free vocals) and manipulated "spoken" clips (instrumental-free vocals, monotonized and rhythm-normalized) from counterbalanced genres with greater ease of vocal isolation: U.S. folk with some intra-genre dialect variation, and blues with strong associations to African American English [De Timmerman *et al*., 2024, J. Socioling. 28]. A superior vocal remover and lyrical transcripts were implemented. As predicted, in the sung context, participants identified genre-associated dialects consistently for both genres, but slightly more so for blues. Varied responses to the "spoken" context supports that listeners use genre information in musical dialect perception.

**4:05**

**3pPP7. Evaluating the relationship between musical sophistication and melodic and harmonic expectations.** Anya E. Shorey (Psychol. and Brain Sci., Univ. of Louisville, 317 Life Sciences Bldg., Louisville, KY 40292, anya.shorey@louisville.edu) and Christian E. Stilp (Psychol. and Brain Sci., Univ. of Louisville, Louisville, KY)

Expectations are a fundamental and key factor in producing emotional responses to music; even musically untrained listeners demonstrate musical expectations. This is seen in priming tasks, in which musicians' and nonmusicians' categorizations of final tones are faster and/or more accurate when they are expected and related to a musical context (e.g., tonic) than when they are less related. Alternatively, in production tasks, participants sing/play what they expect to occur next in a sequence. This approach is limited to participants who can accurately generate and produce a response, usually those with some musical training. Both priming and production tasks typically compare musicians and nonmusicians without considering their musical backgrounds on a continuous scale or musical engagement beyond formal training. Here, participants completed an adapted melodic or harmonic production task (with multiple-choice responses, facilitating nonmusicians' responses) and a priming task for the same sequences. Participants' musical backgrounds freely varied. For production, related targets were predicted to be chosen more often than other targets, with increasing musical sophistication yielding more related responses. For priming, related targets were predicted to be categorized faster and more accurately than other targets, with increasing musical sophistication lessening this difference. Preliminary results offer partial support for these predictions.

**4:15**

**3pPP8. Searching for subcortical indications of interaction between dynamic pitch and timbre cues via the frequency-following response.** Ryan Anderson (Speech and Hearing Sci., Indiana Univ., 200 S Jordan, Bloomington, IN 47401, anderyan@indiana.edu), Yi Shen (Speech and Hearing Sciences, University of Washington, Seattle, WA), and William P. Shofner (Speech and Hearing Sciences, Indiana University, Bloomington, IN)

Pitch and timbre cues, specifically the fundamental frequency (F0) and spectral centroid (SC), respectively, interact in listeners' perception. For example, changes in SC can influence the perceived pitch. F0 and SC appear to interact in the auditory cortex. We investigated whether these cues interact in the auditory brainstem, via frequency-following responses (FFRs). Four dynamic stimuli were constructed with either congruent (both ascending or descending) or incongruent movements of F0 and SC across their 400-ms duration. The nominal values for F0 and SC were 150 and 750 Hz, and the ranges of F0 and SC movements corresponded to 10 times the just noticeable difference. Dynamically varying F0s were encoded faithfully in the FFRs, independent of cue congruency. This suggests a lack of interaction between F0 and SC in FFR. Furthermore, although the two congruent stimuli were time-reverse of each other, the cross-correlation function between FFRs to the two stimuli exhibited temporal asymmetry for some participants. Similar asymmetry was also observed for FFRs to incongruent stimuli. The asymmetry in cross-correlations of FFRs was not observed for the corresponding cross-correlations of the stimuli. This suggests a temporal asymmetry in the stimulus-encoding processes of the subcortical auditory system, which is heterogenous across listeners.

**4:25**

**3pPP9. Phase coherence of spontaneous otoacoustic emissions derived from a cochlear model.** Vaclav Vencovsky (Dept. of Radio Eng., Czech Tech. Univ. in Prague, Technicka 2, Prague 16627, Czechia, vaclav.vencovsky@gmail.com) and Christopher Bergevin (Phys. & Astronomy, York Univ., Toronto, Ontario, Canada)

Spontaneous otoacoustic emissions (SOAEs) are commonly presented by averaging spectral magnitudes calculated in adjacent time segments of the recorded signal. To include the information carried by SOAE phase, Bergevin *et al.* [under preparation] proposed calculating the phase difference between adjacent time segments and then determine the vector strength to compute a measure of phase coherence. In stronger human SOAE peaks (magnitude $>0$ dB SPL), phase coherence plotted as a function of segment duration usually shows a maximum at around 100 ms, which shortens with increasing SOAE frequency above $\sim$2 kHz. Similar bell shaped dependencies coherence can be observed for simulated SOAEs from a nonlinear cochlear model composed of fluid-coupled oscillators, which simulate the transversal displacement of the basilar membrane (BM) [Vencovský *et al.* JASA, 2020]. The simulated BM displacement is amplified by feedback undamping force, which is transformed by a sigmoidal nonlinearity proportional to the second-order Boltzmann function. To evoke SOAEs, irregularities are added into the undamping feedback force and the model is driven by broad band noise applied into the eardrum (external noise) and into the undamping feedback force (internal noise). It appears that a combination of external and internal noise is needed to obtain the bell-shaped dependence of phase coherence on the segment duration similar to the one observed in human SOAEs. These results help guide how stochastic considerations must be included into active cochlear models.

## Session 3pSA

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

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Robert M. Koch, Cochair
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### *Contributed Papers*

**3:00**

**3pSA1. Transient processes in self-leveling pneumatic vibration isolators.** Vyacheslav M. Ryaboy (Photonics Solutions, MKS, 1791 Deere Ave., Irvine, CA 92606, vryaboy@newport.com)

Pneumatic vibration isolators have found wide applications in precision vibration control. They combine low stiffness with high load capacity, which makes them ideal for maintaining low-noise environments necessary for cutting-edge scientific experiments and precise manufacturing. The low stiffness makes them less suitable for applications involving transients such as fast-moving stages and robotic loading-unloading processes; therefore, research of limiting performance and optimization of pneumatic isolators under transient loads is very desirable. Whereas dynamics of pneumatic isolators in a stationary environment in the frequency domain have been thoroughly studied in the literature, modeling and optimization of transient processes in the time domain did not attract adequate attention. This paper presents linear models of single-chamber and dual-chamber pneumatic vibration isolators with self-leveling via mechanical feedback. Time histories under typical initial disturbances are presented. Analytical and numerical solutions for optimal gains in the sense of fastest settling after disturbance, and stability conditions are derived.

**3:15**

**3pSA2. Comparative study of higher modes of vibration in cantilever beams: Exact analytical analysis versus FEM analysis for accordion free reed acoustics.** Giovanni Volpatti (Harmonicum, Mattenstrasse 21, Port 2562, Switzerland, giovanni.volpatti@harmonicum.ch)

This study delves into the critical role of higher vibration modes in cantilever beams, focusing on their significance for the acoustics of accordion free reeds. Through a comparative analysis, we explore exact analytical methods versus Finite Element Method (FEM) simulations to understand the vibrations not only parallel to the airflow direction but also perpendicular and torsional modes. Previous research has underscored the importance of these higher modes in shaping the timbre and influencing attack transients. Our findings reveal that while analytical solutions provide precise insights into fundamental vibrations, FEM offers comprehensive data on complex mode interactions. This dual approach enhances the understanding of reed dynamics, offering valuable insights for musical instrument design and acoustic optimization. By bridging analytical rigor with computational flexibility, this paper contributes to the nuanced study of musical timbre and transient behaviors, laying the groundwork for future innovations in reed instrument engineering.

**3:30**

**3pSA3. Design of optimal discriminant using air-induced structural vibration for groundline health assessment of wooden utility poles.** Yishi Lee (Metropolitan State Univ. of Denver, 1449 7th St., Denver, CO 80204, Denver, CO 80204, ylee24@msudenver.edu)

Wooden utility poles are the critical components of power distribution in the United States and many parts of the world. Due to its essential societal function, comprehensive structural health monitoring technology is lacking to reflect its importance. To address this concern, the study exploits the phenomenon of air-induced oscillation due to the vortex street developed in the wake of the pole. It extracts the features of this vibration and designs an optimal classifier that can infer the health condition around the groundline region based on oscillational acceleration data. The proposed methodology consists of the following steps: (1) Construct a transient and modal orthotropic finite element model with realistic soil conditions to analyze the vibrational properties of a class 2 55-foot wooden utility pole. (2) Produce temporal vibrational responses from periodic disturbance winds of different harmonic conditions from a given transmission cable configuration. (3) Explore feature extraction techniques and design an optimal discriminant function for structural health classification. This presentation will detail the techniques and examine the preliminary efficacy of the proposed discriminant function using empirical data. Its broader impact is to examine the technical feasibility of using structural vibration to determine the groundline health condition.

**3:45**

**3pSA4. Echoes of impact: Unraveling the sonic boom.** Toni Wangui (Ventura College, 4667 Telegraph Rd., Admissions and Record, Ventura, CA 93003, toni_wangui1@my.vcccd.edu)

This study examines the impact of sonic booms on urban communities and architectural structures, specifically focusing on how these intense sound waves interact with buildings. Sonic booms, typically generated by aircraft exceeding the speed of sound, pose potential risks and disturbances to structural integrity and community well-being. To better understand these interactions in detail, this study employs advanced ray tracing methods, a technique used in acoustics to model how sound waves propagate through different environments. A few modifications and additions were made thorough python to the existing ray tracing code that allowed for a more precise simulation of the waves' behavior when encountering buildings. The study's findings aim to provide deeper insight into the mitigation strategies that can be employed to protect structures and/or build better aircraft that can minimize sonic boom or better yet produce sonic thump.

**4:00**

**3pSA5. Nondestructive evaluation of PLA cubes with varying cooling conditions.** Partha Pratim Pandit (Mech. Eng., Purdue Univ., 1901 Union St., Lafayette, IN 47904, pandit18@purdue.edu), Anna Keim, Meher Mirza, Harshith Kumar Adepu, Justin Yoosung Kim, Monique McClain (Mech. Eng., Purdue Univ., West Lafayette, IN), and Luz D. Sotelo (Purdue Univ., West Lafayette, IN)

Fused Filament Fabrication (FFF) is one of the most extensively used additive manufacturing (AM) methods for the manufacturing of thermoplastic polymers. However, there are still many problems related to the quality of the printed parts. Most of the current research work for improving the part quality is limited to optimizing process parameters and using imaging techniques to analyze the part quality. Due to lack of consistency of quality across printers and printed parts, these efforts are insufficient to characterize and predict local material properties within the parts. Ultrasonic evaluation techniques are becoming increasingly recognized in assessing parts created by AM. In this work, Polylactic Acid (PLA) cubes were printed with varying fan activation layers. Ultrasonic and X-ray CT testing are employed to analyze and detect the controllable and observable deformation due to the change in *in-situ* cooling condition within the cubes. At the same time, mechanical properties and defect content were evaluated. It is important to understand the relationship between the fan cooling and mechanical properties so that effective fan cooling strategies can be deployed for consistent fabrication of higher quality parts across printers.

**4:15**

**3pSA6. Investigation of the multifunctional properties of a bio-inspired material based on cuttlebone.** Kara Hardy (Michigan Technol. Univ., Houghton, MI, khardy@mtu.edu) and Bhisham Sharma (Mech. Eng. - Eng. Mech., Michigan Technol. Univ., Houghton, MI)

Nature is a never-ending inspiration for new structures with unique multifunctional properties. For example, the structure of the cuttlebone of the common cuttlefish, Sepia Officinalis, allows it to withstand pressures up to 2 MPa while enabling the flow of water in and out to control buoyancy. Cuttlebone is a lightweight material with a porosity of around 90% and is made of many layers of labyrinthine walls separated by thin septa. Under compression, each layer fails successively, leaving the remaining layers functional. The semi-sinusoidal labyrinthine pattern of the walls is reminiscent of space-coiling structures seen in acoustic metamaterials, making the structure intriguing from the perspective of sound absorption as well. In this paper, we explore the feasibility of using additive manufacturing to produce structures inspired by the multi-layered cuttlebone. After producing samples, we test their acoustical properties using an impedance tube. In addition, we test their mechanical properties. Preliminary results suggest that these bio-inspired structures can be tailored for noise control applications.

**4:30**

**3pSA7. Acoustic characteristics of Parasaurolophus crest: Experimental results from a physical model.** Hongjun Lin (Music and Performing Arts Professions, NYU, 82 Washington Sq. E, New York City, NY 10003, hl5642@nyu.edu)

This study investigates the acoustic characteristics of the Parasaurolophus crest through the construction and experimental analysis of a simplified anatomical model. Motivated by the potential correlation between the crest and sound production and perception, a model was developed consisting of two open pipes connected at specific points to mimic the crest's structure. Inspired by previous models of resonating chambers, the experimental setup involved exciting the model with a small speaker, collecting data via a minimally invasive microphone, and suspending the model with cotton threads to minimize energy dissipation. Verification experiments ensured the stability and repeatability of the setup, while a control group comprising a simple open pipe was used for calibration. Frequency sweeps generated response data, revealing a fundamental frequency of 800 Hz, with peaks present at its multiples. Some peaks were shifted compared to the control group, notably at 1300 and 5100 Hz, with the experimental group displaying non-linear behavior at specific frequency ranges. The results suggest that the crest structure could have functioned as a resonating chamber, supporting the hypothesis that it played a role in sound production and perception, potentially facilitating communication among Parasaurolophus individuals.

**4:45**

**3pSA8. Modal analysis of a magnetorheological membrane.** Seyyedmohammad Aghamiri (Mech. Eng., Concordia Univ., 1455 De Maisonneuve Blvd. W., Montreal, Quebec H3G 1M8, Canada, seyyedmohammad.aghamiri@mail.concordia.ca) and Ramin Sedaghati (Mech. Eng., Concordia Univ., Montreal, Quebec, Canada)

Magnetorheological elastomers (MREs) are smart composite materials that can change their viscoelastic properties and provide actuation under application of an external magnetic field. In this study, a finite element method is developed in COMSOL Multiphysics to investigate the vibration characteristics of a membrane made of MREs under external applied magnetic field. The aim is to understand the effect of thickness, radius, and boundary conditions of the MRE membrane on its resonance frequencies and mode shapes under varied applied magnetic field. Results will offer valuable insight into how adaptive MRE-based acoustic membranes and metamaterials can be designed and tuned for broad-band noise suppression.

## Session 3pSC

## Speech Communication: The State of the Art in Speech Communication

Benjamin V. Tucker, Chair
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### *Invited Paper*

**3:00**

**3pSC1. The acoustics of voice.** Jody Kreiman (Head and Neck Surgery, UCLA, 1000 Veteran Av., 31-19 Rehab Ctr., Los Angeles, CA 90095-1794, jkreiman@ucla.edu)

Recent decades have seen rapid growth in our understanding of the production and perception of the human voice and of the acoustic features that critically link production and perception in communication. This review describes some of the many acoustic measures of voice (and voice quality) that have been proposed to quantify these features, with attention to major changes over the years in what has been measured and why. Linkages between acoustic measurement systems and voice production and perception will be explored, and the current state of the art will be reviewed.

**3:45–4:00 Discussion**

**4:00**

**3pSC2. Interpreting acoustic measures of imitation in word shadowing.** Cynthia G. Clopper (Ohio State Univ., 1712 Neil Ave., Columbus, OH 43210, clopper.1@osu.edu)

Word shadowing tasks, in which a participant repeats words aloud after a pre-recorded model talker in a laboratory setting, elicit imitation of the model talker's speech by the shadowing participant, even in the absence of explicit instructions to imitate. However, the nature of this imitation varies substantially within and across shadowers, model talkers, and linguistic features. Recent research has considered this variation in imitation in word shadowing to address how acoustic measures of imitation in these tasks should be interpreted. The results demonstrate (1) imitation of phonetic and phonological structure in vowel shifts and intonation contours, in addition to direct imitation of acoustic formant frequencies and f0; (2) more imitation of non-stereotyped social variants than of stereotyped social variants; and (3) both convergence to overall production patterns and synchrony with token-by-token variability in the speech of the model talker. Together, these results suggest a strong speech perception-production link that operates in real time and is mediated by linguistic and social representations.

**4:45–5:00 Discussion**

## Session 3pSP

# Signal Processing in Acoustics: Signal Processing Potpourri II

John R. Buck, Cochair
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James C. Preisig, Cochair
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**Chair's Introduction—3:00**

## *Contributed Papers*

**3:05**

**3pSP1. Use of Schwarz-ox direct-to-reverberant power ratio estimator for distance estimation near diffuse noise point in a complex plane.** Chuyang Wang (Mech. Eng., Hongkong Polytechnic Univ., Hung Hom, Kowloon, Hong Kong, chuyang.wang@connect.polyu.hk), Wai Yin Mung, Karhang Chu, and Tracy Y. Choy (Mech. Eng., Hongkong Polytechnic Univ., Hong Kong, Hong Kong)

This study investigated the time- and frequency- dependent direct-to-reverberant power ratio (DRR) estimators from spatial coherence value of omnidirectional microphone array. We use a mathematical 2D geometry interpretation, complex plane, to analysis the deficiency of different DRR estimators bias and construct a better estimator near the diffuse noise point in the same plane. The diffuse noise point is a reflection of a certain distance in practical environment, where the received signals in microphone array are ideal diffuse. This estimator was then applied to distance estimation in a reverberant environment. DRR values are known to exhibit more effective performance in the case of distance estimation than other acoustic indicators in such environment; however, several proposed DRR estimators are not sufficiently robust near diffuse noise points in the complex plane. Therefore, this study proposes a method to accommodate the existing Schwarz DRR estimator to exhibit better performance around the point where the far-distance estimation is made. For validation, our results focused on different directions and inter-microphone distances in two reverberant classrooms. In addition, through an in-depth investigation of the proposed method, we found that it could be expanded into a more freedom-based method, which researchers can manipulate to draw the DRR estimator pattern in the complex plane as required.

**3:15**

**3pSP2. Moving source 2D-hologram in the presence of intensive internal waves in SWARM-95 experiment.** Mohsen Badiey (Elec. and Comput. Eng., Univ. of Delaware, 139 The Green, Room 140, Evans Hall Room 140, Newark, DE 19716, badiey@udel.edu), Sergey A. Pereselkov (Math. Phys. and Inf. Technol., Voronezh State Univ., Voronezh, Russian Federation), and Venedikt Kuz'kin (Sci. Ctr. for Wave Res., General Phys. Inst., Moscow, Russian Federation)

This paper presents the holographic signal processing results from an acoustical oceanography experiment on the New Jersey continental shelf called SWARM-95. This experiment that was conducted in summer of 1995 aimed at propagation of broadband acoustic signals in the presence of intensive internal waves (IIW). Impulsive acoustic signals were from a moving source. The IIW caused modes coupling and significant 3D acoustic effects (Badiey M. *et al.*, JASA, 117(2): 613–625). The current research is based on holographic signal processing (Pereselkov S., Kuz'kin V., JASA, 151(2): 666- 676) utilizing holographic signal processing. Here, the sound intensity distributions in frequency-time coordinates, called the source interferogram, is analyzed using a 2D Fourier transform (2D-FT). The result of the 2D-FT is called the Fourier-hologram (hologram). Two distinct sets of spectral spots were identified in the hologram of the moving source. The first set corresponds to the sound field in an unperturbed waveguide (without IIW). The second set relates to the hydrodynamic perturbation of the sound field caused by IIW. Such a hologram structure allows for the reconstruction of the parameters of the moving source. Results of this research can be used for detection and localization of Autonomous Underwater Vehicles (AUV). [This research was supported by a grant from the Russian Science Foundation (grant number 23-61-10024).]

**3:25**

**3pSP3. Enunciation—An important factor in speech-to-text medical transcription systems.** Bozena Kostek (Audio Acoustics Lab., Gdansk Univ. of Technol., Narutowicza 11/12, Gdansk 80-233, Poland, bokostek@audioakustyka.org)

This study aims to explore the extent to which enunciation plays a crucial role in a speech-to-text (STT) system, especially when dealing with medical terminology. To achieve this, an audio dataset was recorded containing Polish medical terms and spoken diagnoses pronounced by healthcare professionals, including general practitioners and specialists in various fields such as cardiology, pulmonology, and radiology. The next step involved comprehensive acoustical and lexical analyses of the audio recordings. Features such as harmonic-to-noise ratio, spectral tilt, zero-crossing rate, formant dispersion, jitter, and shimmer were considered. Moreover, a transformer-based ASR (Automatic Speech Recognition) model was engaged in speech-to-text transcription. Several speech quality evaluation measures were used, such as WER (Word Error Rate), MER (Match Error Rate), WIL (Word Information Loss), WIP (Word Information Preserved), CER (Character Error Rate), etc. By measuring the STT model's quality, it was possible to analyze the correlation between acoustical features and the expression style, as well as the speaker's distinctive vocabulary choices when reading acronyms. [Work supported by the Polish National Center for Research and Development (NCBR) project: "ADMEDVOICE-Adaptive intelligent speech processing system of medical personnel with the structuring of test results and support of therapeutic process," no. INFOSTRATEG4/0003/2022.]

**3:35**

**3pSP4. Advancing emotion recognition of audio: Testing and verifying machine learning techniques for speech and music.** Diana Gomez-Fonseca (None, 4539 Basswood Dr., Erie, PA 16506, dianacgomezfonseca@gmail.com)

The aim of this research is to investigate machine learning techniques for emotion recognition for audio such as human speech and music.

Emotion recognition technology has a history in commercial applications to enhance personalized music recommendations, mood-based ambience detection and interpret human interactions or human-machine interaction such as job interviews, caller-agent calls, streaming videos, and music platforms such as Spotify. Moreover, enhancing these algorithms can significantly benefit individuals on the autism spectrum by promoting accurate and practical support. In this study, we employed a combination of techniques to build a machine learning approach for emotion recognition. By adjusting the audio features and given test and train data, we aimed to identify and enhance the relationships between audio perception and its features. This approach seeks to eventually improve the accuracy and applicability of emotion recognition systems, contributing to the ongoing development and promotion of this technology in various domains.

**3:45**

**3pSP5. The added value of radio-acoustic virtual environment.** Gabriel Ouaknine-Beaulieu (École de technologie supérieure, Univ. du Québec, 1100 Notre-Dame St. W, Montreal, Quebec H3C 1K3, Canada, gabriel.ouaknine-beaulieu.1@ens.etsmtl.ca), Xinyi Zhang, Jeremie Voix, Rachel Bouserhal, and Pascal Giard (École de technologie supérieure, Univ. du Québec, Montreal, Quebec, Canada)

As hearing protection devices (HPDs) also attenuate voice, workers tend to remove them to talk to each other, disabling their protection. Radio-acoustic virtual environment (RAVE) proposes an ideal situation, where workers can communicate in noisy environments while being protected. The voice is recorded with in-ear microphones (IEM), denoised, and transmitted within a certain communication radius determined by the talker's vocal effort. The audio is played at a comfortable level with directionality. Many articles addressed elements necessary for RAVE, such as voice activity detection [N. Lezzoum *et al.*, in IEEE JCE, 2014, pp. 737–744], wearer induce disturbances detection [F. Bonnet *et al.*, in JERGON, 2019, pp. 102862], and communication radius [R. Bouserhal *et al.*, in JSLHR, 2017, pp. 3393–3403]. This pioneering research integrated them and tested a mock-up version of RAVE in live scenarios. In groups of three, twenty-one participants completed manual tasks that required communication. They used RAVE's mock-up and a broadcasting push-to-talk device, with and without noise presence, totaling four scenarios. Participants completed a questionnaire after each scenario, and their speech and motion were recorded. This research contributed with a mock-up version of RAVE with improved signal treatment for real-time purposes. Our results confirm RAVE's added value and illustrates its potential.

**3:55–4:00 Break**

**4:00**

**3pSP6. Estimating the power law exponent of shear waves in soft tissue with a time-domain parametric model.** Robert J. McGough (Elec. and Comput. Eng., Michigan State Univ., 428 S. Shaw Lane, Room 2120, East Lansing, MI 48824, mcgough@egr.msu.edu) and Matthew W. Urban (Dept. of Radiol., Mayo Clinic, Rochester, MN)

In soft tissue, the attenuation of shear waves satisfies a power law. However, the values of power law exponents for shear waves in various soft tissues remain unknown due to windowing artifacts that limit the effectiveness of present methods that are based on two-dimensional fast Fourier transforms. To solve this problem, a time-domain signal processing approach is proposed. Robust shear wave parameter estimates are obtained by minimizing the least-squares error between measured time-domain shear wave data and stable probability density functions that describe the solution to a closely related fractional-order diffusion equation. Assessments are performed with measured shear wave data collected from *ex vivo* pig liver using a Verasonics system with an L7-4 linear array transducer. Power law exponents are obtained from measured shear wave data at the focal depth between 1.2 and 13.9 mm from the center of the push beam. The mean estimated power law exponent for these locations is 0.74, and the standard deviation is 0.05. The results suggest that fractional calculus models that support power law exponents less than one are needed for shear waves in pig liver.

**4:10**

**3pSP7. In-ear acoustic monitoring for swallow detection.** Elyes Ben Cheikh (Elec. Eng., École de technologie supérieure, 1100 R. Notre Dame O, Montréal, QC H3C 1K3, Canada, elyes.ben-cheikh.1@ens.etsmtl.ca), Imane Hocine (Univ. of Montreal, Montréal, Quebec, Canada), Alessandro Braga, Arian Shamei (Elec. Eng., École de technologie supérieure, Montreal, Quebec, Canada), Ingrid Verduyckt (Univ. of Montreal, Montreal, Quebec, Canada), Catherine Laporte, and Rachel Bouserhal (Elec. Eng., École de technologie supérieure, Montréal, Quebec, Canada)

Sialorrhea, or involuntary drooling, affects up to 25% of patients with neurodegenerative diseases, causing significant discomfort. Current treatments are mostly invasive. Non-invasive treatment may be possible by providing patients with reminders to swallow based on their swallowing frequency. A promising approach involves detecting swallowing using in-ear devices. To enable such treatment, we are collecting a database of 13 non-verbal human-produced events such as blinking, coughing, grinding teeth, etc. These events are captured by various sensors inside the ear, including microphones, air pressure microphones, PPG sensors, and IMUs, in different acoustic environments, both quiet and noisy. Additionally, we use ultrasound videos of the tongue as a ground truth for spontaneous swallowing. We will use this database to train machine learning algorithms for detecting and classifying swallowing and other non-verbal events. This work will enable the development of health monitoring in-ear devices for vulnerable individuals.

**4:20**

**3pSP8. Framework for multiclass vocal pathology classification using convolution neural network, continuous wavelet transform, and signal-to-noise ratio analysis.** Bhawna Rathi (Music Technol., IUPUI, 535 W. Michigan St., IT371, Indianapolis, IN 46202, brathi@iu.edu) and Timothy Hsu (Music and Arts Technol., Indiana Univ. - Purdue Univ., Indianapolis, Indianapolis, IN)

Vocal disorders are potentially an underreported condition due to invasive diagnostic methods and a lack of general awareness. Traditionally, vocal damage is diagnosed through laryngoscopies, but recent research supports using audio processing and machine learning to distinguish between not only healthy and unhealthy voices but also between different vocal pathologies. This study introduces a framework for classifying multiple vocal pathologies—dysphonia, polyps, nodules, and paralysis, based on previous research focused on two-class classification. The current approach employs continuous wavelet transforms (CWT) and spectrograms from pathological voices, /AH/ and /EE/ audio files from the Indiana University (IU) Health Voice Center, as inputs for a convolutional neural network (CNN) classifier. The images from the wavelet transform achieve higher accuracy than spectrograms for both two-class and multiclass classification. Various wavelet shapes and sizes are compared for accuracy rates and processing time. Different signal-to-noise ratios and augmentation techniques are tested for classification robustness and accuracy. The study also compares male and female voices on classification accuracy and how parameters, like learning rates and filter layers, affect CNN performance. Results indicate that accuracy improves with a 0.0001 learning rate and optimal image sizes, demonstrating the potential for enhancing classification models through precise parameter tuning.

**4:30**

**3pSP9. An optimal control point method for cross talk cancellation.** Sipei Zhao (Centre for Audio, Acoust. and Vibration, Univ. of Technology Sydney, 32-34 Lord St., UTS Tech Lab, Botany, New South Wales 2019, Australia, sipei.zhao@uts.edu.au), Yang Huang, and Jing Lu (Key Lab. of Modern Acoust., Nanjing Univ., Nanjing, China)

Spatial audio is critical for improving immersive experience in various virtual and augmented reality applications. Cross-talk cancellation (CTC) is required to deliver binaural audio through loudspeakers to people's two ears discretely. Various CTC methods have been widely studied in literature, among which the optimal source distribution (OSD) is a popular one. By analyzing the singular values of the plat matrix, the OSD method chooses the loudspeaker positions that lead to an optimally conditioned system. Theoretically, an infinite number of continuously distributed sound sources are

required in the OSD method, but in practical implementations, several pairs of discrete loudspeakers are used, with each pair controlling sound in a frequency band. Instead of changing the sound source positions in the OSD method, this paper proposes an alternative optimal control point distribution (OCPD) approach using a single pair of loudspeakers at fixed locations, where the control point can move freely. Theoretical analysis shows that although the control points are away from the evaluation points (two ears), the performance of the CTC system can be dramatically improved as long as the control points are properly chosen. Numerical simulations demonstrate that the proposed OCPD approach can achieve similar performance to the OSD method with only two loudspeakers.

**4:40**

**3pSP10. Interpreting user-generated recordings from the Trump assassination attempt on July 13, 2024.** Robert C. Maher (Elec. & Comput. Eng., Montana State Univ., 610 Cobleigh Hall, P.O. Box 173780, Bozeman, MT 59717, rmaher@montana.edu)

Handheld smartphones are common at public events, and this increases the likelihood that user-generated recordings (UGRs) may capture sounds of interest to audio forensic investigations. The assassination attempt against presidential candidate Donald Trump that took place in Bulter, PA, on Saturday, July 13, 2024, is a prominent example, with a dozen or more official and user-generated videos of the incident. Audio forensic expertise is called upon to determine the physical meaning of the recorded material and to ascertain its authenticity. In the Trump rally shooting incident, multiple concurrent recordings were made from different positions around the scene, capturing the sounds of several audible gunshots. The set of multiple UGR recordings provides information useful to the forensic investigation, such as spatial information about the number, type, position, and orientation of firearms. Examination of the multiple recordings from the Trump rally incident requires interpretation of unsynchronized audio from locations that are not precisely known. Nevertheless, acoustical information obtained from user-generated material at the Trump incident provides essential corroborative information. Examples are presented showing the audio forensic analysis principles applied to the UGR evidence.

THURSDAY AFTERNOON, 21 NOVEMBER 2024 — 3:00 P.M. TO 5:00 P.M.

## Session 3pUW

## Underwater Acoustics: Trivia Among AO/AB/UW

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

**Chair's Introduction—3:00**

### *Invited Paper*

**3:05**

**3pUW1. Trivia competition among technical committees: AO/AB/UW.** Robert W. Drinnan (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Box 355640, Seattle, WA 98105-6785, drinnan@uw.edu)

Join for an exciting trivia competition among the three ocean-related technical committees: Acoustical Oceanography, Animal Bioacoustics, and Underwater Acoustics. These teams will go head-to-head, answering skill-testing questions on acoustics, marine science, and general ocean-themed trivia. Open to all, this contest session a fun and educational experience with questions designed at a high-undergraduate to beginning graduate level.

# AUTHOR INDEX

to papers presented at

## 187th Meeting of the Acoustical Society of America