Session 4aAA

Architectural Acoustics: Acoustics in Mass Timber

Evelyn Way, Cochair
Maxxon Corp, 920 Hamel Road, Hamel, MN 55340

David Manley, Cochair
DLR Group, 6457 Frances St., Omaha, NE 68106

Chair's Introduction—8:00

Invited Papers

8:05

4aAA1. The project design includes a mass timber structure. Now what? Jessica S. Clements (Acoust. Studio, Newcomb & Boyd, LLP, 303 Peachtree Ctr. Av NE, Ste. 525, Atlanta, GA 30303, jclements@newcomb-boyd.com)

Mass timber comes with any number of acoustical design challenges depending on the project type and use. The many advantages in sustainability and cost mean that mass timber design is here to stay. As an acoustical consultant what does this mean for your project? This presentation will review some of the typical concerns you might have or questions to consider asking of the design team. We will review some typical pitfalls, touch on available resources, and discuss a few potential solutions to consider.

8:25

4aAA2. Design considerations for mass timber in commercial buildings. David Manley (DLR Group, 6457 Frances St., Omaha, NE 68106, dmanley@dlrgroup.com)

Mass timber commercial buildings can significantly shift the building and design industry to a more sustainable path, but integrating architectural, engineering, and acoustic systems within this relatively new superstructure presents many challenges for designers. This talk will discuss DLR Group's integrated design approach and mass timber research including architectural design, structures, and acoustics.

8:45

4aAA3. Listening to the structure: Mass-timber construction at the Groton Hill Music Center. Carl P. Giegold (Threshold Acoust. LLC, 2622 Grant St., Evanston, IL 60201, cgiegold@thresholdacoustics.com), Laurie Kamper, Brandon Cudequest, and Matt Skarha (Threshold Acoust. LLC, Chicago, IL)

Historically, Western music spaces were built with massive masonry-bearing walls. While these designs offer structural and acoustical benefits, today's construction culture is conscious of more fluid forms and embodied carbon, which influences how consultants think about the design of spaces for music. The recently completed Groton Hill Music Center is unique for its use of mass timber's sculptural capabilities to shape performance space acoustic volumes forthrightly, with little additional finish treatment. The center features studio classrooms for students of all ages, an orchestral rehearsal space, a 300-seat recital hall for soloists and small ensembles, and a grand 1000-seat concert hall that opens to view for a 500-seat lawn audience. Structure and architecture are largely unified, with one's acoustic experience defined by the sculpted mass timber structure. The comparatively light-weight timber/concrete composite structure represents a significant and somewhat acoustically risky reduction in mass to provide an acoustically warm environment. This presentation will explore how the structure was optimized to provide stiffness and multi-scale diffusion, taking the ideals of classic concert hall design, and embracing contemporary construction technologies and acoustical analyses.

9:05

4aAA4. Mass timber acoustic design: Data informed design. Denis Blount (Acoust., Audioviusal, Arup, 1191 Second Ave., Ste. 400, Seattle, WA 98101, denis.blount@arup.com) and Ben Loshin (Acoust., Audioviusal, Arup, Seattle, WA)

Acoustic isolation is a fundamental challenge for mass timber construction, and solutions that work are critical for decarbonizing the built environment. While a broad range of acoustic laboratory test data exists for mass timber assemblies, field test data and flanking paths data are scarce. This talk will cover motivation for leveraging mass timber as a construction material (i.e. sustainability, embodied carbon, biophilic design, etc.), and inherent acoustic challenges. Additionally, it will address Arup's current practices and lessons learned from project work and research collaborations. The information presented will be drawn from project case studies, field test flanking data, and analytical studies aimed at developing the next generation of mass timber products and solutions. Case studies will focus on building types where mass timber has posed significant acoustics and vibration design challenges (e.g., healthcare, modular).

Analytical studies include novel inventions to solve vibration and room acoustic challenges. Field test data includes flanking field tests validated against numerical models.

9:25

4aAA5. Prediction of sound and vibration levels from heavy-hard impacts in a cross laminated timber—Concrete hybrid building. Joshua Brophy (Architectural Acoust., Acentech, 33 Moulton St., Cambridge, MA 02138, jbrophy@acentech.com) and Aedan Callaghan (Pliteq Inc., Toronto, ON, Canada)

Heavy-hard weight impacts in fitness spaces often result in elevated noise and vibration levels, which can lead to complaints in sensitive adjacencies. Previous works have demonstrated a method to predict the resulting noise and vibration levels in fitness space adjacencies. This method involves the analysis of an *in situ* transfer function in combination with laboratory force data and sound pressure level measurements from a shotput dropped on a calibrated reference rubber impact sheet. It was originally developed for use in concrete structures, which are typical for large multi-residential, commercial, and mixed-use buildings that commonly feature fitness amenity rooms or commercial gyms. The growing popularity of mass timber as a sustainable building material provides an opportunity to expand upon the previous research by examining the validity of the prediction method in another structural type. This paper serves as a case study of heavy-hard impacts in a residential hybrid mass timber structure, where weight drops from a fitness center at the top level of the concrete podium were disturbing apartment residents occupying the first CLT level directly above.

9:45–10:00 Break

10:00

4aAA6. Impact sound insulation performance of mass timber floors. Jianhui Zhou (School of Engineeing, Univ. of Northern BC, 499 George St., Wood Innovation and Design Ctr., Prince George, BC V2L 1R5, Canada, jianhui.zhou@unbc.ca) and Mohammad Hossein A. Jafari (School of Engineeing, Univ. of Northern BC, Prince George, BC, Canada)

Mass timber floors are being used increasingly in both mass timber and hybrid timber buildings due to its dry and rapid construction. Bare mass timber structural slabs have relatively low impact sound insulation performance. Though certain floating floor assemblies on mass timber slabs can provide adequate single number ratings, such assemblies are mainly effective in the middle to high frequency range. This paper will provide an overview of the research on the impact sound insulation perforamnce of mass timber floors with exposed ceiling and floating floor assembies conducted at the Universtiy of Northern British Columbia. The test data of bare mass timber floors, with continous floating concrete toppings, with raised discrete floating floor assemblies will be presented and compared. The effect of interlayer, concrete thickness and excitation sources on the single number ratings and frequency spectra will be discussed. Recommendations for future reseach will be given based on the discussion.

10:20

4aAA7. Vibroacoustic performance of a mass timber cassette floor through mock-up tests. Jianhui Zhou (School of Engineeing, Univ. of Northern BC, 499 George St., Wood Innovation and Design Ctr., Prince George, BC V2L 1R5, Canada, jianhui.zhou@unbc.ca), Chenyue Guo, Mohammad Hossein A. Jafari (School of Engineeing, Univ. of Northern BC, Prince George, BC, Canada), and Brant York (Intelligent City Inc., Vancouver, BC, Canada)

Though cross laminated timber slab floors are being used increasingly in mass timber construction, solid mass timber slab floors are limited to short-to-medium span applications. The sound insulation performance of solid mass timber slab floors is often achieved through assemblies on the floor surface as exposed wood ceilings are often prefered by achitects and occupants. A mass timber cassete floor system has been recently designed and tested for its structural, vibrational and acoustical performance. This paper reports the vibroacoustic performance of the proprietary floor system through mock-up building tests. In particular, the natural frequency and mode shapes of the floor under 200 Hz was measured by experimental modal tests. The effect of floating concrete topping on the dynamic properties of the floor was assessed by experimental modal testing. The impact sound insulation performance of the bare cassette floor was first measured, and then with floating concrete toppings. The airborne sound insulation performance of a partition wall was measured to reveal the effect of flanking sound casued by the exposed continous mass timber ceiling. Detailed results will be presented in the full paper.

10:40

4aAA8. Use of a micro acoustic chamber for rapid iterative design, teaching, and demonstration in the development of innovative lower-carbon mass timber assemblies. Mark Fretz (Architecture, Univ. of Oregon, 70 NW Couch St., 105A White Stag Bldg., Portland, OR 97209, mfretz@uoregon.edu), Jason Stenson (Architecture, Univ. of Oregon, Portland, OR), and Dale Northcutt (Architecture, Univ. of Oregon, Eugene, OR)

Mass timber construction offers a range of benefits to the design industry, including the potential for sustainability and reduced building embodied carbon, biophilia, speed of assembly, and prefabrication. However, these benefits can create acoustical challenges for sound isolation between spaces. Thus, the University of Oregon's Institute for Health in the Built Environment, in collaboration with the TallWood Design Institute, has developed a micro acoustic chamber as a conceptual testbed, educational resource, and demonstration tool for design professionals and students, to develop new and innovative mass timber acoustic assemblies, understand acoustic implications of material choices in composite form, and comparatively analyze the performance of acoustic detailing decisions more rapidly at a small scale. The chamber hosts specimens that are 813 mm $\times$ 813 mm and up to 300 mm thick, far smaller than a typical laboratory acoustic floor/ceiling assembly test chamber, creating known limitations; however, test samples can be cost and material efficient, allowing relative performance differences between assemblies to be evaluated quickly and easily for initial design feedback. This presentation will share how the use of a micro acoustic chamber has been integrated into design education, student, and industry pilot research to accelerate the development of mass timber acoustic solutions.

11:00

4aAA9. Cross laminated timber geometry and joinery implications on low-frequency sound transmission: FEA and Laser Doppler Vibrometry studies. Tomás I. Méndez Echenagucia (Univ. of Washington, 3950 University Way NE, Gould Hall, Seattle, WA 98105, tmendeze@uw.edu), Andrew A. Piacsek (Phys., Central Washington Univ., Ellensburg, WA), and Nicolaas B. Roozen (KU Leuven, Leuven, Belgium)

Cross Laminated Timber (CLT) slabs are becoming more widespread in North American construction in large part because of the environmental advantage of lower embodied carbon when compared to concrete and steel slabs. This advantage is significantly reduced by the use of high embodied carbon layers in the mass timber floor assemblies, such as concrete or gypsum screeds, that are used to reduce the direct sound transmission. This paper presents a study on the potential to reduce low-frequency sound transmission in cross-laminated timber floors by means of geometric stiffness. A laser doppler vibrometry (LDV) experiment on two different geometries is presented. One sample has a flat geometry, the other has a folded plate design. The folded plate sample is built with the same CLT panels used on the flat one, but is made out of 12 custom parts assembled into a domical shape. Both samples were installed on a 6×7 foot glulam frame, subjected to shaker excitations in the low frequencies on their top side, and the vibration on their bottom sides measured with the LDV. The resulting data is used to calibrate an FEA model, and subsequently study the sound transmission implications of the sample's boundary conditions and joinery.

11:20

4aAA10. Acoustic implications of thin mass timber panels. Evelyn Way (Maxxon Corp, 920 Hamel Rd., Hamel, MN 55340, eway@maxxon.com)

As mass timber construction becomes more widely adopted, there is an incentive to use thinner, lighter panels. Research sponsored by the American Wood Council and WoodWorks explores the relative performance of 5-ply CLT and 5-ply equivalent panels to 3-ply CLT and 3-ply equivalent panels both bare and with topping slabs. In addition to the comparison of thicker and thinner panels, assemblies using resilient mats and gypsum concrete topping slabs were also developed to meet the IBC residential code minimum performance. Results and acoustic analysis will be presented along with a discussion of the non-acoustic considerations of the move to thinner mass timber structural panels.

11:40

4aAA11. Calculation of sound transmission parameters for wood-frame and mass timber assemblies. Jason Smart (American Wood Council, 74 Driftwood Court, Heathsville, VA 22473, jsmart@awc.org)

While much of it is proprietary, a significant amount of the data from sound transmission tests performed on mass timber assemblies has been shared in the public domain. Previous efforts at using this data to develop empirical models for estimating single-number ratings of floor assemblies have been limited in scope and have not always yielded accurate results. In order to provide an additional means of demonstrating compliance with code requirements, the American Wood Council (AWC) has developed calculation-based analysis approaches for deriving STC and IIC ratings for assemblies constructed with wood-based framing and mass timber. As required by the International Building Code (IBC), these analysis procedures are based on comparisons of data from floor/ceiling assemblies having STC and IIC ratings as determined by the test procedures set forth in ASTM E90 and ASTM E492, respectively. The models presented in this report can be used to estimate STC and IIC ratings of: a.) light-frame floor/ceiling assemblies framed with either sawn lumber, wood I-joists, or wood trusses and b) mass timber floor/ceiling assemblies constructed with either cross-laminated timber (CLT), dowel-laminated timber (DLT), or veneer-based mass timber. Descriptions of the models are provided. Validation and application of the models are also discussed.

Session 4aAB

Animal Bioacoustics: Acoustic Ecology, Biological Soundscapes, and Animal Vocal Communication and Physiology II

Laura Kloepper, Chair
Department of Biological Sciences, University of New Hampshire, 230 Spaulding Hall, Durham, NH 03824

Contributed Papers

8:30

4aAB1. An open-source acoustic detector for beluga whales, with evaluations in the Western Canadian Arctic. Fabio Frazao (Comput. Sci., Dalhousie Univ., 6050 University Ave. Halifax, NS B3H 1W5, Canada, fsfrazao@dal.ca), Marie-Ana Mikus, Valeria Vergara (Raincoast Conservation Foundation, Sidney, BC, Canada), William D. Halliday (Wildlife Conservation Society Canada, Whitehorse, YT, Canada), and Mike Dowd (Mathematics & Statistics, Dalhousie Univ., Halifax, NS, Canada)

Belugas (*Delphinapterus leucas*) face threats from various sources, including noise from oil and gas exploration, vessel traffic, and other human activities. Here we present the development of a deep learning-based acoustic detector to automatically detect the species, and measure its performance when applied to study sites in the western Canadian Arctic. We used over 20,000 individually annotated beluga vocalizations to train deep learning models in the binary task of classifying 3-second audio clips into containing beluga vocalizations or not. Approximately 7,000 annotated vocalizations were reserved for testing, and models were evaluated on their ability to correctly label audio clips of two lengths: 3 s and 60 s. The average F1 score (across 10 models) on 3 s clips was 0.82 with a standard deviation of 0.027, with the best model achieving 0.86. When applied to 60 s clips, the best model achieved an F1 score of 0.96. We used the trained classifier to build a detector that processes longer recordings and will make it available as an open-source tool.

8:45

4aAB2. Quantifying northern bottlenose and sperm whale acoustic behavioural responses to anthropogenic noise in Baffin Bay, Canada. Kimberly Franklin (Oceanogr., Dalhousie Univ., Dept. of Oceanogr. Dalhousie University, LSC Rm. 3631, 1355 Oxford St., Halifax, NS B3H 4R2, Canada, kimberly.j.franklin@dal.ca), William D. Halliday (Wildlife Conservation Society Canada, Whitehorse, YT, Canada), David R. Barclay, and Sarah Fortune (Oceanogr., Dalhousie Univ., Halifax, NS, Canada)

Marine mammals rely on their auditory system for a myriad of life functions (e.g., navigating, foraging, socializing) and consequently, are vulnerable to loud human activities (e.g., vessel traffic, fishing, military activities). These activities can impede communication, cause behavioural disturbances, and can even cause injuries. As the Arctic warms and sea ice coverage decreases, more opportunities for human activities are arising. How noise impacts the acoustic behaviour of Arctic marine mammals is unclear. In October 2022 and 2023 controlled noise exposure experiments were conducted using military sonar (source level of 176.4 dB re 1 μPa) on northern bottlenose and sperm whales in Baffin Bay while they were foraging around vessels actively fishing. Hydrophone suction-cup biologgers (DTAGs; $n = 5$, $\sim$72 cumulative hours) were used to capture vessel and sonar noises, and whale vocalizations before, during, and after the noise exposure periods. Using a click detector, focal whale clicks were identified and quantitatively compared to received noise levels. This information will then be used to determine noise thresholds for acoustical behavioural responses. These results will support risk-mitigation strategies for the Department of National Defence Canada and Fisheries and Oceans Canada, as well as address Inuit concerns about the effects of military sonar on marine mammals.

9:00

4aAB3. Narwhal and beluga seasonal presence in Barrow Strait and Eclipse Sound, Nunavut from 2013–2019. Eva C. Hidalgo Pla (Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093, echidalg@ucsd.edu), Kait Frasier, John Hildebrand, and Joshua Jones (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA)

We detected narwhal and beluga echolocation clicks in acoustic recordings between 2013 and 2019 in two Canadian Arctic locations: central Barrow Strait and at the eastern entrance of Eclipse Sound. We determined the presence of narwhals and belugas in the region in long-term acoustic recordings, using a semi-automated process for echolocation click detection and differentiation between the two monodontid species in large acoustic datasets. Additionally, we compare mean daily sea-ice concentrations in the recording areas with the acoustic presence of the species. This work provides insights into the trends in monodontid presence between sites and across the acoustic record. Our findings also reveal instances when echolocation clicks coincided with the presence of commercial ships transiting past the recording locations, exposing animals to underwater radiated noise. This study contributes to our understanding of the seasonal movements of belugas and narwhals in Nunavut waters, revealing information on multi-year trends in their habitat utilization, relationships with sea-ice, and exposure to underwater noise from ships.

9:15

4aAB4. Diel and tidal cycles in summer habitat use by St. Lawrence Estuary Belugas through a pluriannual passive acoustics monitoring network. Samuel Giard (Pêches et Océans Canada, 850 Rte. de la Mer, Mont-Joli, QC G5H 3Z4, Canada, samuel.giard@dfo-mpo.gc.ca), Yvan Simard (Pêches et Océans Canada, Mont-Joli, QC, Canada), Nathalie Roy (Mont-Joli, QC, Canada), Florian J. Aulanier (Fisheries and Oceans Canada, Mont-Joli, QC, Canada), and Véronique Lesage (Pêches et Océans Canada, Mont-Joli, QC, Canada)

A network of 10 passive acoustics monitoring stations is used to examine patterns of habitat use at diel and tidal timescales by St. Lawrence Estuary beluga during summer 2018 and 2019. An occurrence index of vocal activity within the preferred frequency band of communications for belugas is used as a proxy for presence at the stations. Diel and tidal patterns of activity and mean residency time are extracted from statistics of hourly occurrence timeseries at the 10 stations. Spatially, diel and tidal occurrence levels of beluga communication sounds show patterns of variation that differ among the stations, but tend to be locally stable from one summer to next. Mean residency time at the 9 Estuary stations vary between 4 to 15 hours and most have an occurrence maximum during early morning. The Saguenay fjord station shows a distinct profile, with a mean residency time of 30 hours and high level of activity at evening and night. This work underlines

the ability of passive acoustics, through continuous monitoring at high spatio-temporal resolution, to reveal the complexity of the habitat use by this confined marine mammal population and understand its responses to diel and tidal forcings of the ecosystem.

9:30

4aAB5. Cetacean presence and ambient sound level analysis in Bermuda: A comparison to historical records. Dawn Parry (K. Lisa Yang Ctr. for Conservation Bioacoustics, Cornell Lab of Ornithology, Cornell Univ., 159 Sapsucker Woods Rd., Ithaca, NY 14850, dmp295@cornell.edu), Jean-Pierre Rouja (The Nonsuch Expeditions & Station-B, St. David's, Bermuda), and Holger Klinck (K. Lisa Yang Ctr. for Conservation Bioacoustics, Cornell Lab of Ornithology, Cornell Univ., Ithaca, NY)

Marine Protected Areas (MPAs) are established to conserve nature while also preserving ecosystem services and cultural values. Many ongoing MPA discussions are centered on offshore islands, including Bermuda, a British Overseas Territory in the North Atlantic Ocean, which is currently assessing a management plan to protect 20% of its EEZ. Marine mammals are essential targets to monitor as part of effective MPA management because they are keystone species. In addition, as human activities raise ocean noise levels worldwide, monitoring ambient sound levels is essential to evaluating impacts on marine ecosystems. We collected nine months of passive acoustic data (at 250 kHz sampling rate and with 24-bit resolution) between December 2021 and September 2022, approximately 24 nm south of Bermuda, close to a location where previous studies were conducted in 1966 and 2013–2014. The cetacean presence and ambient sound levels derived from our dataset will be compared to these historical datasets. Our comparative analysis focused on humpback whales (*Megaptera novaeangliae*), fin whales (*Balaenoptera physalus*), minke whales (*Balaenoptera acutorostrata*), blue whales (*Balaenoptera musculus*), sei whales (*Balaenoptera borealis)*, and beaked whales (family Ziphiidae). Results will illustrate how Bermuda's offshore underwater soundscape has changed over nearly six decades.

9:45

4aAB6. Using sensors of opportunity to estimate the density of blue and fin whales. Kerri D. Seger (Appl. Ocean Sci., 2127 1/2 Stewart St., Santa Monica, CA 90404, kerri.seger.d@gmail.com), Danielle V. Harris (Univ. of St. Andrews, St Andrews, United Kingdom), and Jennifer Miksis-Olds (Univ. of New Hampshire, Durham, NH)

The "Combining global OBS and CTBTO recordings to estimate abundance and density of fin and blue whales", or CORTADO project, is using data from two bottom-sensor types to implement a suite of methods for estimating density of fin and blue whales. While previous studies have demonstrated the utility of Ocean Bottom Seismometers (OBS) and Comprehensive Test Ban Treaty Organization (CTBTO) hydroacoustic data, the techniques have not yet been developed to the point where they can be routinely applied by the wider research community. The primary CORTADO goal is to provide a set of software tools and training materials to expand access and usability of large, historic datasets. This talk will focus on using the CTBTO sensors for estimating density with a bearing method. This technique uses the bearing of and the signal-to-noise ratio for each detected call to estimate how many animals are producing a set of calls at any given time over the detection range of the sensor at all bearings. Applying this tool to several CTBTO sensors, we can compare the amount of blue and fin whales over time as well as spatially across the northern and southern coverage areas of the sensor array.

10:00–10:15 Break

10:15

4aAB7. Of bats and robots. Rolf Müller (Mech. Eng., Virginia Tech, ICTAS II, 1075 Life Sci. Cir, (Mail Code 0917), Blacksburg, VA 24061, rolf.mueller@vt.edu), Ben Westcott (Elec. and Comput. Eng., Virginia Tech, Blacksburg, VA), and Chi Nnoka (Mech. and Aerosp. Eng., Univ. at Buffalo, The State Univ. of New York, Buffalo, NY)

Some bat species feature joint adaptations to their biosonar sensing and flight systems that allow them to navigate and hunt in dense vegetation. Understanding how biosonar sensing and flight are connected in these species poses a challenge: Kinematics recordings of bat-flight can document the system outputs, but due to various limitations, especially on processing, analysis of bat flight has been limited to minimal numbers of flights and hence has had difficulty to capture the natural variability in flight maneuvers. On the input side of the bats' flight-control system, it is necessary to understand the stimulus ensemble that guides flight in dense vegetation. Navigation in these cases must be based on clutter echoes, i.e., signals consisting of many unresolved components. Since the biosonar inputs into a bat are difficult to record without heavy interference with the animals' behaviors. Hence, biomimetic sonar robots can be used to collect stimuli from natural environments and their recreations. With their superior abilities to find patterns, deep-learning offer an opportunity to cut through the complexity of the input and output data of bat flight control and elucidate how clutter echoes are interfaced with the high-dimensional flight kinematics of bats to create the animals' exceptional maneuvering capabilities.

10:30

4aAB8. A robotic platform for integrating biomimetic sonar and AI. Ben Westcott (Elec. and Comput. Eng., Virginia Tech, Blacksburg, VA 24061, bwestcott@vt.edu), Ibrahim M. Eshera (Elec. and Comput. Eng., Virginia Tech, Blacksburg, VA), and Rolf Müller (Mech. Eng., Virginia Tech, Blacksburg, VA)

To gain a first-hand view of the biosonar inputs that a bat receives when operating in a complex natural environment, a bioinspired sonar head to mimic the auditory periphery of the bats' biosonar system is currently under development. The sonar head seeks to replicate the static geometric complexity of the baffles that emit the biosonar pulses and receive the returning echoes in horseshoe bats. Furthermore, some of the dynamic deformations of these structures are mimicked using tension-driven soft-robotics actuated by a set of 12 motors (two for the noseleaf and five for the each pinna). The periphery has been interfaced with acoustic emission and reception systems as well as a modular system control system for the acoustic input and output as well as the motors. Besides low-level control, the sonar head has onboard computing resources that can support deep-learning inference in the loop with acoustic data acquisition and pulse generation. All functional components have been integrated into a custom-designed shell that allows operating the system in natural outdoor environments. A wireless user interface allows remote control over the system so that the tasks of carrying and orienting the sonar head and controlling its function can be shared between two individuals.

10:45

4aAB9. Using a biomimetic sonar bat robot to explore the sensory world of bats in the field and in the laboratory. Nicholas Rock (Phys., Univ. of Rhode Island, 2 Lippitt Rd., Kingston, RI 02881, nicholasrock@uri.edu) and Rolf Müller (Mech. Eng., Virginia Tech, Blacksburg, VA)

Bat species such as those belonging to the families of horseshoe and Old World leaf-nosed bats make their navigation decisions based on echoes that are often received from dense vegetation. To gain an understanding of how

these animals accomplish navigation based on such complicated clutter echoes, a biomimetic sonar system that replicates the auditory periphery of bats has been used as synthetic observer to probe the sensory stimulus ensemble of the bats in a flight tunnel as well as in the field. Both laboratory and field experiments were carried out in Brunei on the island of Borneo. For the flight tunnel, the biomimetic sonar robot has been used to record a large number of pulse and echo pairs in conjunction with an array consisting of 28 ultrasonic microphones. In this experiment, the sonar head was positioned at points spaced on a 3d grid that covers the tunnel's volume. This data set will be used to train a deep neural network to predict the inputs to the ears of a bat maneuvering in the tunnel. In the field experiments, the sonar head was carried off road in various tropical-forest habitats to record echo samples in conjunction with precise GPS location references.

11:00

4aAB10. An experimental array setup to study the integration of biosonar and maneuvering flight in bats. Chi Nnoka (Dept. of Mech. and Aerosp. Eng., Univ. at Buffalo, The State Univ. of New York, 804 Furnas Hall, Buffalo, NY 14260, chinnoka@buffalo.edu), Yihao Hu (Dept. of Elec. and Comput. Eng., Virginia Tech, Blacksburg, VA), Ulmar Grafe (Dept. of Biology, Universiti Brunei Darussalam, Bandar Seri Begawan, Brunei Darussalam), Javid Bayandor (Dept. of Mech. and Aerosp. Eng., Univ. at Buffalo, The State Univ. of New York, Buffalo, NY), and Rolf Müller (Dept. of Mech. Eng., Virginia Tech, Blacksburg, VA)

Understanding biosonar-based flight control in bats remains a challenge for fundamental bioacoustics and holds promise for engineered flight-control systems, especially for highly maneuverable drones. To accomplish this, detailed data on bat flight is required that captures the variability across flight situations, individuals, and species. Hence, a flight tunnel has been constructed that provides enough space for natural flight maneuvers. Reconstructing the detailed geometry of freely maneuvering bats requires capturing a flying bat from many different angles to ensure freedom from occlusions. Ideally, this would also be done without placing artificial markers on the bat. Hence, a flight tunnel has been instrumented with arrays of 50 high-speed video cameras and 28 ultrasonic microphones that were all integrated within a modular canvas setup. The optical properties of each camera have been determined from multiple images taken of a panel with a calibration grid seen from different orientations. The spatial relationship between the cameras was estimated based on a large set of images containing random points created with an LED. Reconstructing the 3D geometries of the bats from large numbers of high-speed video frames requires an automated method due to the problem's complexity. To this end, deep-learning methods are currently under development.

11:15

4aAB11. A robotic platform for mimicking the flapping flight of bats. Logan Kelley (Mech. Eng., Virginia Tech, 635 Prices Fork Rd., Blacksburg, VA 24061, loganckelley@vt.edu), Jungsoo Park, Alexander Leonessa, and Rolf Müller (Mech. Eng., Virginia Tech, Blacksburg, VA)

To understand how bats control their wing movements based on biosonar inputs, it is important to identify the key kinematic synergies that need to be replicated on a robotic prototype so its wingbeat patterns and aerodynamic capabilities matches its biological counterpart. Attempting to replicate the approximately 20 degrees of freedom in each bat wing is challenging since more mechanical degrees of freedom increase mass and reduce reliability. Our proposed approach, is based on recordings of the kinematics of maneuvering bats that are obtained from high-speed camera data. The first two synergistic actions to implemented in the robot were wing flapping and folding that work together to increase net lift generation. During straight flight, these synergies follow a fixed synchronized pattern allowing a single-actuator robot model to mimic key bat flight characteristics. By recording the robot prototype's flight kinematics with the same high-speed camera setup, we can compare the flight kinematics of bats and flapping-flight robots inspired by them, quantifying differences in complexity and performance. The goal is a bioinspired robotic platform enabling detailed study of bat flight biomechanics and control strategies. This will provide new biological insights and advance flapping-flight robot development by determining kinematic complexity required for bat-like performance.

Session 4aBAa

Biomedical Acoustics, Structural Acoustics and Vibration, Computational Acoustics, Acoustical Oceanography, and Physical Acoustics: Wave Propagation in Complex Media: From Theory to Applications I

Pierre Bélanger, Cochair
1100 Rue Notre Dame O, Montréal, H3C1K3, Canada

Guillaume Haiat, Cochair
Multiscale modeling and simulation laboratory, CNRS, Laboratoire MSMS, Faculté des Sciences, UPEC, 61 avenue du gal de Gaulle, Creteil, 94010, France

Chair's Introduction—7:55

Invited Paper

8:00

4aBAa1. The acoustic trap theory and its application to lung ultrasound. Libertario Demi (Information Eng. and Comput. Sci., Univ. of Trento, Via sommarive 9, Trento, Italia 38123, Italy, libertario.demi@unitn.it)

In this lecture, the acoustic trap theory, as first introduced in 2016 in the context of lung ultrasound (DOI: 10.7863/ultra.15.08023), will be illustrated. Recent publications focusing on the design and validation of lung mimicking phantoms (DOI: 10.1038/s41598-017-13078-9, 10.1121/10.0001797, 10.1121/10.0007482) will be reviewed in the light of this theory. Moreover, its relevance to the monitoring and diagnosis of lung diseases by means of quantitative lung ultrasound spectroscopy will be discussed. To this end, results obtained within published (DOI: 10.1109/TUFFC.2020.3012289, 10.1121/10.0001723) and ongoing multicenter clinical studies, performed with dedicated multi-frequency imaging modalities implemented on open research scanners, will be presented and looked at through the lens of this theory.

Contributed Paper

8:20

4aBAa2. Poroelastic model of the lungs at low frequencies predicted by Biot's theory. Arife Uzundurukan (Ctr. de Recherche Acoustique-signal-humain, Université de Sherbrooke, Ctr. de Recherche acoustique-signal-humain, Université de Sherbrooke, Sherbrooke, QC J1K 2R1, Canada, arife.uzundurukan@usherbrooke.ca), Sébastien Poncet (Ctr. de Recherche Acoustique-signal-humain, Université de Sherbrooke, Sherbrooke, QC, Canada), Daria C. Boffito (Dept. of Chemical Eng., Polytechnique Montréal, Montréal, QC, Canada), and Philippe Micheau (Ctr. de Recherche Acoustique-signal-humain, Université de Sherbrooke, Sherbrooke, QC, Canada)

Biot's theory of poroelastic wave propagation inherently lends itself to elucidate the characteristics of a biphasic medium comprising solid and fluid components, such as biological tissues. One of the intricately complex biological tissues that remains poorly understood is the lungs since their properties diversify significantly through their pore geometries affected by inspiratory positive airway pressure (IPAP) and applied frequency range. One hypothesizes that the vibroacoustic behaviour of the lungs can be predicted by Biot's theory, as the nature of the lungs aligns with the principles of the theory at low frequencies. This study aims to analytically investigate the vibroacoustic behaviour of the lungs, considering 10 and 20-cm H_2O IPAP. Utilizing a fractional derivative formulation, one predicts the complex-valued shear wave speed, as well as the fast and slow compression wave speeds, for frequencies ranging from 5 to 100 Hz. A 3D digital thorax twin study using these predicted wave speeds, particularly at 28 Hz and 20 cm H_2O IPAP, is validated against experimental data from the literature. Consequently, applying Biot's theory provides a valuable framework for understanding the dynamic vibroacoustic behaviour of the lung tissues in response to varying IPAP and low frequencies.

Invited Paper

8:35

4aBAa3. 2D boundary-condition-free nonlinear inversion technique applied to optical shear vibration induced microelastography. Elijah E. W. Van Houten (Univ. of Sherbrooke, University of Sherbrooke, Sherbrooke, QC, Canada, Sherbrook, QC, Canada, elijah.van.houten@usherbrooke.ca), Sajad Ghazavi, Guillaume Fle, Hari s. Nair, Boris Chayer (Lab. of Biorheology and Medical Ultrason., Univ. of Montreal Hospital, Montreal, QC, Canada), Ruchi Goswami, Salvatore Girardo, Jochen Guck (Max Planck Inst. for the Sci. of Light & Max-Planck-Zentrum für Physik und Medizin, Erlangen, Germany), and Guy Cloutier (Lab. of Biorheology and Medical Ultrason., Univ. of Montreal Hospital, Montréal, QC, Canada)

Optical microelastography (OME) has emerged as a new technique for quantifying cellular mechanical properties. However, accurately reconstructing viscoelastic properties at the microscale level from noisy 2D displacement fields remains a challenge. This study introduces a 2D boundary-condition-free nonlinear inversion (2D-NoBC-NLI) approach, addressing challenges of interpreting noisy data and deducing full-field 3D displacements from 2D measurements. OME requires vibrating the cell and mapping the shear modulus based on wave-induced displacements within the cell. The shear modulus distribution is recovered via a coupled adjoint field NLI reconstruction to allow 2D-NoBC-NLI. Validation was conducted through numerical simulations at 36 kHz on a homogeneous sphere of 75 μm diameter and an assigned viscoelastic modulus, G*, of $800 + i150$ Pa. The same reconstruction approach was also applied to experimental data obtained from polyacrylamide (PAAm) microbeads of the same diameter. Results demonstrated relative differences from true simulated values of 0.7% and 45% for storage and loss moduli, respectively, with a coefficient of variation under 1% for homogeneous regions. When applying this method to PAAm microbeads, viscoelastic reconstructions showed the potential of OME under experimental conditions. These findings highlight the accuracy of 2D-No BC-NLI reconstruction in OME for precise microscale characterization and mapping of the viscoelastic cell structure.

Contributed Papers

8:55

4aBAa4. Signal detrending in elastographic wave data with large motion artifacts through a 2D Whittaker smoother. Rosen P. David (Physiol. & Biomedical Eng., Mayo Clinic College of Medicine and Sci., Medical Sci. Bldg., 321 3rd Ave SW, Rochester, MN 55902, Rosen.David@mayo.edu), Azra Alizad (Radiology, Mayo Clinic College of Medicine and Sci., Rocheter, MN), and Mostafa Fatemi (Physiol. & Biomedical Eng., Mayo Clinic College of Medicine and Sci., Rochester, MN)

A variety of advanced shear wave elastography techniques require the measurement of frequency-resolved phase velocity to properly characterize wave signals where factors such as tissue viscosity or guided wave effects are present. However, when applying these techniques in a dynamic environment where extrinsic motion cannot be limited, nonlinear and nonstationary trends can be expected in the motion signal used for phase velocity estimation. Because such extrinsic motion signals can contaminate a broad range of frequencies, they can produce significant errors in the phase velocity estimates. In this study, we propose a spatiotemporal detrending approach for elastography wave motion signals built from a 2D Whittaker smoother. This smoother has its origins in nonparametric regression and is well suited for fitting nonlinear curves of arbitrary shape. The resulting detrending filter was applied to Lamb wave signals from clinical ultrasound bladder vibrometry measurements. We found that the proposed detrending filter allowed for phase velocity estimates to be made in acquisitions that are otherwise be too corrupted by large motion. This works was supported by a grant from the NIH (DK131685)

9:10

4aBAa5. Surfacic characterization of soft tissues biomechanical properties using impact-based methods: A comparative study. Arthur Bouffandeau (CNRS, MSME UMR CNRS 8208, 61, Ave. du Général de Gaulle, Créteil 94010, France, arthur.bouffandeau@u-pec.fr), Giuseppe Rosi (Université Paris-Est Créteil, Creteil, France), Sabine Bensamoun (Université de technologie de Compiègne-UTC, Biomechanics and Bioengineering UMR CNRS 7338, Compiègne, France), Charles-Henri Flouzat-Lachaniette (INSERM U955, IMRB Université Paris-Est, Créteil, France), Vu-Hieu Nguyen (Université Paris-Est Créteil, Créteil, France), and Guillaume Haiat (CNRS, MSME UMR CNRS 8208, Créteil, France)

Various methods have been developed to assess the skin stiffness in order to assist the clinician for therapeutic monitoring or these diagnoses. The objective of this work was to compare the performances of a new acoustical characterization method based on impact analysis with those of two existing approaches, namely (i) a suction device, the Cutometer®, and (ii) a digital palpation tool, the MyotonPro®. This new impact analysis method is based on the analysis of the temporal variation of the force obtained during the impact of an instrumented hammer on a cylindrical punch placed in contact with a soft tissue mimicking phantom (Technogel®). The performances of the three aforementioned techniques (sensitivity and resolution) were assessed using homogeneous or bilayer structures with various thickness and rigidity, formed by different soft tissues mimicking gel pads. The impact analysis based method (IBAM) was two times better than the other two approaches in terms of reproducibility and sensitivity. The axial resolution of the IBAM was around 20 times better compared to the two other methods. The results open the way for the development of a cheap, non-invasive and objective method that could be used in the future in the cosmetic industry and in dermatology. This project has received funding from the projects OrthAncil (ANR-21-CE19-0035-03) and from the project OrthoMat (ANR-21-CE17-0004).

9:25

4aBAa6. The effect of cavitation-induced pressure on bubble cloud density in histotripsy. Adam Maxwell (Biomedical Eng. and Mech., Virginia Tech, Dept. of Urology, University of Washington, Seattle, WA 98195, amax38@u.washington.edu) and Eli Vlaisavljevich (Biomedical Eng. and Mech., Virginia Tech, Blacksburg, VA)

Tissue ablation in histotripsy is achieved by generating clouds of cavitation or boiling bubbles in the tissue. The rate of tissue ablation increases for clouds with a greater number density of nucleated bubbles. Therefore, controlling density may offer a mechanism to improve treatment efficacy. Experiments demonstrate several trends in density with ultrasound frequency, transducer f-number, and other variables. This presentation will describe one potential mechanism governing the density of cavitation bubbles nucleated during a focused ultrasound pulse. In particular, the rapid expansion of a cavitation bubble generates a partial cancellation of the incident pressure in the vicinity of the bubble, mitigating potential nucleation of other bubbles. We demonstrate this effect through a single-bubble numerical model, and further evaluate dependences of the pressure field with nuclei size, pressure amplitude, pulse frequency, and medium properties. The single-bubble model is further extended to consider the evaluate the effect of transducer focusing, and the combined pressure fields of multiple bubbles during cavitation nucleation. The predictions from simulation show good agreement with experimentally reported trends with frequency and transducer f-number, supporting the role of the mechanism in limiting the nucleation density in histotripsy.

9:40–9:55 Break

Invited Papers

9:55

4aBAa7. Ultrasound propagation in cancellous bone: Dependence on mechanical, material, and structural properties. Keith A. Wear (Ctr. for Devices and Radiological Health, Food and Drug Administration, Bldg 62 Rm. 2114, 10903 New Hampshire Ave., Silver Spring, MD 20993, keith.wear@fda.hhs.gov)

Critical skeletal sites for osteoporotic fractures include femur and vertebrae. However, many ultrasound devices designed for the management of osteoporosis instead target the calcaneus, which is far more accessible to ultrasound than femur or vertebrae. Calcaneus-based devices have been shown to be very effective for predicting osteoporotic fractures of the hip. MicroCT-based finite element models support targeting of calcaneus based on similar biomechanical structure-function relations in calcaneus compared with femoral neck, greater trochanter, proximal tibia, and vertebra. Finite element analysis can be used to relate these mechanical properties to ultrasonic properties such as broadband ultrasound attenuation, speed of sound, and broadband ultrasound backscatter. As an ultrasound beam propagates through cancellous bone, it loses energy to absorption, longitudinal-shear scattering, and longitudinal-longitudinal scattering. The relative roles of these mechanisms can be demonstrated in phantom models.

10:15

4aBAa8. Competing effects of ultrasound absorption and scattering in porous bone. Brett A. McCandless (Mech. and Aerosp. Eng., North Carolina State Univ., Raleigh, NC), Kay Raum (Charite Univ. Hospital, Berlin, Germany), and Marie Muller (MAE, North Carolina State Univ., 911 Oval Dr., Eng. Bldg. III, Raleigh, NC 27606, mmuller2@ncsu.edu)

The mechanisms and effects of ultrasonic attenuation in porous cortical bone are poorly understood, and it is necessary to better understand them to evaluate bone porosity noninvasively using ultrasound. A finite-difference time domain numerical study was conducted in which ultrasound propagation was simulated in human femur cross-sections obtained via scanning acoustic microscopy. The effect of absorption on overall attenuation was studied by varying the nominal absorption level attributed to the solid matrix. Ultrasound pulses were emitted with a central frequency of 8 MHz in through-transmission and backscattering configurations. From these data, the respective extinctions lengths due to overall attenuation, scattering, and absorption were obtained. Two regimes seem to exist depending on the nominal absorption value. At low absorption values, scattering dominates overall attenuation, scattering and absorption appear to have a synergistic effect on overall attenuation, and the diffusion constant decreases with increasing average pore diameter. At high absorption values, absorption dominates overall attenuation, the extinction length for scattering increases with pore diameter, and the diffusion constant increases with increasing average pore diameter. These regimes affect how ultrasound parameters, such as the extinction lengths due to scattering, absorption, and overall attenuation, should be used to evaluate the porosity of cortical bone.

10:35

4aBAa9. Electrical potentials in bone induced by ultrasound propagation. Mami Matsukawa (Doshisha Univ., 1–3, Tatara Miyakodani, Kyotanabe, Kyoto 610-0321, Japan, mmatsuka@mail.doshisha.ac.jp), Shouta Kitajima (Doshisha Univ., Kyotanabe, Japan), and Atsushi Hosokawa (Dept. of Elec. and Comput. Eng., National Inst. of Technol., Akashi College, Akashi, Japan)

Bone with complex geometry is hard, heterogeneous and anisotropic tissue, which makes ultrasound diagnosis very difficult. Therefore, the simulation of wave propagation is often performed to understand the complicated wave propagation in bone, where we find solid-liquid coexisting conditions. In addition to the diagnosis, one of the recent topics of bone studies are ultrasound fracture healing. The low intensity pulsed ultrasound (LIPUS) is popular as a fracture treatment technique in the field of orthopedic surgery, although how bones detect high frequency ultrasound is still under discussion. In this study, we focus on the weak piezoelectricity of bone as one of the key properties for the LIPUS treatment. According to the piezoelectric finite difference time-domain (PE-FDTD) method, we have investigated ultrasound propagation and generation of electrical potentials in bone. First, we verified the PE-FDTD method by comparing the simulated electric fields with the experimental data obtained by an ultrasound receiver using bone as the piezoelectric element. Next, we have tried to understand the wave propagation and generation in a real bone model (the radius of a 66-year-old woman). The generation of electrical potentials in the cancellous bone was also studied by simulation and experiments. The use of human bone to fabricate a model was permitted by the ethical committee at Doshisha University.

10:55

4aBAa10. Fast and accurate transcranial ultrasound simulation using the asymptotic model of the Civa Healthcare platform. Sylvain Chatillon (LIST, CEA, Institut CEA LIST CEA Saclay, Bât. Digiteo - 565, Gif sur yvette 91191, France, sylvain.chatillon@cea.fr), Andrew Drainville (LabTAU INSERM 1032, Lyon, France), John Snell, David Moore, Frederic Padilla (Focused Ultrasound Foundation, Charlottesville, VA), and Cyril Lafon (LabTAU INSERM 1032, Lyon, France)

To ensure its efficacy and safety, transcranial ultrasound therapy treatment planning requires accurate pressure field simulations and phase law corrections. Despite their long computation time and high memory usage, full numerical methods are often used since they are considered more accurate than semi-analytical methods. This work will present the so-called “pencil method“, a fast asymptotic model embedded in the CIVA HealthCare simulation platform. It allows computation in harmonic and impulse mode and the consideration of complex configurations, including solid obstacles, considering, at each interface, refractions and reflections with or without mode conversion of the acoustic field. This model was successfully compared to a recent collaborative work by Aubry *et al*. that presented a set of numerical benchmarks for transcranial propagation, to allow comparisons between various modeling tools. It was used to investigate the influence of parametric variation of skull material properties on the quality of acoustic focusing through the human skull. Its ability to predict the thermal rise at the intracranial target was validated against experimental data obtained *ex-vivo* through human skulls. Finally, works in progress will be shared about its connection to the open-source Kranion software developed at the FUS Foundation to facilitate comparison between clinical and simulated data.

11:15

4aBAa11. Validation of mSOUND using a fully heterogeneous skull model. Jeff J. Bell (Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, jjb7481@psu.edu), Lu Xu, Hong Chen (Washington Univ. in St.Louis, St Louis, MO), and Yun Jing (Acoust., Penn State Univ., State College, PA)

Transcranial ultrasound has found an increasing number of applications in recent years, including the treatment of neurological conditions through thermal ablation and neuromodulation. Ensuring the success and safety of such treatments necessitates precise numerical simulations of transcranial ultrasound, a pivotal aspect of treatment planning involving phase correction. Addressing this demand, an open-source wave solver named mSOUND (https://m-sound.github.io/mSOUND/home) was developed specifically for modeling focused ultrasound in heterogeneous media. A recent intercomparison study (J. Acoust. Soc. Am. 152, 1003–1019, 2022) scrutinized mSOUND alongside other wave solvers like k-Wave, demonstrating its accuracy in modeling wave propagation through a homogeneous skull. This study extends the assessment to evaluate mSOUND's accuracy in modeling wave propagation through a fully heterogeneous skull, utilizing CT images of an *ex vivo* human skull. The obtained results are systematically compared with those from k-Wave, revealing a high level of agreement.

11:30

4aBAa12. Simulation-corrected focusing to the vertebral canal. David Martin (Physical Sci., Sunnybrook Res. Inst., 2075 Bayview Ave., Toronto, ON M4N 3M5, Canada, dave.martin@mail.utoronto.ca), Rui Xu (Medical Phys. and Biomedical Eng., Univ. College London, London, United Kingdom), and Meaghan O'Reilly (Physical Sci., Sunnybrook Res. Inst., Toronto, ON, Canada)

Phased arrays have long been used to deliver focused ultrasound to the brain, but applications in the spinal cord remain comparatively unexplored. This is largely due to the aberrating effect of vertebral bone on the incoming wavefront, where variable density and complex geometry distort the pressure field in the canal. For controlled focusing, phase and amplitude corrections must therefore be calculated in advance. Here, we validate a previously-developed ray acoustics model for transvertebral focusing with a bilateral spine-specific array. Benchtop trials were conducted with *ex vivo* human vertebrae and ray acoustics beamforming was compared to a geometric baseline and hydrophone-corrected gold standard. Planar shift in the 90% contour was evaluated via raster scans of the sagittal and coronal planes. Ray acoustics correction reduced mean sagittal shift from 1.76 ± 0.79 mm to 1.63 ± 0.86 mm and reduced mean coronal shift from 0.99 ± 0.54 mm to 0.76 ± 0.63 mm, while hydrophone correction produced mean sagittal and coronal shifts of 1.40 ± 0.83 mm and 0.53 ± 0.47 mm, respectively. Large variance in simulation-corrected results is hypothesized to stem from nonuniform attenuation and intervertebral acoustic windows, a possibility that will be explored in future *in silico* and benchtop work.

Session 4aBAb

Biomedical Acoustics and Physical Acoustics: Droplets Strike Back I

Virginie Papadopoulou, Cochair
Biomedical Engineering, The University of North Carolina at Chapel Hill, 116 Manning Drive, 9004 Mary Ellen Jones Building, CB 7575, Chapel Hill, NC 27599-7575

Mario L. Fabiilli, Cochair
University of Michigan, 1301 Catherine Street, 3226A Med Sci I, Ann Arbor, MI 48109

Kevin J. Haworth, Cochair
Department of Internal Medicine, University of Cincinnati, 231 Albert Sabin Way, CVC 3939, Cincinnati, OH 45267

Chair's Introduction—9:00

Invited Papers

9:05

4aBAb1. Acoustic vaporization of perfluorohexane droplets is induced by heterogeneous nucleation at 1.1 MHz. Rashmi Ramesh, Chloë Thimonier (Laboratoire d'Imagerie Biomédicale, Sorbonne Université, Paris, France), Stéphane Desgranges (Avignon Université, Avignon, France), Vincent Faugeras (Ecole Normale Supérieure, Paris, France), François Coulouvrat (Sorbonne Université, Paris, France), Justine Laurent (ESPCI, Paris, France), Guillaume Marrelec (Laboratoire d'Imagerie Biomédicale, Sorbonne Université, Paris, France), Christiane Contino-Pépin (Avignon Université, Avignon, France), Wladimir Urbach, Christophe Tribet (Ecole Normale Supérieure, Paris, France), and Nicolas TAULIER (Laboratoire d'Imagerie Biomédicale, Sorbonne Université, 15 rue de l'école de médecine, Paris 75006, France, nicolas.taulier@sorbonne-universite.fr)

We investigated the ADV of perfluorohexane (PFH) nano and micro-droplets at a frequency of 1.1 MHz, at conditions where there is no superharmonic focusing. Our experiments were performed on suspensions of droplets in glycerol to avoid sedimentation. The ADV pressure threshold was defined as the pressure for which there is half a chance to observe a vaporization event. Our experiments showed that the threshold depends on the number of insonified droplets and the data were fitted by a statistical model, that is independant on the mechanism leading to ADV. The fit allows the derivation of the threshold in the case of a single insonified droplet. We further observed that the value of the ADV pressure threshold of a single PFH droplet decreases as its radius increases. In addition, at a constant droplet radius, the threshold did not vary when the PFH volume inside the droplet decreases, thanks to the encapsulation of an increasing number of water droplets. These results are incompatible with a model of homogeneous nucleation (where a nucleus can appear anywhere into the PFH volume). However, we developed a heterogeneous nucleation model, where the nucleus appears at the inner droplet surface, that successfully predicts our experimental ADV results.

9:30

4aBAb2. Predicting the spontaneous vaporisation of superheated perfluorocarbon droplets. Luca Bau, QIANG WU (Univ. of Oxford, Oxford, United Kingdom), Nicholas Ovenden (Mathematics, Univ. College London, London, United Kingdom), and Eleanor P. Stride (Univ. of Oxford, Inst. of Biomedical Eng., Oxford OX3 7DQ, United Kingdom, eleanor.stride@eng.ox.ac.uk)

Superheated perfluorocarbon droplets have been widely explored as agents for ultrasound imaging and therapy, as well as for other applications such as radiation dosimetry. Submicrometre, or "nano" droplets offer a number of potential advantages over microbubbles, e.g., longer circulation half-lives, higher surface area-to-volume ratio and the ability to perfuse the microvasculature more easily. A key challenge in the use of nanodroplets, however, is the need to avoid spontaneous vaporisation whilst keeping the energy required for acoustic activation within the range of pressures that can be used safely in humans. This is especially important for imaging applications. Perfluoropropane (C_3F_8) microbubbles can be condensed to form liquid nanodroplets that offer a good trade-off between thermal stability and acoustic vaporisation threshold. Anecdotal reports, however, suggest that C_3F_8 droplets can spontaneously vaporise, and may therefore pose a potential safety risk, especially if bubble coalescence occurs. The aim of this study was to build on recent theoretical models of droplet vaporisation and investigate the probability of vaporisation as a function of temperature and interfacial tension. The results are compared with experimental measurements of vaporisation rates for different droplet formulations.

Contributed Papers

9:55

4aBAb3. Non-equilibrium activation dynamics of superheated phase-change contrast agents. Nicholas Ovenden (Mathematics, Univ. College London, Dept of Mathematics, University College London, London WC1E 6BT, United Kingdom, n.ovenden@ucl.ac.uk), Luca Bau, and Eleanor P. Stride (Univ. of Oxford, Oxford, United Kingdom)

Phase-change contrast agents, such as superheated perfluorocarbon droplets, potentially offer exciting new opportunities in diagnostic and therapeutic applications of ultrasound. An important advantage of droplets over, e.g., microbubble agents is that they can be "activated" (vaporised) by a reduction in ambient pressure, initiated by an externally applied ultrasound pulse. Controlling the timing and location of droplet activation, however, is vitally important to avoid safety concerns. Current theoretical models for droplet vaporisation assume that a critical stable bubble embryo forms that is in thermal and mechanical equilibrium. Model predictions are, however, inconsistent with some experimental observations. The aim of this study is to develop a non-equilibrium model for droplet activation wherein initial bubble growth rate and vapour pressure are dependent upon the bubble nucleus size. We demonstrate how incorporating nonequilibrium effects into theoretical models of droplet vaporisation dynamics enables better agreement of the droplet expansion with experimental data and thus provide greater insight into control of the activation process.

10:10

4aBAb4. Microcavitation dynamics of vaporized perfluorocarbon nanodroplets captured with an acoustical camera. Mark Burgess (Dept. of Medical Phys., Memorial Sloan Kettering Cancer Ctr., 321 E 61st St., New York, NY 10065, burgesm1@mskcc.org) and Jeffrey A. Ketterling (Radiology, Weill Cornell Medicine, New York, NY)

Perfluorocarbon nanodroplets are on-demand cavitation nuclei for numerous biomedical applications. These nanodroplets remain in a metastable state until an ultrasound pulse of sufficient negative pressure initiates conversion from a liquid droplet into a gaseous micro- or nano-bubble in a process termed acoustic droplet vaporization (ADV). Extensive research has gone into understanding their interaction with ultrasound at varying acoustic and ambient conditions. Recent studies have detected intra- and post-excitation collapse emissions indicative of inertial cavitation. This study aims to further understand the post-ADV dynamics of perfluorocarbon nanodroplets with an acoustical camera technique. Nanodroplets were activated with a low-frequency ultrasound pulse (5 MHz) while being insonified with a high-frequency (35 MHz) probing wave. The side scattered emissions from the probing wave were captured with another matched high-frequency transducer. The amplitude modulation of these emissions is proportional to the radial strain; thus, a relative radius-time curve can be extracted from the scattered signal after filtering and enveloping. Results show that this technique can capture the radial growth and collapse curves of these newly formed bubbles and aligns well with intra- and post-excitation emissions indicative of inertial cavitation. This approach could be useful to understand post-ADV dynamics of various droplet formulations.

10:25–10:50 Break

Invited Paper

10:50

4aBAb5. Peak positive pressure matters in acoustic droplet vaporization. Samuele Fiorini, Anunay Prasanna, Gazendra Shakya (ETH Zurich, Zurich, Switzerland), and Outi Supponen (ETH Zurich, Sonneggstrasse 3, ETH Zurich - D-MAVT - IFD - ML H31, Zurich 8092, Switzerland, outis@ethz.ch)

We demonstrate the significance of the positive pressure component of an ultrasound wave in acoustic droplet vaporization. Theory and acoustic simulations reveal that the distorted compression part of the incoming wave, which includes a broad spectrum of frequencies above the fundamental, presents a pronounced shift in phase when focusing within the droplet and crossing the focal point. The extent of this so-called Gouy phase shift is sufficient to change the sign of the compression phase of the wave, actively creating a tension region in the droplet bulk in the same location where we experimentally observe vapor nucleation. A sign reversal of the rarefaction component of the ultrasound wave is, on the other hand, not expected. The extent of distortion of the incoming ultrasound wave can influence the occurrence of droplet vaporization, even at constant peak negative pressure, which may partially explain the broad range of vaporization threshold values reported in literature. The results suggest that vaporization can be achieved exploiting ultrasound waves with high peak positive pressures and reduced peak negative pressures.

Contributed Papers

11:15

4aBAb6. Gold nanoparticle-coating reduces the acoustic pressure threshold for nanodroplet vaporization. Nishita Mistry, Ruchika Dhawan (Dept. of Elec. Eng., Indian Inst. of Technol. Gandhinagar, Gandhinagar, Gujarat, India), and Karla Patricia Mercado-Shekhar (Dept. of Biological Sci. and Eng., Indian Inst. of Technol. Gandhinagar, Academic Block 6/207, Indian Inst. of Technol. Gandhinagar, Gandhinagar, Gujarat 382355, India, karlamshekhar@iitgn.ac.in)

Nanodroplets with a low-boiling-point perfluorocarbon (PFC) may vaporize spontaneously at physiological temperatures. Using a high-boiling-point PFC, e.g., perfluorohexane (PFH), can enable ultrasound-triggered vaporization for applications in imaging and drug delivery. However, PFH requires relatively high ultrasound (US) pressures for vaporization compared to low-boiling-point PFCs, making its use challenging. We investigated the feasibility of reducing the vaporization threshold by gold-coating lipid-encapsulated PFH nanodroplets (Au-PFH-NDs). We synthesized Au-PFH-NDs with 200 nm mean particle size and 5.12×10^{-4} pg mass of gold-coating per nanodroplet. B-mode images of the emulsion perfused in a flow phantom were used to determine the pressure threshold for ND vaporization upon exposure to 2 MHz focused US (f-number of 1.27, 0.5% duty cycle). The pressure threshold for Au-PFH-NDs (3.97 ± 0.63 MPa) was significantly lower ($p < 0.05$) than that of NDs without gold-coating (7.07 ± 0.02 MPa), indicating that gold-coating reduced the vaporization

pressure threshold. In addition, the pressure threshold of the Au-PFH-NDs was not significantly different ($p > 0.05$) from that of perfluoropentane (PFP) NDs (4.16 ± 0.01 MPa). These results suggest that the Au-PFH-NDs can be vaporized at similar pressures as PFP NDs, but are more stable at physiological temperatures. These findings are the first step towards employing gold-coated PFC nanodroplets with a lower vaporization pressure threshold for multimodal imaging and drug delivery.

11:30

4aBAb7. Droplets for acoustic vaporization: Formulations, properties and applications. Gaio Paradossi (Dept. of Chemical Sci. and Technologies, Univ. of Tor Vergata, Dept. of Chemica Sci. and Technologies, Via della Ricerca Scientifica 1, Rome, RM 00133, Italy, gaio.paradossi@uniroma2.it) and Fabio Domenici (Dept. of Chemical Sci. and Technologies, Univ. of Tor Vergata, Rome, RM, Italy)

The Acoustic Droplet Vaporization (ADV) (1), a phase-change of the droplets core from liquid to vapor phase upon ultrasound irradiation, burst renewed interest in droplet emulsions, a traditional topic of colloidal science, opening up innovative applications in biomedicine. Droplets undergoing ADV share similar liquid cores, typically perfluorocarbons (PFCs), however, the nature of the shells can be polymeric (2) or lipidic (3, 4). Such difference imparts to phase-change droplets diverse acoustic and mechanical behaviors (5). In this contribution we present some results concerning a general strategy for the formulation of polymer or lipid shelled submicron droplets. Other key points for the use of ADV-responsive in biomedical applications will be addressed. Recently we have extended the concepts of ADV to radiation responsive droplets for dosimetry in cancer treatment with hadronic radiation. Results on this activity will be also reported (6). References: (1) *J. Acoust. Soc. Am.* (2004) **116** (1): 272–281; (2) *Chem. Commun.*, 2013, 49, 5763; (3) *JoVE*, *169* (2021); (4) *Langmuir* (2019), 35, 10116–10127; (5) *Phys. Chem. Chem. Phys.*, 2016, 18, 8378; (6) *COCIS*, *49*, 118–132 (2020).

THURSDAY MORNING, 16 MAY 2024 — ROOM 211, 9:00 A.M. TO 10:15 A.M.

Session 4aCA

Computational Acoustics, Structural Acoustics and Vibration and Engineering Acoustics: Validation and Verification

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

Anthony L. Bonomo, Cochair
Naval Surface Warfare Center, Carderock Division, 9500 MacArthur Blvd, West Bethesda, MD 20817

Chair's Introduction—9:00

Invited Papers

9:05

4aCA1. Validation and verification of models for setting SONAR system requirements. Jennifer Cooper (Johns Hopkins Univ. Appl. Phys. Lab., 11100 Johns Hopkins Rd., Mailstop 8-220, Laurel, MD 20723, jennifer.cooper@jhuapl.edu), Jean Dougherty, and Marina Johnson (Johns Hopkins Univ. Appl. Phys. Lab., Laurel, MD)

When determining requirements for a SONAR system as a whole, typically very high-level requirements such as number of detections and total false alarms allowed per unit time must be converted into specific requirements for system components. These requirements may include array requirements like hydrophone count and position. Validation of probability of detection via sea test is fraught with complications, not least of which is environmental uncertainty. As a precursor, it is generally advisable to approach and attempt to validate models for each term in the SONAR equation one at a time. The approaches to validation vary depending on the term and can range in complexity from comparison to analytical solutions, to comparison with alternate available models, up through data collections at sea. Each of these approaches comes with its own challenges and limitations, which are discussed.

9:25

4aCA2. Data and information management for acoustics research. Amanda Hanford (Penn State Univ., PO Box 30, State College, PA 16802, ald227@psu.edu), Tyler P. Dare (Penn State Univ., State College, PA), Keith Rice (Penn State Univ., University Park, PA), and Andrew S. Wixom (Penn State Univ., State College, PA)

A critical part of verification and validation in academic research includes the important consideration of data and information management. As researchers grapple with escalating volumes of data, effective data management becomes imperative for optimizing operational processes and ensuring reproducibility and archivability. Data management involves the organization, storage, and retrieval of information to support research advancements and strengthen the foundation of decision making and scientific knowledge. Key components also include data operations, which involves the orchestration of data workflows, and data quality management, which focuses on maintaining accurate and reliable data. This talk explores the multifaceted aspects of data management, emphasizing its significance in ensuring data quality, accuracy and accessibility.

Contributed Papers

9:45

4aCA3. Predicting sound transmission loss in particle-reinforced polymer composites using machine learning: A comparative study with experimental and theoretical results. Jonty Mago (Automotive Health Monitoring Laboratory/Ctr. for Automotive Res. and Tribology, Indian Inst. of Technol. Delhi, Hauz Khas, New Delhi, Delhi 110016, India, jontymago@gmail.com), Sunali Jaish (Automotive Health Monitoring Laboratory/Ctr. for Automotive Res. and Tribology, Indian Inst. of Technol. Delhi, New Delhi, Delhi, India), Ashutosh Negi (School of Interdisciplinary Res., Indian Inst. of Technol. Delhi, New Delhi, Delhi, India), J. S. Bolton (Ray W. Herrick Laboratories/School of Mech. Eng., Purdue Univ., USA, West Lafayette, IN), and S Fatima (Automotive Health Monitoring Laboratory/Ctr. for Automotive Res. and Tribology, Indian Inst. of Technol. Delhi, New Delhi, Delhi, India)

In this study, the use of machine learning (ML) models to predict sound transmission loss (STL) in particle-reinforced polymer composites has been examined. The work included extensive literature research and data extraction for training various ML models, focusing on their effectiveness in accurately predicting STL, which is crucial for evaluating acoustic performance. The method involves processing data and applying it to different ML algorithms, with the models calibrated and tested for reliability. A key aspect is comparing these models' predictions with actual experimental results and theoretical models based on the mass law in acoustics. The findings reveal the potential and limitations of ML in materials science, showing their accuracy in predicting STL and comparing them with traditional theories. This research advances the use of data-driven methods in developing and assessing acoustic materials, significantly impacting materials science and machine learning.

10:00

4aCA4. Sound propagation modeling in forests using the transmission line matrix method: Comparison with experimental *in situ* measurements. Quentin Goestchel (School of Oceanogr., Univ. of Washington, 1503 NE Boat St., Seattle, WA 98115, qgoestch@uw.edu), Gwenaël Guillaume, Ecotière David (UMRAE, CEREMA, Univ Gustave Eiffel, F-67035 Strasbourg, France, Strasbourg, France), and Gauvreau Benoit (UMRAE, Univ Gustave Eiffel, CEREMA, F-44344 Bouguenais, France, Nantes, France)

We present efforts to adapt the time-domain Transmission Line Matrix (TLM) method for modeling sound propagation in forests. It is relevant to professionals and researchers in environmental acoustics and to those interested in studying sound propagation in outdoor environments using numerical modeling techniques. The study aims to demonstrate the applicability of the method in complex media where sound waves undergo multiple reflections combined with ground effects during their propagation from a sound source to a receiver point. The TLM method is used to numerically model sound propagation in a 3D forest geometry generated based on a real tree distribution case. Data from an experimental campaign conducted in the rainforest of French Guyana are used as a reference. The results show an encouraging comparison between them and the *in situ* measurements, even if the hypotheses made to adapt the input data in the model were strong. We find that more measurement points and more specifications (ground impedance measurements, canopy density, etc.) about the experimental site are needed for further simulations, even if obtaining them presents technical difficulties.

10:15

4aCA5. Predicting excess attenuation: An investigation of turbulence in a littoral environment. Andrea Vecchiotti (Dept. of Eng., East Carolina Univ., Washington, DC), Diego Turo (Mech. Eng., The Catholic Univ. of America, 620 Michigan Ave., N.E., Washington, Washington, DC 20064, turo@cua.edu), Matthew Stengrim, Jeff Foeller, Teresa J. Ryan (Dept. of Eng., East Carolina Univ., Greenville, NC), and Joseph Vignola (Mech. Eng., The Catholic Univ. of America, Washington, DC)

This work presents an effort to effectively represent turbulence in a numerical model of atmospheric sound propagation in a near-shore environment. Model results are compared with a set of atmospheric transmission loss measurements made in a pitch-catch configuration with a stationary source 500 m from shore. High resolution temperature profiling and scanning Doppler LIDAR wind profiling measurements in the source-to-receiver direction are concurrent with the acoustic transmission loss measurements at the shoreline. These meteorological measurements inform the input parameters used in the parabolic equation. The numerical predictions are compared to the measured transmission loss values to evaluate model performance with a variety of different approaches to implement turbulence parameters.

Session 4aID

Interdisciplinary, Student Council, Physical Acoustics, and Education in Acoustics: Graduate Studies in Acoustics Poster Session

Pratik A. Ambekar, Cochair
Univ. of Washington, 1013 NE 40th St, Seattle, WA 98105

John A. Case, Cochair
Graduate Program in Acoustics, Penn State, 201 Applied Science Building, University Park, PA 16802

Marissa Garcia, Cochair
Natural Resources and the Environment, Cornell University, 159 Sapsucker Woods Road, Cornell Lab of Ornithology - K. Lisa Yang Center for Conservation Bioacoustics, Ithaca, NY 14850

All posters will be on display and all authors at their posters from 10:00 a.m. to 11:00 a.m.

Contributed Papers

4aID1. Acoustics at UMass Dartmouth. David A. Brown (Elec. and Comput. Eng., UMass Dartmouth, 151 Martine St., UMass Dartmouth, Fall River, MA 02723, dbAcousticsdb@gmail.com), John R. Buck, and Paul J. Gendron (Elec. and Comput. Eng., UMass Dartmouth, Dartmouth, MA)

The University of Massachusetts Dartmouth has a long history spanning 5 decades of graduate courses offerings and research in underwater acoustics, transduction and signal processing leading to M.S. and Ph.D. degrees in Electrical Engineering. Collaborations between Marine Science, Physics, Mechanical, Bioengineering, and Electrical Engineering departments offer many interdisciplinary opportunities in Acoustics. Courses include Fundamentals of Acoustics, Vibrations, Underwater Acoustics, Electroacoustic Transducers, Medical Ultrasonics, Sonar, Digital Signal Processing, Array Processing, Random Signals, Information Theory, Communications, Detection and Estimation. Many unique facilities including an Underwater Acoustic Test Tank, Open-Ocean water access, and Unmanned Underwater Vehicles support both undergraduate and graduate projects. Research focuses include transducers and transduction science, materials characterization, calibration, array and sonar signal processing, animal bioacoustics, communications, detection and estimation, active and passive sonar funded by Office of Naval Research and Industry.

4aID2. Graduate acoustics at Brigham Young University. Kent L. Gee, Micah Shepherd, Brian E. Anderson (Phys. & Astronomy, Brigham Young Univ., Provo, UT), Tracianne B. Neilsen (Phys. & Astronomy, Brigham Young Univ., N251 ESC, Provo, UT 84602, tbn@byu.edu), Matthew S. Allen, and Jonathan D. Blotter (Mech. Eng., Brigham Young Univ., Provo, UT)

Graduate studies in acoustics at BYU prepare students for industry, research, and academia by complementing in-depth coursework with publishable research. Coursework provides students with a foundation in acoustical principles, practices and measurement skills, including a experimental techniques and technical writing. Labs across the curriculum cover calibration, directivity, scattering, absorption, laser Doppler vibrometry, experimental methods for dynamic structures, lumped-element mechanical systems, equivalent circuit modeling, arrays, filters, room acoustics, active noise control, and near-field acoustical holography. Recent thesis and dissertation topics include active noise control, directivity, room acoustics, energy-based acoustics, time reversal, nondestructive evaluation, vibration and acoustics of aerospace vehicles, biomedical applications, flow-based acoustics, voice production, aeroacoustics, sound propagation modeling, nonlinear propagation, high-amplitude noise analyses, machine and deep learning applied to ambient noise level prediction, crowd noise interpretation, and underwater acoustic source localization, and ocean environment classification. Graduate students are expected to present research at professional meetings and publish in peer-reviewed acoustics journals. Graduate students often serve as peer mentors to undergraduate students on related projects and may participate in field experiments to gain additional experience. @BYUAcoustics

4aID3. The Graduate Program in Acoustics at Penn State. Andrew Barnard (Graduate Program in Acoust., Penn State, 201C Appl. Sci. Bldg., University Park, PA 16802, barnard@psu.edu) and Daniel A. Russell (Graduate Program in Acoust., Penn State Univ., University Park, PA)

The Graduate Program in Acoustics at Penn State offers graduate degrees (M.Eng., M.S., Ph.D.) in Acoustics, with courses and research opportunities in a wide variety of subfields. Our 820 alumni are employed around the world in military and government labs, academic institutions, consulting firms, and consumer audio and related industries. Our 40 + faculty from several disciplines conduct research and teach courses in structural acoustics, nonlinear acoustics, architectural acoustics, signal processing, aeroacoustics, biomedical ultrasound, transducers, computational acoustics, noise and vibration control, acoustic metamaterials, psychoacoustics, and underwater acoustics. Course offerings include fundamentals of acoustics and vibration, electroacoustic transducers, signal processing, acoustics in fluid media, sound and structure interaction, digital signal processing, experimental techniques, acoustic measurements and data analysis, ocean acoustics, architectural acoustics, noise control engineering, nonlinear acoustics, outdoor sound propagation, computational acoustics, biomedical ultrasound, flow induced noise, spatial sound and three-dimensional audio, and the acoustics of musical instruments. Distance education students pursuing the M.Eng. degree join resident students in a hybrid classroom environment. This poster highlights faculty research areas, laboratory facilities, student demographics, successful graduates, and recent enrollment and employment trends.

Session 4aMU

Musical Acoustics: General Topics in Musical Acoustics II

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

Contributed Papers

9:00

4aMU1. The applications of dynamic time warping in the source separation of percussive sounds. Christopher Grabow (Penn State, 205 Hallowell Bldg, State College, PA 16802, cng5270@psu.edu) and Tyler P. Dare (Penn State, State College, PA)

Music source separation (MSS) is the process of splitting various components of a musical piece into individual tracks. This process combines the fields of acoustics and machine learning to extract useful data from music, which assists in a variety of music information retrieval tasks. In the past decade, many methods have been employed to perform MSS with varying levels of success. This research explores the use of dynamic time warping (DTW) for MSS tasks in the time domain. DTW is an algorithm that performs a temporal alignment of two time series to measure their similarity. It is unique in that the algorithm will minimize the Euclidean distance between the two sequences by stretching or compressing them to optimize similarity. This makes DTW a distinctive method for MSS, as it operates entirely in the time domain and classifies sounds without the interference of time warping. The research performed focuses only on the separation of transient, percussive sounds. Measurements taken with a drum kit and a selection of digital drum sounds served as the foundation for tests of the algorithm. The results of this research illustrate the potential of DTW in time domain MSS applications.

9:15

4aMU2. An investigation into the directivity, spectra, and modes of a Tambourine. Emma Todd (Brigham Young Univ., 5 Heritage Halls #2204, Provo, UT 84602, et444@byu.edu), Hanna Pavill, Jacob B. Hales, Matt Coleman, and Micah Shepherd (Brigham Young Univ., Provo, UT)

As an instrument, the tambourine has been used by various cultures for thousands of years. However, despite this rich history, there remains little literature that investigates the acoustical properties of the tambourine itself and an even greater lack in research that moves beyond discussing the instrument's membrane alone. As a tambourine's membrane is only one part of a larger whole, this is problematic; in fact, the membrane is not usually considered the instrument's main radiator as many tambourines completely lack a membrane and players often actively damp the membrane while playing. Therefore, in order to develop a more in-depth and foundational understanding of the tambourine's physics, we will discuss the nature of the instrument's directivity, analyze how its sound spectra varies across diverse playing styles, and present the various modes associated with its wooden frame and metal jingles. The modes have been estimated using finite element techniques.

9:30

4aMU3. The sound radiation and directivity of glockenspiel bars. Hanna Pavill (Brigham Young Univ., N284 ESC, Provo, UT 84602, pavillh@byu.edu) and Micah Shepherd (Brigham Young Univ., Provo, UT)

The directivity of a musical instrument describes the predominant directions that sound radiates from that instrument when it is being played. The shape of an instrument, as well as the note and playing style, can greatly affect its sound radiation and directivity. The glockenspiel is a percussion instrument made of pitched metal bars of uniform thickness laid out in a keyboard pattern and set on a frame. It produces sound when the player strikes the bars with a mallet. Uniform beam theory is often used to describe the modal behavior of the individual glockenspiel bars. However, when a single bar is stuck, the other surrounding bars and support frame influence the sound radiation and directivity. Therefore, glockenspiel measurements were performed using a directivity measurement system which has previously been used to measure musical instrument directivity. To eliminate strike-to-strike variance, an automatic striking device was used. This work will compare the sound radiation and directivity of a glockenspiel bar in the standard configuration, to an individual glockenspiel bar outside of the frame.

9:45

4aMU4. Further investigation of the vibration characteristics and sound radiation of Balinese gamelan gongs. Dallin T. Harwood (Brigham Young Univ., N283 ESC, Provo, UT 84602, dth37@byu.edu), Hanna Pavill, Emma Todd, Samuel D. Bellows, and Micah Shepherd (Brigham Young Univ., Provo, UT)

Balinese gamelan gongs are percussion instruments of special interest because of their unique shape and sound. Unlike a Chinese tam-tam, the gongs are thick and deep, with a protruding dome in the center and long edges that sharply wrap around the circumference of the gong. When struck, the larger gongs are designed to produce a strong beating pattern. Recent work in both the analytical modeling and high-resolution measurements of the gong has advanced our understanding of the physics of this suite of instruments. However, there is still much to be understood about the influence of the shape of the gong and what boundary conditions to expect from an instrument of the dimensions described. This paper will present measurements and discussion of the nature of both the gong's boundary conditions and its back cavity. Both will be used to support the continued development of theoretical models of the gong.

10:00–10:15 Break

10:15

4aMU5. The development of an automated striking device for repeatable percussion strikes. Jacob Sampson (Utah Valley Univ., 800 W University Parkway, Orem, UT 84058, jacobbsampson@gmail.com), Hanna Pavill, and Micah Shepherd (Brigham Young Univ., Provo, UT)

Many high-quality directivity systems involve a single microphone arc which rotates around a sound source. Since multiple data captures are necessary as the arc rotates around a sound source, the source should emit the sound in a steady and repeatable way at all arc positions. To obtain directivity measurements of a struck percussion instrument, it is crucial to be able to strike the instrument during each microphone arc position as consistently as possible. However, a human player typically cannot strike the instrument in the exact same way for many consecutive measurements. To overcome the shortcomings of human players, an automatic striking device was developed to aid in the study of the directivity of percussion instruments. The device consists of a striking implement holder attached to a stepper motor that

is controlled by an Arduino. The striking implement holder allows the user of the striking device to use different-sized mallets/drumsticks depending on the instrument that is being measured. The automatic striking device has been used to obtain directivity measurements on several instruments including a glockenspiel, cymbal, and drum. The details of the automatic striking device will be presented along with a subset of directivity results.

10:30

4aMU6. Ultrasound tongue imaging of vowel spaces across pitches in singing. May Pik Yu Chan (Linguist., Univ. of Pennsylvania, Dept. of Linguist 3401-C Walnut St., Ste. 300, C Wing University of Pennsylvania, Philadelphia, PA PA 19104-6228, pikyu@sas.upenn.edu) and Jianjing Kuang (Linguist., Univ. of Pennsylvania, Philadelphia, PA)

One important technique in singing is vowel modification: the adjustment of the resonance space based on the sung pitch for more efficient voice production. We explore whether vowel modification is a learned technique for enhanced acoustics, or if it is a necessary articulatory adjustment for high pitch production. 16 participants without vocal training participated in a singing experiment with ultrasound tongue imaging. Participants were asked to sing sets of English vowels across their comfortable pitch range rising by semitone in a steady tempo, resembling a vocal warm up exercise. Participants sang 5 sets of vowels in total, each set consists of 5 target vowels ([i], [ɛ], [æ], [ɑ], [u]) in randomized order with 1 filler ([ɔ]) closing each breath group. Images of tongue position were splined using DeepLabCut. Preliminary results show that untrained singers tend not to adjust their tongue positions by pitch, though cases of tongue lowering occasionally occurred, particularly for the participants who sing a wide pitch range. In contrast, additional pilot data from 2 trained operatic singers showed gradual tongue adjustments across their pitch range, neutralizing vowel contrasts at their highest pitches. We discuss findings with respect to vowel-pitch interaction, drawing implications on theories of voice production.

10:45

4aMU7. Analysis and interpretation of complex vibrato patterns: A novel parametric approach to genre-specific singing performance. Theodora I. Nestorova (Schulich School of Music, McGill Univ., 555 Sherbrooke St. W., Montréal, QC H3A 1E3, Canada, theodora.nestorova@mail.mcgill.ca), Gary Scavone (Music Res., McGill Univ., Montreal, QC, Canada), Ian Howell (Ann Arbor, MI), and Josh Gilbert (READS Lab, Harvard Univ., Boston, MA)

A complex, under-researched phenomenon, vibrato exists in a variety of musical styles and genre contexts. However, currently accepted acoustical analysis methods presume the vibrato is uniform, consistent, and persistent. This Western Art Music lens disregards significant stylistic characteristics of many genres with non-normative vibrato features. Therefore, a new system considering both regularity and shape of vibrato metrics in many genres over time is essential. 2 cross-genre songs and 1 exercise task were disseminated to 15 professional Operatic, Musical Theater, and Jazz singers. 16 pitch segments from each singer were subjected to sinusoidal extraction, $f_{;o}$ band-pass filtering, and an FFT LTAS in Praat. Each sample's mean half extent, pitch, and vowel was calculated and assessed using standard deviation, Coefficient of Variation (CV%), linear/polynomial/non-linear regression techniques in R. The results, corroborated by a perceptual survey distributed to professional singing pedagogues, indicated that vibrato variability predictably distinguishes performed genres. The CV% well characterizes vibrato variability and is higher in the Musical Theater and Jazz singing samples. A novel model – 4 parameter logistic s-curve regression – is proposed as optimal representation of multi-phasic vibrato with complex shapes. Such novel vibrato models may be employed to acoustically examine complex vibrato patterns in many instruments beyond singing.

11:00

4aMU8. Short-term retention of popular music in older adults: Support for a plasticity theory of implicit music knowledge acquisition. Annabel J. Cohen (Psych., Univ. of PE, Dept. of Psych., University of PE, 550 University Ave., Charlottetown, PE C1A 4P3, Canada, acohen@upei.ca), Corey Collett (Psych., Univ. of PE, Charlottetown, PE, Canada), and Kristen Gallant (Psych., Univ. of Waterloo, Waterloo, ON, Canada)

Based on our Plasticity Theory of Implicit Music Knowledge Acquisition (PTIMKA), we tested the hypothesis that adolescence is a sensitive period for acquiring musical information and lifelong musical grammar. Older adults ($N = 27$, mean age = 65.7 years; SD = 6.7) identified artist, title, and year of popularity and rated their familiarity for short excerpts of 36 songs popular between 1962 and 2021. Knowledge and familiarity were greater for music popular during the participants' adolescence. A subsequent surprise retention task required participants to choose which of 2 excerpts had been presented in the first task; for each of 36 trials, targets and foils represented the same era of popularity. Retention (dõ) scores and confidence in retention judgment were higher for the songs popular during adolescence, even though targets of all eras had just been presented in the previous 15 minutes. It is argued that popular music styles congruent with an adolescence-established grammar were more accurately encoded than styles violating this grammar, as would songs popular before or after adolescence. Prior data from younger adults showing trajectories opposite to those for older adults as a function of decade of popularity are further consistent with this interpretation and with PTIMKA. (Work supported by NSERC)

Session 4aNS

Noise, Physical Acoustics and Engineering Acoustics: Methods for Community Noise Testing and Analysis I

Alexandra Loubeau, Cochair
NASA Langley Research Center, MS 463, Hampton, VA 23681

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

Duncan Halsead, Cochair
Aercoustics Engineering Ltd, 1004 Middlegate Rd, Mississauga, L4Y 0G1, Canada

Chair's Introduction—8:30

Invited Papers

8:35

4aNS1. Dose error correction using simulation extrapolation for community noise dose-response modeling. Aaron B. Vaughn (NASA Langley Res. Ctr., 2 N Dryden StMS 463, Hampton, VA 23681-2199, aaron.b.vaughn@nasa.gov), Nathan B. Cruze, Matthew Boucher, and William Doebler (NASA Langley Res. Ctr., Hampton, VA)

The objective of this work is to provide a framework to account and correct for dose error in dose-response modeling due to measurement uncertainty. Error in noise measurements, especially in the case of limited monitoring locations in a community, can lead to an attenuation or misestimation of parameters in dose-response models. This error can result in overpredicted annoyance at lower doses and underpredicted annoyance at higher doses. Simulated data in the present work are based on previous NASA community studies and incorporate a notional design for future studies with the X-59 aircraft. Several populations of different annoyance response sensitivities are included. Simulation extrapolation (SIMEX) is used to correct for the dose error in a simple, fully pooled logistic regression. Results indicate the negative impact of attenuation is greatly diminished for all amounts of dose error considered, regardless of a population's annoyance sensitivity. Therefore, SIMEX can help produce a more accurate dose-response relationship.

8:55

4aNS2. The influence of atmospheric variability on predicted sonic boom metrics. Shane V. Lympany (Blue Ridge Res. and Consulting, 29 N Market St., Ste. 700, Asheville, NC 28801, shane.lympany@blueridgeresearch.com) and Juliet A. Page (Blue Ridge Res. and Consulting, Asheville, NC)

As part of the Quesst mission, NASA will fly the X-59 aircraft over selected communities to survey community responses to low sonic booms. Previously, we developed a Kalman filter method to estimate the loudness metrics experienced by each survey participant during each flight. The Kalman filter fuses acoustic measurements with predictions from PCBoom, a sonic boom propagation model. PCBoom requires vertical profiles of the temperature, humidity, and wind to propagate sonic booms through the atmosphere and predict the loudness metrics at the ground. Prior to each X-59 flight, NASA will launch weather balloons to measure the vertical atmospheric profile. However, the atmosphere changes continually with geographic location and time. The purpose of this work is to determine where and when to launch these weather balloons to minimize the uncertainty in the predicted loudness metrics caused by atmospheric variability. We analyzed the effect of atmospheric variability on the predicted Perceived Level in three communities in different climate zones in the United States. To achieve acceptable uncertainty in the Perceived Level, weather balloons should be launched from at least two different sites within the survey area within one hour of each X-59 flight.

9:15

4aNS3. Turbulence-induced variability of a far-field Falcon-9 sonic boom measurement. Kaylee Nyborg (Dept. of Phys. and Astronomy, Brigham Young Univ., BYU Dept. of Phys. and Astronomy, N284 ESC, Provo, UT 84602, kaylee.nyborg@byu.net), Mark C. Anderson, and Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Although aircraft sonic booms have been researched for decades, sonic booms from rocket booster landings are relatively recent events that are less studied. However, as booster landings become more prevalent, a better understanding of the associated sonic booms and community impact is needed. Atmospheric turbulence distorts sonic boom waveforms, affecting both the rise time and peak pressure, and can have a large impact on human perception metrics. As an initial investigation into the impact of turbulence on sonic boom

measurements, measurements were made of the reentry sonic boom during the SpaceX Falcon-9 SDA Tranche 0B mission at Vandenberg Space Force Base, California. An 11-microphone linear array spanning 150 m (500 ft) recorded the boom 8.5 km from the landing pad. This paper discusses results from this measurement, including variability of human perception metrics. For example, Perceived Level varied by up to 8 dB across the array. [Work supported through a NASA Graduate Research Fellowship.]

Contributed Papers

9:35

4aNS4. Development of a weather robust microphone configuration for sonic boom measurments. Jesse Blaine (Phys., Brigham Young Univ., C110 ESC, Brigham Young University, Provo, UT 84602, jlatch@byu.edu), Mark C. Anderson, and Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

This paper discusses ongoing development and testing of ground-based, weather-robust microphone measurement systems in conjunction with preparation for community testing of NASA's X-59 low-boom aircraft. Prior efforts [Anderson *et al.*, Proc. Mtgs. Acoust. **42**, 040005 (**2022**)] resulted in the refinement of a ground-plate setup by investigating varied windscreen and plate diameter and thickness. The most recent goal has been to make the setup more compact and easier to manufacture. This paper discusses the results of additional tests performed on updated designs meant to find the balance between compactness and performance. Anechoic chamber testing was performed to show ground plate and windscreen performance and to compared results against prior versions. Outdoor tests included measurements during different wind conditions and over different ground surfaces to examine low-frequency wind noise reduction and ground impedance effects. Results discussed during the presentation suggest the new design strikes an acceptable compromise between compactness, manufacturing ease and robustness, and performance. [Work supported by NASA Langley Research Center through Analytical Mechanics Associates]

9:50

4aNS5. Investigating sonic booms using impulse metrics. Avery K. Sorrell (Dept. of Phys. and Astronomy, Brigham Young Univ., N281 Eyring Sci. Ctr., Provo, UT 84602, averysorrell137@gmail.com), Mark C. Anderson, Kaylee Nyborg, and Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

As an impulsive sound, a sonic boom exhibits traits similar to other transient acoustic impulses, such as a single strike of a sledgehammer strike or an exploding firework. Although several metrics exist to assess the nature and potential acoustic hazards of these other impulsive sounds, these metrics are rarely applied to sonic booms. This paper investigates the use of several metrics commonly applied to impulsive sounds, such as B and D durations or the kurtosis of the pressure waveform, for sonic booms recorded during previous NASA flight test campaigns, including CarpetDIEM (Carpet Determination in Entirety Measurements) and QSF18 (Quiet Supersonic Flights 2018). Understanding the behavior of these metrics in urban (QSF18) and rural (CarpetDIEM) environments adds to the larger body of knowledge concerning sonic boom measurements and properties and may be useful in quantifying sonic boom variability in communities.

10:05–10:20 Break

Invited Paper

10:20

4aNS6. Abstract withdrawn.

Contributed Papers

10:40

4aNS7. Assessment of noise complaints as an indicator of long-term effects of aircraft noise on communities surrounding aerodromes. Colin Novak (Mech. Automotive and Mater. Eng., Univ. of Windsor, 401 Sunset Ave., Windsor, ON N9B 3P4, Canada, novak1@uwindsor.ca) and Julia Jovanovic (Mech. Automotive and Mater. Eng., Univ. of Windsor, Windsor, ON, Canada)

Aircraft noise is considered by many as the most burdensome part of aircraft operations. Communities neighbouring airports often express concerns about the possible health effects of chronic exposure to noise. To quantify the long-term effects of environmental noise exposure, researchers often use the metric of annoyance. Annoyance is widely considered to be the most well-corroborated health effect of aircraft noise and a moderating factor for other suspected health effects. Annoyance can also be correlated to average cumulative noise levels, such that higher levels of chronic noise likely evoke higher levels of annoyance within the population. Annoyance data is typically collected using extensive surveys and/or interviews, which are costly and time-consuming. In the absence of annoyance data, complaints are often used as a proxy for annoyance. This research used complaint, noise and annoyance data to demonstrate that complaints do not equate to annoyance, nor do they correlate to cumulative noise metrics. Thus, while complaint data may prove useful in analyzing short-term response to operations, it should not be relied upon for the assessment of long-term impacts from aircraft noise exposure, nor should it be used to direct noise and annoyance mitigation initiatives.

10:55

4aNS8. Aircraft noise contours—From everything for everyone to nothing for anyone. Colin Novak (Mech. Automotive and Mater. Eng., Univ. of Windsor, 401 Sunset Ave., Windsor, ON N9B 3P4, Canada, novak1@uwindsor.ca) and Julia Jovanovic (Mech. Automotive and Mater. Eng., Univ. of Windsor, Windsor, ON, Canada)

Aircraft noise exposure contours were originally intended as a tool to facilitate compatible land use in the vicinity of airports. The simple concept was for authorities to use historic data and reasonable predictions for near future demands to create a map that identified areas of high aircraft noise impacts. Municipalities and planners would refer to this map to determine appropriate zoning around the airport, restricting noise sensitive development in high noise exposure areas. With time, different demands and applications of the noise contours emerged. Some stakeholders demanded longer term forecasts to allow for planning farther into the future. Some co-opted contours as a PR tool to suggest reduced impacts on communities. Some began using noise contours as a tool to protect against encroachment. Others began to look to noise contours as a representation of daily acoustic conditions in areas surrounding the airport. In Canada, a lack of oversight and guidance for the selection of inputs parameters for noise models, make it

such that noise contours have become a product illustrating whatever their creator intends, lacking objectivity and scientific rigour. This research demonstrates how noise contours can be designed to achieve desired results by modulating different input parameters.

11:10

4aNS9. What pandemic era travel restrictions revealed about aircraft noise, complaints and annoyance—SONICC 2020. Julia Jovanovic (Mech. Automotive and Mater. Eng., Univ. of Windsor, 401 Sunset Ave., Windsor, ON N9B 3P4, Canada, jovano11@uwindsor.ca) and Colin Novak (Mech. Automotive and Mater. Eng., Univ. of Windsor, Windsor, ON, Canada)

At the start of COVID-19 travel restrictions, Toronto Pearson International Airport experienced an approximate 80% reduction in traffic. This gave an unprecedented opportunity to investigate the impacts that a drastic reduction in aircraft noise would have on the communities surrounding the airport. Using the results of the Survey of Noise Impacts on Canadian Communities (SONICC) distributed in the summer of 2020, this research evaluated pre-pandemic and amidst pandemic aircraft noise annoyance in neighbourhoods surrounding Pearson Airport. The research investigated the effects of air traffic reduction on noise levels, complaint behaviour and annoyance. Complaint volumes correlated closely to the number of operations, experiencing a significant reduction. Despite the notable reduction in complaints, many complainants continued to vigorously complain, and some locations even experienced an increase in complaints. Pre-pandemic compared to amidst pandemic annoyance experienced reductions proportional to the average reduction in noise exposure. Despite significant reductions in noise, 33% of pre-pandemic highly annoyed (HA) respondents, remained highly annoyed, suggesting that anything short of a complete halt of operations would result in severe annoyance amongst a portion of the population.

11:25

4aNS10. Antiquated Canadian Aircraft Noise Guidance—Setting the grounds for encroachment and adverse community impacts. Julia Jovanovic (Mech. Automotive and Mater. Eng., Univ. of Windsor, 401 Sunset Ave., Windsor, ON N9B 3P4, Canada, jovano11@uwindsor.ca) and Colin Novak (Mech. Automotive and Mater. Eng., Univ. of Windsor, Windsor, ON, Canada)

A comprehensive research project at the University of Windsor entitled *Prediction and Management of Aircraft Noise Annoyance Around Canadian Airports* reviewed multiple components of Canada's TP 1247 Land Use in the Vicinity of Aerodromes. Using noise, complaints and survey data from Toronto Pearson International Airport, researchers evaluated the Noise Exposure Forecast (NEF) system as a tool for aircraft noise annoyance prediction and management. The NEF metric and its correlation to annoyance was examined as well as applicability of the NEF 30 threshold for the onset of significant annoyance. Further, the guideline for expected community response to noise was tested. The research found that most components of the NEF system are antiquated and in need of revision. A lack of prompt action to update Canadian guidelines for aircraft noise is setting the grounds for conflicts between stakeholders with competing interests and increasing the risks for greater noise impacts on future communities.

11:40

4aNS11. Investigation of speed and altitude effects on sound exposure level calculations for multiple helicopters. Mary L. Houston (NASA Langley Res. Ctr., NASA Langley Res. Ctr., M.S. 461, Hampton, VA 23601, Mary.L.Houston@NASA.gov), Kyle Pascioni (NASA Langley Res. Ctr., Hampton, VA), and James Stephenson (DEVCOM AvMC, Hampton, VA)

International and United States Federal regulations evaluate helicopter acoustic emissions for certification purposes as well as community noise impact assessments. The Sound Exposure Level (SEL) is a commonly used metric that represents the loudness of a single event, with a penalty for duration. For example, a flyover lasting ten seconds can have the same SEL as a five-second duration flyover that is 6 dB higher in peak level. Ten helicopters spanning the light and medium weight classes were modeled using flight test data. They were then virtually flown using the Advanced Acoustic Model at multiple speeds and altitudes. SEL and A-weighted SEL (SELA) were calculated at ground-based observer locations for each condition. Scaling laws with altitude are determined, and cruising airspeeds which result in the minimum SEL are found. Implications for future land-use planning will be discussed, and the relationships are reduced to general or vehicle-specific operational guidance for pilots.

Session 4aPA

Physical Acoustics and Signal Processing in Acoustics: Acousto-Optic Sensing

Carl R. Hart, Cochair
U.S. Army Engineer Research and Development Center, Cold Regions Research and Engineering Laboratory, 72 Lyme Rd., Hanover, NH 03755

Oleg A. Sapozhnikov, Cochair
University of Washington, 1013 NE 40th Street, Seattle, WA 98105

Yangfan Liu, Cochair
Purdue University, Ray W. Herrick Laboratories, 177 S, Russell Street, West Lafayette, IN 47907

Chair's Introduction—8:00

Invited Papers

8:05

4aPA1. Experimental studies with parallel phase-shifting interferometry. Yasuhiro Oikawa (Waseda Univ., Tokyo, Japan, yoikawa@waseda.jp), Risako Tanigawa (Waseda Univ., Tokyo, Japan), Mariko Akutsu (Railway Tech. Res. Inst., Tokyo, Japan), Kenji Ishikawa (NTT, Atsugi, Japan), and Denny Hermawanto (National Res. and Innovation Agency, Jakarta, Indonesia)

One of the most popular of devices currently used for acousto-optic sensing is the laser Doppler vibrometer (LDV), an interferometer originally designed to measure the vibration velocity of objects. LDVs are a popular way to measure and visualize acoustic fields, however it requires scanning the laser, which limits the measurement sound field. Recently, as the camera captures 2D images of the sound field on thousands of pixels simultaneously, the sound fields that can be measured are no longer limited to those that can be repeated. A high-speed camera captures 2D sound fields with tens or hundreds of thousands of frames per second, making it possible to film a slow-motion video of propagating sound in real time. The parallel phase-shifting interferometry (PPSI) with polarized high-speed cameras has been developed and introduced in the acoustics field as well. It has demonstrated impressive visualizations of airborne acoustic phenomena due to its high sensitivity and spatiotemporal resolution. It has also found many applications in recent years. This paper presents our experimental studies, especially measurement of aerodynamic sound, sound radiated from fast-moving sources or musical instruments, determination of acoustic center, etc.

8:25

4aPA2. Optical method for *direct* temperature measurement of high intensity focused ultrasound heating. Ghanem Oweis (Mech. Eng., American Univ. of Beirut, Bliss St., AUB, Bechtel Bldg. 408, Beirut 1107-2020, Lebanon, goweis@aub.edu.lb) and Hussein Daoud (Mech. Eng., American Univ. of Beirut, Beirut, Lebanon)

The acoustic pressure field or heat deposition from the passage of an ultrasonic wave can produce a perceptible alteration in the medium's optical refractive index. With proper placement of the light source and imaging camera, the optical effect can be recorded. This talk describes an optical temperature measurement methodology dubbed laser-ray-bundle (LRB) of the heat deposition from a high intensity focused ultrasound (HIFU) transducer in a transparent tissue phantom. The method does *not* require calibration, and *directly* converts the thermo-optic signal to temperature without recourse to any iterative computations. LRB boasts of excellent temporal and spatial resolutions and high temperature sensitivity better than any fine wire thermocouple. Furthermore, it is completely non-invasive. The LRB method is demonstrated with the heat deposition from single millisecond long HIFU pulses in a PDMS tissue phantom or from a train of single pulses with very good performance. The method represents a new capability for quickly assess a HIFU transducer output, and for laboratory and preclinical studies of heating near tissue interfaces including bone, air, or fat. A preliminary experimental setup for measuring the acoustic pressure field is presented.

8:45

4aPA3. Deeper, faster, and colorful photoacoustic imaging in life sciences. Junjie Yao (Duke Univ., 100 Sci. Dr., Hudson Hall Annex 261, Durham, NC 27708, junjie.yao@duke.edu)

Photoacoustic imaging (PAI) is an increasingly powerful technique for multi-scale anatomical, functional, and molecular imaging by acoustically detecting the optical absorption contrast in biological tissues. I will focus on several technological advancements in PAI that have collectively enabled fast, deep, and high-sensitivity biomedical applications and discoveries in life sciences. First, PAI has overcome the penetration limit by utilizing advanced internal light delivery techniques, allowing for super-deep (>10 cm) imaging. This

breakthrough has extended the applicability of PAI to internal organ imaging in large animal models and humans. Second, innovative scanning technologies and deep-learning models have significantly accelerated PAI, enabling imaging speeds that are more than 1000 times faster while maintaining a large field of view and high spatial resolution. This enhancement facilitates the monitoring of highly dynamic biological processes at the microscopic scale, such as functional brain activities and glassfrog transparency. Lastly, PAI has greatly benefited from the genetically encoded switchable or tunable near-infrared photoacoustic-specific probes. By incorporating these probes, the sensitivity and specificity of PAI have been improved by more than 1000 times, enabling highly sensitive detection of malignant cancer, tissue hypoxia, and neuronal activities.

9:05

4aPA4. On the sensitivity and noise of acousto-optic sensing: Exploring the detection limits. Kenji Ishikawa (NTT, 3-1 Morinosato-Wakamiya, Atsugi 243-0198, Japan, ke.ishikawa@ntt.com)

Acousto-optic sensing has garnered increasing attention in recent years due to its non-contact nature, gaining importance in a range of acoustic measurement applications. Although its potential is widely recognized, the signal-to-noise ratio often requires enhancement in practical applications, which can impede further development. While more sensitive measurement methods have been evolving, the discussion around noise and detection limits in acousto-optic sensing remains limited. This presentation will highlight the author's recent findings on sensitivity and noise in acousto-optic sensing. It aims to illuminate the achievable detection limits through a detailed analysis of optical and acoustic noises. The insights provided will offer a deeper understanding of the complexities involved and potentially guide future advancements in this field.

9:25

4aPA5. Acousto-optic sensing: A scattering interpretation. Samuel A. Verburg (Dept. of Elec. and Photonics Eng., Tech. Univ. of Denmark, Ørsteds Plads, Bldg. 352, Kgs. Lyngby 2800, Denmark, saveri@dtu.dk) and Efren Fernandez-Grande (Dept. of Elec. and Photonics Eng., Tech. Univ. of Denmark, Kgs Lyngby, Hovedstaden, Denmark)

Acousto-optic sensing consists in using (laser) light as the sensing element in order to measure, visualise and study acoustic phenomena in a remote, non-invasive manner. One of the most common sensing techniques involves measuring acoustically-induced phase shifts of an optical probing beam by means of interferometry. Arguably, the main fundamental limitation of such sensing technique is that it provides projections of the acoustic field, instead of acoustic quantities at specific locations. In this study we examine the theory of spontaneous light scattering in transparent media, and use this framework to formulate a general acousto-optic sensing problem. Preliminary results based on numerical simulations indicate that light scattering could be applied to the measurement of acoustic pressure fluctuations. Some of the main challenges associated with the principle—mainly related to the weaknesses of the acousto-optic effect in air and relatively low signal-to-noise ratio—as well as the main fundamental differences from conventional interferometry techniques are discussed in this study.

9:45–10:00 Break

10:00

4aPA6. Sensing viscous acoustic flow: Using spider silk to hear. Jian Zhou, Junpeng Lai (Mech. Eng., Binghamton Univ., Vestal, NY), and Ronald Miles (Mech. Eng., Binghamton Univ., 85 Murray Hill Rd., Mech. Eng. Dept, Binghamton, NY 13902, miles@binghamton.edu)

Measurements obtained with a laser vibrometer show that a single strand of spider silk captures sound with almost full fidelity across a broad frequency range from infrasound to ultrasound, surpassing the performance of any known microphone or ear. The high responsivity of the silk to acoustic particle velocity enables the orb-weaving spider to hear sounds from meters away by utilizing its web—a hearing mechanism distinct from that of other animals that rely on an organ within their body. Laser-based measurements have also revealed that small creatures like mosquitoes can hear far-field sounds, similar to the spider, by sensing acoustic particle velocity rather than pressure. These findings offer insights for designing novel acoustic flow detectors and have led to the development of a recently commercialized flow microphone.

10:20

4aPA7. Laser interferometer measurement and reconstruction of supersonic projectile acoustic pressure fields. Matthew G. Blevins (U.S. Army Engineer Res. and Development Ctr., 2902 Newmark Dr., Champaign, IL 61822, matthew.g.blevins@usace.army.mil) and Gregory W. Lyons (DEVCOM Army Res. Lab., Adelphi, MD)

Sensing acoustic fields with a laser interferometer offers several advantages over conventional techniques: the probe beam does not disturb the medium in which it operates, the interferometer optics can be placed far from the source, and interferometry typically has a wider bandwidth than microphones. These advantages are particularly useful in situations where precise measurements of high-amplitude shock-generating sources are required. In this work we use a laser interferometer to study the generation and propagation of shock wave signatures of ballistic projectiles. A laboratory experiment was conducted to collect interferometer data from a variety of supersonic projectiles over a range of standoff distances. Simultaneously-captured microphone data is compared to the interferometer measurements, and techniques for inversion and reconstruction of the sound pressure field are discussed.

10:40

4aPA8. Deconvolution of spatially averaged shock wave forms. Carl R. Hart (U.S. Army Engineer Res. and Development Ctr., Cold Regions Res. and Eng. Lab., 72 Lyme Rd., Hanover, NH 03755, carl.r.hart@erdc.dren.mil) and Michael J. White (U.S. Army Engineer Res. and Development Ctr., Construction Eng. Res. Lab., Champaign, IL)

Noncontact optical methods (e.g., shadowgraphy, schlieren, and interferometry) allow visualization of propagating shock waves, avoid sensor-field interaction, and retain the potential for very high bandwidth. The heterodyne Mach-Zehnder interferometer produces quantifiable noncontact samples of the accumulated phase shifts induced by a shock wave, as viewed through its beam projection. This instrument is commercially available as a Laser Doppler Vibrometer (LDV). For shocked wave fields in air obeying spherical symmetry, point values for density and overpressure can be inferred. An important shortcoming for inference quality is the finite spatial dependence of the probe beam cross section, the profile shape itself serving to limit both the spatial resolution and the measurement bandwidth. We propose to model both of these effects of spatial averaging by constructing a model waveform and a model beam profile, convolving them, and comparing the result to synthesized LDV output. Mismatch between the convolved result and spatially averaged measurement can be quantified by a cost function, which is minimized via gradient descent.

10:55

4aPA9. Microphone diaphragm thermal noise floor measurement using a laser vibrometer. Morteza Karimi, Junpeng Lai (Mech. Eng., Binghamton Univ., Vestal, NY), Johar Pourghader (Mech. Eng., Binghamton Univ., Binghamton, NY), and Ronald Miles (Mech. Eng., Binghamton Univ., 85 Murray Hill Rd., Mech. Eng. Dept, Binghamton, NY 13902, miles@binghamton.edu)

The response of a microphone to thermal noise can be extremely difficult to predict and can adversely affect the performance of a given design. The direct measurement of the thermal noise floor of a microphone diaphragm could greatly improve our understanding of the parameters that affect the noise performance. A laser vibrometer can serve as a powerful tool, enabling precise and non-intrusive measurements of the diaphragm thermal noise. The experimental setup for measuring the diaphragm thermal noise using a laser vibrometer is described. Having both the noise floor, and the response to sound for a given microphone, the sound pressure-referred noise floor can be estimated. Measured and predicted results are presented for a variety of microphones. These measured results describe how design parameters, such as the size of the diaphragm, the dimensions of the backside volume, and overall geometry, impact the thermal noise floor measurements.

11:10

4aPA10. Development of an arrayed ultrasonic sensor using surface plasmon resonance (SPR). Kota Dezao (Doshisha Univ., 1-3 Tataramiyakotani, Kyotanabe-shi, Kyoto-fu 610-0321, Japan, kouta.nakiri@gmail.com), Shouta Kitajima, and Mami Matsukawa (Doshisha Univ., Kyotanabe, Kyoto, Japan)

Surface plasmon resonance (SPR) ultrasonic sensors are non-resonant and expected to show flat frequency response over a wide frequency range. Then, the sensors seem suitable for the detection of photoacoustic waves. We have developed a simple glass prism SPR sensor with thin Ag layer (deposited by an electron beam evaporator) and checked the frequency response as a function of Ag area diameter. The diameter of 1 mm was sufficient to obtain better frequency response than that of the calibrated needle transducer in the range of 2.5 to 7.0 MHz. By covering a very thin Au film on the Ag layer, we succeeded in longer working time (>6 months) of the sensor. An array-type SPR sensor (5×5, diameter of each sensor 1 mm) has also been fabricated to evaluate pressure distribution of ultrasonic waves. The response of each sensor was similar, telling the high reproducibility of the sensor fabrication.

11:25

4aPA11. Sound-field image denoising using deep neural network considering physical characteristics of sound. Daisuke Urata (Waseda Univ., 2-19-3, Haruhino, Asao-ku, Kawasaki, Kanagawa 215-0036, Japan, daisuke.u.0530@gmail.com), Yasuhiro Oikawa (Waseda Univ., Tokyo, Japan), and Kenji Ishikawa (NTT, Atsugi, Japan)

This paper presents a method of denoising utilizing deep neural networks (DNN) in sound field images measured by optical methods. Optical sound measurement has attracted much attention because of its capability to visualize sound field with high spatial resolution from a distance, which is difficult with conventional microphones. However, optical methods have the issue of the measured sound-field images often being heavily distorted by noise. This is caused by the weak phase of light changed by sound. Conventionally, noise-reduction filters considering the physical properties of sound have been known to be used for sound-field denoising. In this background, we propose a DNN-based sound-field denoising method that takes into account the physical properties of sound. In the DNN, a network architecture originally used for denoising natural images is employed, and we integrate loss functions based on the Helmholtz equation. Additionally, a 2D sound field dataset obtained from numerical acoustic simulation with random parameters is used during the training. Numerical and experimental data comparison experiments showed that the proposed DNN-based sound-field denoising outperformed the previous non-DNN methods.

11:40

4aPA12. Acoustic ground effects simulations from asteroid disruption via the "Pulverize It" method. Brin Bailey (Phys., Univ. of California, Santa Barbara, University of California, Santa Barbara, Broida Hall, Santa Barbara, CA 93106, brittanybailey@ucsb.edu), Alexander N. Cohen, Philip Lubin (Phys., Univ. of California, Santa Barbara, Santa Barbara, CA), Darrel K. Robertson (NASA Ames Res. Ctr., Moffett Field, CA), Mark Boslough (Univ. of New Mexico, Albuquerque, NM), Sasha Egan (Energetic Mater. Res. and Testing Ctr., New Mexico Inst. of Mining and Technol., Socorro, NM), and Elizabeth Silber (Sandia National Labs., Albuquerque, NM)

Our simulations show that PI ("Pulverize It"), a NASA Phase II NIAC study, is an effective multi-modal approach for planetary defense that can operate in extremely short interdiction modes (with intercepts as short as hours prior to atmospheric entry) as well as long interdiction time scales with months to years of warning. The basic process is complete disruption of the threat via fragmentation. In long warning time cases, the fragment cloud spreads enough to miss Earth, resulting in no ground effects. In cases where the warning time is short, the fragments (typically <10 m in diameter) will enter Earth's atmosphere where their energy is dissipated in a series of ground-level optical pulses and de-correlated shock waves, mitigating any significant damage. We investigate the acoustic ground effects through a set of simulation codes that model the interaction of asteroid fragments with the Earth's atmosphere following threat interception. Even in the short warning time cases where the fragments enter the atmosphere, our simulations show that threats mitigated by the PI method produce vastly less damage on the ground when compared to the same unfragmented case, yielding shock wave over-pressures under 3 kPa. Our simulations support the proposition that threats sized 20 m—1000 m in diameter can be effectively mitigated through fragmentation, resulting in acoustic ground effects that are below estimated damage thresholds and yield short- and long-term non-lethal effects.

Session 4aPP

Psychological and Physiological Acoustics and Speech Communication: Interactions Between Voice and Speech Perception

Etienne Gaudrain, Cochair
Lyon Neuroscience Research Center, CNRS, Centre Hospitalier Le Vinatier - Bâtiment 462 - Neurocampus, 95 bvd Pinel, Lyon, 69000, France

Emma Holmes, Cochair
University College London (UCL), Department of Speech Hearing and Phonetic Sciences, Chandler House, 2 Wakefield Street, London, WC1N 1PF, United Kingdom

Jens Kreitewolf, Cochair
McGill University, 2001 McGill College, Montreal, H3A 1G1, Canada

Invited Papers

8:00

4aPP1. Voice perception as a bridge between psychoacoustics and speech intelligibility. Etienne Gaudrain (Lyon Neurosci. Res. Ctr., CNRS, Ctr. Hospitalier Le Vinatier - Bâtiment 462 - Neurocampus, 95 bvd Pinel, Lyon 69000, France, etienne.gaudrain@cnrs.fr)

While clinical assessments of speech intelligibility focus on the reception of phonetic to lexical cues by a listener, in natural situations, speech also contains cues that inform the listener about the identity of the talker, or their emotional state. These *indexical* cues contribute to the access of phonetic information, either through talker normalisation, or by promoting the segregation of competing talkers. They can also directly affect the interpretation of the lexical content, e.g., through prosody. In other words, indexical cues play a crucial role in everyday communication. Yet, they are rarely considered in clinical evaluations of hearing impairment. Through various studies involving child and adult hearing aid and cochlear implant users, it appears that some voice cues are more affected by hearing loss than others, and this could have consequences on the development and remediation strategies. These studies also highlight the central role that voice perception study could play in connecting psychoacoustics to communication sciences. As voice literally carries the linguistic information in speech, it sits at the forefront of auditory processing. The mechanisms underpinning hearing loss thus tend to directly affect the neural representations of voice properties. As voice manipulation technologies are becoming more precise and more widely available, new opportunities arise to link peripheral auditory deficits to speech processing difficulties in populations with impaired hearing.

8:20

4aPP2. A mechanistic investigation of processing costs associated with cross-talker acoustic-phonetic variability. Sahil Luthra (Psych., Carnegie Mellon Univ., 5000 Forbes Ave., Pittsburgh, PA 15213, sahil.bamba.luthra@gmail.com)

The acoustic realization of speech sounds may differ substantially across talkers, such that (for example) a sound interpreted as "s" in one voice might be considered a "sh" in another. In this talk, I will review select neurobiological and computational data illustrating how listeners cope with this variability. I will also review behavioral data showing how talker variability can hinder processing, with slower and/or less accurate processing when the listening environment contains multiple voices compared to one voice. I argue that at least three mechanisms underlie these multi-talker processing costs. First, multi-talker processing costs may arise because talker changes disrupt a listener's ability to attend to the target talker. Second, I make the novel claim that talker changes might impose processing penalties by inducing uncertainty about whether the extant acoustic-to-phonetic mapping is appropriate, motivated by recent data that when multi-talker processing costs are attenuated by a preceding carrier phrase, this manifests as slowed responses to single-talker speech rather than improved processing of mixed-talker speech. Finally, talker variability costs may result from making on-the-fly adjustments to the mapping between acoustics and phonetic categories. Overall, this talk highlights the ways in which listeners contend with and are hindered by talker variability.

8:40

4aPP3. What do voices contribute to acoustic context effects in speech perception? Christian E. Stilp (Psychol. and Brain Sci., Univ. of Louisville, 317 Life Sci. Bldg., Louisville, KY 40292, christian.stilp@louisville.edu)

All perception takes place in context, and speech perception is no exception. Surrounding sounds form a context that guides speech perception. This is particularly true when earlier sounds inform the perception and/or recognition of later sounds. Several lines of research have advanced models where listeners use preceding context to account for characteristics of the talker's voice, such as vocal tract properties, articulatory maneuvers, or the speaker's identity. Conversely, research with nonspeech sounds and/or nonhuman animals

has advanced alternative models where perception is responding more to signal acoustics than to vocal characteristics. In this talk, I will review recent evidence from our research on acoustic context effects and talker normalization when hearing speech and nonspeech (music) sounds. Parallel patterns of performance across speech and music domains are conducive to, but not definitive evidence of, acoustic context being utilized according to general signal acoustics, with speech- and voice-specific contributions coming later in the processing stream.

9:00

4aPP4. How does voice familiarity affect speech intelligibility? Emma Holmes (Univ. College London (UCL), Dept. of Speech Hearing and Phonetic Sci., Chandler House, 2 Wakefield St., London WC1N 1PF, United Kingdom, emma.holmes@ucl.ac.uk)

People often face the challenge of understanding speech when other sounds are present (speech-in-noise perception)—which involves a variety of cognitive processes, such as attention and prior knowledge. We have consistently found that familiarity with a person's voice improves the ability to understand speech-in-noise, using both naturally familiar (e.g., friends and partners) and lab-trained voices. In this talk, I will describe experiments in which we manipulated voice acoustics (such as fundamental frequency and formant spacing). For example, we have measured the smallest deviations in acoustics that listeners can discriminate for familiar and unfamiliar voices, and have examined how manipulations to voice acoustics affect voice recognition and speech intelligibility for familiar voices. This work has provided insights into the processes underlying the familiar-voice intelligibility benefit—for example, by contrasting explanations based on predictions of voice acoustics with those involving cognitive demands. I will discuss the implications of our findings for theories of speech perception, and the implications for populations who typically find speech perception particularly challenging (e.g., older adults).

9:20

4aPP5. Speech comprehension in the context of speaker changes: The importance of voice-feature continuity at the cocktail party. Jens Kreitewolf (McGill Univ., 2001 McGill College, Montreal, QC H3A 1G1, Canada, jens.kreitewolf@mcgill.ca)

In this talk, I will show how different groups of listeners use voice acoustics to enhance speech comprehension under adverse listening conditions, specifically when the auditory scene comprises a multitude of sounds heard at once. These "cocktail-party"-like situations pose a difficult conceptual problem: To comprehend target speech, listeners need to attend to the target voice while at the same time ignoring other irrelevant sounds. The cocktail-party problem is made considerably easier when all target sounds are spoken by the same talker. Previous work suggests that such benefits from voice continuity can be—in large part—attributed to two prominent voice features: Glottal-pulse rate (GPR) and vocal-tract length (VTL). GPR determines the fundamental frequency of a speech sound and is perceived as vocal pitch; VTL determines the spectral envelope of a speech sound and is perceived as an aspect of vocal timbre. Apart from being important voice identity cues, GPR and VTL have been shown to play a crucial role in cocktail-party listening. Here, I will present data from a series of experiments highlighting the importance of voice-feature continuity for speech comprehension at the cocktail party.

9:40–9:55 Break

9:55

4aPP6. Perception of dynamic pitch and prominence in speech. Hae-Sung Jeon (School of Psych. and Humanities, Univ. of Central Lancashire, University of Central Lancashire, Preston PR1 2HE, United Kingdom, hjeon1@uclan.ac.uk)

The perception of dynamic – constantly changing – pitch in speech has been extensively studied in psychoacoustics and linguistics. In psychoacoustic studies, listeners are usually presented with short stimuli such as vowels or syllables, and their ability to discriminate a pair of stimuli is assessed. On the other hand, linguistic studies concern intonation over an utterance. Intonation entails not only acoustic prominence realised by pitch, duration, and loudness, but also listeners' knowledge about the relative prominence between syllables or between words in their language. This paper discusses a series of speech perception experiments using both psychoacoustic and linguistic tasks. Participants judged either relative pitch height or prominence between two pitch peaks or valleys in an utterance. Native English speakers in different age and dialectal groups were tested. The results showed that, first, listeners' pitch height discrimination in the utterance context seems to be more accurate than previously reported. Second, there is a robust perceptual asymmetry between pitch peaks and valleys, the valleys posing significant challenges in perception. Third, listeners' perception of pitch height and prominence is disassociated. The findings taken together suggest an intricate interaction between the physical properties of the stimuli and listeners' top-down knowledge in the perception of speech intonation.

10:15

4aPP7. Sensitivity to speech-relevant features in hallucination-prone individuals. Julia Erb (Inst. for Systems and Robotics - Lisboa and Dept. of Bioengineering, Instituto Superior Técnico, Universidade de Lisboa, Portugal, Avenida Rovisco Pais, 1, Lisbon 1049-001, Portugal, erbjulia@gmail.com), Patrícia Figueiredo (Inst. for Systems and Robotics - Lisboa and Dept. of Bioengineering, Instituto Superior Técnico, Universidade de Lisboa, Portugal, Lisbon, Portugal), and Ana P. Pinheiro (Faculdade de Psicologia, Universidade de Lisboa, Lisbon, Portugal)

As hallucinations occur in the absence of an external stimulus, they constitute an intriguing model for how percepts are generated and for how perception can fail. This study explores whether hallucination proneness is linked to an altered perception of speech-related acoustic features. Involving 320 healthy adults with varying predispositions for hallucinations, participants evaluated ambiguous sound textures for their speech-likeness. Psychophysical reverse correlation revealed that higher hallucination proneness was associated with reduced weighting of speech-typical low-frequency acoustic energy. Temporal modulation discrimination capabilities were unrelated to hallucination proneness in a subset of 41 participants. Confidence judgments in individual trials were influenced by both acoustic evidence and individual hallucination proneness and schizotypy scores. Overall, these findings suggest that hallucination-prone individuals exhibit qualitative and quantitative changes in their perception of speech-relevant modulations, supporting the notion of altered perceptual priors and differential weighting of sensory evidence.

10:35

4aPP8. Enhancing sarcasm detection through multimodal data integration: A proposal for augmenting audio with text and emoticon. Xiyuan Gao, Shekhar Nayak (Campus Fryslân (Lang., Technol. and Culture), Univ. of Groningen, Leeuwarden, Netherlands), and Matt Coler (Campus Fryslân (Lang., Technol. and Culture), Univ. of Groningen, Wirdumerdijk 34, Leeuwarden 8911CE, Netherlands, m.coler@rug.nl)

Sarcasm detection presents unique challenges in speech technology, particularly for individuals with disorders that affect pitch perception or those lacking contextual auditory cues. While previous research [1, 2] has established the significance of pitch variation in sarcasm detection, these studies have primarily focused on singular modalities, often overlooking the potential synergies of integrating multimodal data. We propose an approach that synergizes auditory, textual, and emoticon data to enhance sarcasm detection. This involves augmenting sarcastic audio data with corresponding text using Automatic Speech Recognition (ASR), supplemented with emoticons based on emotion recognition and sentiment analysis. Emotional cues from multi-modal data are mapped to emoticons. Our methodology leverages the strengths of each modality: emotion recognition algorithms analyze the audio data for affective cues, while sentiment analysis processes the text generated from ASR. The integration of these modalities aims to compensate for limitations in pitch perception by providing complementary cues essential for accurate sarcasm interpretation. Our approach is expected to significantly improve sarcasm detection, especially for those with auditory processing challenges. This research highlights the potential of multimodal data fusion in enhancing the subtleties of speech perception and understanding, thus contributing to the advancement of speech technology applications.

10:55

4aPP9. Exploring auditory emotion perception in cochlear implant users: Investigating the interplay of speech processing and affective signals. Sebastien Paquette (Psych., Trent Univ., A-806 Foxe St., Peterborough, ON K9H6Y7, Canada, sebastienpaquette@trentu.ca)

Human emotions are intricately expressed through vocal sounds, encompassing affective prosody in speech and non-verbal cues such as screams and laughter. Recent evidence indicates that vocalizations take neurophysiological precedence over speech-embedded emotions and are generally easier to identify. However, Cochlear implant (CI) users still face challenges deciphering the subtle nuances in these primal signals. The implant's limited fidelity in transmitting acoustic information results in highly variable levels of emotion perception abilities among its users. Identifying the factors explaining this significant variability in abilities among CI users remains of great interest. Our recent investigations into CI users' abilities to perceive emotions and speaker sincerity have often incorporated diverse aspects of auditory proficiency, including pitch discrimination, music processing, and speech intelligibility. The combination of results from these different projects can help shed light on the intricate interplay between speech processing and emotional recognition in CI users. Surprisingly, even when presented with emotional musical stimuli, CI users' proficiency often leaned toward processes related to speech intelligibility, proposing common mechanisms underlying linguistic and affective processes in CI users that do not readily relate to musical skills or pitch sensitivity. Hence, maintaining a clinical focus on speech processing remains crucial, even when exploring affective skills in CI users.

Contributed Papers

11:15

4aPP10. Spectral resolution and voice cue weighting in adults with normal hearing. Victoria A. Sevich (Dept. of Speech and Hearing Sci., The Ohio State Univ., 1070 Carmack Rd., Columbus, OH 43210, sevich.1@osu.edu) and Terrin N. Tamati (Dept. of Otolaryngology—Head & Neck Surgery, Vanderbilt Univ. Medical Ctr., Nashville, TN)

Cochlear implant (CI) users use acoustic voice cues differently than normal hearing (NH) adults to identify a talker's gender. Specifically, whereas NH listeners weight a talker's fundamental frequency (f0) and resonance (operationalized as vocal tract length or VTL) equally, CI users rely almost exclusively on f0. CI users' abnormal cue weighting may partially arise due to degraded auditory information delivered by the CI. We hypothesized that altering the amount of spectral information in the signal impacts voice cue weighting in NH listeners. Thirty NH adults performed a gender identification task. Auditory stimuli were monosyllabic words synthesized to have one of five f0 values and one of five VTL values. Synthesized voices were processed using a 16-, 8-, and 4-channel noise-band vocoder. Perceptual weights for each voice cue were estimated as the coefficients for f0 and VTL in regression models, with higher coefficients corresponding to stronger perceptual weightings. Listeners relied more on f0 in conditions with less spectral resolution than in conditions with greater spectral resolution. Cue weights for VTL did not change across conditions. Results suggest that removing frequency information from the auditory signal can modify the extent to which listeners use f0 to identify the gender of a talker.

11:30

4aPP11. Acoustic divergence from the training sample determines talker identification accuracy for emotional voices. Lue Shen (Dept. of Speech, Lang., and Hearing Sci., Boston Univ., Boston University, Dept. of Speech Lang. and Hearing Sci., Boston, MA 02215, shenlue@bu.edu), Yuxuan Wang (Dept. of Biostatistics, Boston Univ., Boston, MA), Patrick Wong (Dept. of Linguist., The Chinese Univ. of Hong Kong, Hong Kong, Hong Kong), and Tyler K. Perrachione (Dept. of Speech, Lang., and Hearing Sci., Boston Univ., Boston, MA)

Different emotional states introduce substantial acoustic variations in talkers' voices. It remains unclear how within-talker variability across emotional states affects listeners' ability to maintain perceptual constancy during talker identification. Here, we investigated (1) how changes in talkers' emotional state affected talker identification accuracy, (2) how emotional state affected key features of voice acoustics, and (3) how emotion-related changes in these acoustic features affected listeners' talker identification performance. Forty-eight listeners learned to identify talkers from speech expressing one emotional state (neutral, fearful, or angry) and then attempted to generalize that knowledge to speech expressing another emotional state. Talker identification accuracy was significantly worse in untrained emotions. Changes in voice acoustics across emotions were characterized for mean F0, F0 variability, jitter, HNR, speaking rate, and mean F2. To determine how emotion-related acoustic changes affected talker identification, we modeled talker identification accuracy as a function of difference in these features between training and test stimuli. Accuracy decreased as acoustic differences increased, regardless of talkers' emotion. Thus, perceptual constancy depends on acoustic similarity to prior experience with a talker's voice. Larger acoustic deviations, like those introduced by changes in emotional state, are more likely to cause a listener to misidentify a talker.

Session 4aSA

Structural Acoustics and Vibration, Biomedical Acoustics, Musical Acoustics, Physical Acoustics, and Engineering Acoustics: Additive Manufacturing and Acoustics

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

Kathryn Matlack, Cochair
University of Illinois at Urbana-Champaign, 1206 W Green St, Urbana, IL 61801

Matthew Luu, Cochair
Penn State, 446 Bluecourse Dr (Apt907), State College, PA 16803

Thomas Bowling, Cochair
Naval Surface Warfare Center Carderock Division, Bethesda, MD 20817

Chair's Introduction—9:00

Invited Papers

9:05

4aSA1. Ultrasonic characterization of material anisotropy in additively manufactured AlSi10Mg. Joseph A. Turner (Mech. and Mater. Eng., Univ. of Nebraska-Lincoln, University of Nebraska-Lincoln, Lincoln, NE 68588, jaturner@unl.edu), Nathanial Matz, W. T. Brandl (Mech. and Mater. Eng., Univ. of Nebraska-Lincoln, Lincoln, NE), Pulkit Kumar, Ronald A. Roberts, and Peter C. Collins (Ctr. for Nondestruct. Eval., Iowa State Univ., Ames, IA)

Metal additive manufacturing (AM) based on laser powder bed fusion (LPBF) generates components by melting metal powder on a layer-by-layer basis. The melting and cooling process often generates samples with a preferred material symmetry aligned with the build direction. This anisotropy affects mechanical performance and can be challenging to characterize nondestructively. Here, LPBF was used to create samples of AlSi10Mg with specific geometries to affect the overall anisotropy. In addition, the hybrid AM process of interlayer milling was used to impact the microstructure and residual stress of some samples. Ultrasonic measurements were used to characterize the samples using both coherent wave and diffuse wave experiments to capture the anisotropic nature of the wave speed and scattering. The material symmetry and morphology of the grains affect the wave speed, attenuation, and backscatter with respect to direction. Furthermore, spatially resolved acoustic spectroscopy was used to provide insight regarding the localized wave speeds with respect to sample location and propagation direction. The experimental data were used collectively to quantify differences between the AM processes used to create the samples. Such information can be used to guide AM process parameters to optimize sample performance. Finally, prospects for characterization of residual stresses will be discussed.

9:25

4aSA2. Direct sound printing: Way of manipulating ultrasonic chemistry to print directly engineering structures and remotely inside body. Shervin Foroughi (Mech. Eng., Concordia Univ., Montreal, QC, Canada), Mohsen Habibi (Mech. Eng., Univ. of California Davis, Davis, CA), and Muthukumaran Packirisamy (Mech. Eng., Concordia Univ., EV4-145, 1455 de Maisonneuve Blvd W, Montreal, QC H3G 1M8, Canada, m.packirisamy@concordia.ca)

Direct sound printing (DSP) is a new class of additive manufacturing processes developed in our lab, in which chemical reactions during the 3D printing process are driven by sonochemical route using cavitation bubbles induced by focused ultrasound waves. This invited paper will present methods and possibilities of printing engineering structures with DSP. In addition, this talk will cover a new area called remote distance printing (RDP) and consequent applications. RDP is a new realm introduced by DSP method in which the printing location is not accessible by common energy sources like light or heat. In this situation, ultrasound could penetrate optically opaque materials and conduct printing without direct access to the printing location. This concept opens a wide variety of applications in engineering or medical fields. The focus of this paper is the application of DSP-RDP in biomedical application to print objects inside body without open surgery in a non-invasive manner. Ultrasound penetrates skin and tissues in DSP and is focused on the printing location inside body where the printing material is injected. This work explains DSP in detail and the interaction of the sound with the printing material and how the material is transformed from liquid to solid. The process is demonstrated using a test study conducted using tissue phantoms and also real porcine tissue. This work opens new applications to 3D print with ultrasound where no other 3D printing approaches can achieve.

9:45

4aSA3. Monitoring the build history of a wire-arc additively manufactured part using structural resonances. Karl A. Fisher (Mater. Eng., Lawrence Livermore Natl. Lab., 7000 E. Ave. Liveremore, CA 94551, fisher34@llnl.gov), John Elmer (Mater. Eng., Lawrence Livermore Natl. Lab., Livermore, CA), and James Candy (Computational Eng. Div., Lawrence Livermore National Security, Livermore, CA)

Wire arc additively manufactured parts can take a significant amount of time to fabricate from several hours to days. During this process, the part is exposed to a steady localized heating at the build zone, and rapid cooling away from the arc contact point. The heated zone is constantly moving across the part geometry during the build resulting in spatially localized heating and cooling throughout the part geometry. In this investigation, we utilize the broadband emission signal from the wire arc to measure the acoustic response of the part build. Experimental results are obtained using contact emission transducers mounted such that they are isolated from the direct heat to avoid damage. Transient acoustic signals are recorded throughout the entire build process and evaluated using Fourier analysis. Acoustic spectra are utilized to capture modal data which slowly track the parts construction. In a second effort the modal response of the part is modeled as a function of each build layer and the two modal structures are compared. The emphasis is to understand and track the modal response of the structure of the part as it is fabricated and develop a real time process monitoring capability.

10:05–10:20 Break

Contributed Papers

10:20

4aSA4. Preliminary investigation and screening study of relationships between acoustic emission signal features and powder blown laser-directed energy deposition parameters for *in situ* monitoring of metals additive manufacturing. Emmeline Evans (Mech. Eng., Georgia Inst. of Technol., 801 Ferst Dr., Atlanta, GA 30308, eevans70@gatech.edu), Erin Lanigan (NASA Marshall Space Flight Ctr., Huntsville, AL), and Aaron Stebner (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

Metal additive manufacturing (AM) processes have allowed for easier fabrication of parts made from alloys that are difficult to machine and typically yield less material waste than traditional manufacturing methods. Therefore, interest in metal AM for in-space components has grown in recent years. As this interest in metal AM as grown, so too has the need for real-time process monitoring for defect detection during printing. However, commonly used process monitoring methods, such as melt-pool imaging, are too data intensive for automated analysis to be performed and reported in real-time. Acoustic emission (AE) monitoring presents an alternative approach in which one-dimensional data can be captured continuously during printing and analyzed more quickly than image data. This work presents a methodology for and preliminary findings from a screening study that explores the effects of powder blown laser beam directed energy deposition (DED-LB) parameters on resulting AE that occur during printing. The primary factors considered in this study are print parameters that have been found in literature to predict part density, such as laser power and mass flow rate, and are studied here to determine their contributions to AE. This work is supported by the NASA Space Technology Graduate Research Opportunities program.

10:35

4aSA5. Internal damping measurements of additive manufactured metal beams. Peter K. Jensen (Dept. of Phys. and Astronomy, Brigham Young Univ., Brigham Young University, Provo, UT 84602, peterkj@byu.edu), Joshua T. Mills, and Micah Shepherd (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Modern additive manufacturing (AM) techniques have created an endless number of new design possibilities. Naturally, it is important to understand how the material properties, including internal damping, differ for AM structures. An experimental procedure has been developed to measure an upper bound for the internal damping of metal beams by minimizing the effects of energy dissipation at the supports and acoustic radiation into the surrounding air. In this talk, measurements for the loss factor of several common metals will be compared to equivalent samples constructed by powder bed fusion at varying angles. The results will also be compared to Zener's thermoelasticity model, developed for isotropic Euler beams in flexure. Reasons for deviation from theory which arise from the manufacturing technique will be explored.

10:50

4aSA6. Influence of ultrasonic parameters on microstructural refinement and defect elimination in ultrasound-assisted laser-based metal additive manufacturing. Lovejoy Mutswatiwa (Eng. Sci. and Mech., The Penn State Univ., 212 Earth And Eng. Sci, University Park, PA 16802, lpm5609@psu.edu), Judith A. Todd (Eng. Sci. and Mech., The Penn State Univ., State College, University Park, PA), Edward Reutzel (CIMP 3D, Penn State Appl. Res. Lab., University Park, PA), and Christopher M. Kube (Eng. Sci. and Mech., The Penn State Univ., University Park, PA)

Acoustic cavitation, streaming, and energy absorption during solidification in ultrasound-assisted direct energy deposition additive manufacturing (DED-AM) have been reported to drive microstructural refinement, defect elimination, and mechanical property improvement. However, the influence of individual ultrasonic parameters such as frequency, amplitude, intensity, and sonication duration remains unknown. This is mainly because of challenges in real-time quantification of the influence of ultrasound on the rapid solidification and microstructural development processes encountered in ultrasound-assisted AM. High temperature and opaque molten metal confined within micro-length scale melt pools further challenge the characterization of ultrasound's influence on microstructural development. Building upon our recent *in situ* observation of acoustic cavitation and streaming in sonicated laser-generated melt pools, this talk will highlight our efforts to correlate vibration frequency, amplitude, and intensity with grain size and texture of Al7075 alloy fabricated using ultrasound-assisted DED-AM. DED additively manufactured Al7075 is susceptible to solidification cracking. Therefore, this presentation will also showcase the influence of ultrasonic parameters on cracking suppression and defect elimination. In addition, the effect of sonication duration on microstructure will also be elaborated. Lastly, the presentation will showcase our future work toward upscaling ultrasound-assisted AM to large parts with complex geometries.

11:05

4aSA7. Sound absorption properties of additively manufactured porous materials with minimal surface pore geometries. Anthony Ciletti (Aerosp. Engineerign, Wichita State Univ., Wichita, KS), Martha C. Brown (NASA Langley Res. Ctr., Hampton, VA), and Bhisham Sharma (Mech. Eng. - Eng. Mech., Michigan Tech Univ., 1400 Townsend Dr., Houghton, MI 49931, bnsharma@mtu.edu)

This study explores the use of additive manufacturing, specifically stereolithography (SLA), to create porous acoustical materials with precise pore geometries for aircraft engine noise reduction. Unlike traditional methods like foaming and fiber-spinning, SLA allows for exact control over pore shapes. Three triply periodic minimal surface (TPMS) pore designs were investigated: split-P, lidinoid, and diamond. These structures were designed through implicit modeling and made from polymeric resin. Their sound absorption capabilities were tested under normal incidence in a two-microphone impedance tube to examine how porosity influences their

acoustic performance. Findings revealed unique sound absorption profiles for each geometry, which can be modified by adjusting porosity. Additionally, the static flow resistivity was measured using a raylometer, where results showed that flow resistance is a key factor in their absorption efficiency. Among the three samples, the split-P geometry shows superior sound absorption. This preliminary research highlights the potential of using tailored TPMS geometries in acoustic liners for effective noise control in aerospace applications.

11:20

4aSA8. Abstract withdrawn.

11:35

4aSA9. Effects of printer variation on vibration response of fused deposition modeling-fabricated elastic beams with symmetric and asymmetric resonators: A round robin study. Christina J. Naify (Appl. Res. Labs., The Univ. of Texas at Austin, 4555 Overlook Ave SW, Washington, DC 20375, christina.naify@gmail.com), Colby W. Cushing, Nathan P. Geib, Matthew Wash (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Jared Allison (Univ. of Texas at Austin, Austin, TX), Caleb F. Sieck (Code 7160, U.S. Naval Res. Lab., Washington, DC), Alec K. Ikei, Amelia Ware (Acoust., US Naval Res. Lab., Washington, DC), Thomas Bowling (Naval Surface Warfare Ctr. Carderock Div., Bethesda, MD), Benjamin S. Beck, Callie Zawaski (Eng. Acoust., Penn State Appl. Res. Lab, State College, PA), and Abigail T. Juhl (Air Force Res. Lab., Wright Patterson Air Force Base, OH)

Additive manufacturing has expanded rapidly as a production tool due to its ease of use for rapidly producing parts. The most common polymer-based additive approach is fused deposition modeling (FDM) which uses plastic filament to build parts additively, one line at a time, to build a 3D shape. FDM printers are ubiquitous in university, government and industry settings, making them ideal for mass-production of shared designs across institutions. While it is commonly known that FDM finished products have variations due to differences in printer model and slicing and extrusion settings, little has been done to quantify the effects of these variations on elasto-dynamic response. In this study, a multi-institutional round robin approach is used to quantify the printer-to-printer variations of a structure comprised of a thin beam with attached resonators. The parameters of the round robin study involve printing the same geometry, with the same base material, on whatever FDM-type printers are available at each of six contributing institutions. All samples were then sent to a common location and tested on the same apparatus to limit experimental variability. This presentation will discuss the design of the study, test and modeling efforts, and a summary of results.

THURSDAY MORNING, 16 MAY 2024 — ROOM 203, 8:00 A.M. TO 9:25 A.M.

Session 4aSCa

Speech Communication: Language and Sports

Shiloh Drake, Cochair
Department of Linguistics, University of Oregon, 1290 University of Oregon, Eugene, OR 97403

Melissa Baese-Berk, Cochair
Linguistics, University of Chicago, 1115 E 58th St, Rosenwald Hall Room 203, Chicago, IL 60657

Chair's Introduction—8:00

Invited Papers

8:05

4aSCa1. "He speaks great English—For a guy from Moscow": Language ideologies in NHL media. Shiloh Drake (Dept. of Linguist., Univ. of Oregon, 1290 University of Oregon, Eugene, OR 97403, sdrake@uoregon.edu) and Melissa Baese-Berk (Linguist., Univ. of Chicago, Chicago, IL)

Ideologies about languages and countries are hard to shake, even in a multinational, multilingual setting like the National Hockey League (NHL) and the journalists who report on it. Despite its historical roots in Montréal and the dominance of Canadian and European players, the lingua franca of the NHL is English. In this work, we used qualitative analyses to examine players', journalists', and

4a THU. AM

coaches' attitudes toward languages other than English used on the ice. Across all groups, we found that Russian speakers were most likely to be assessed negatively, from being taciturn and unwilling to be interviewed (Frederickson, 2023) to being unlikely to speak good English (Keefe, 2023). Additionally, English-speaking players were more likely to associate positive sentiments with native North American English, Swedish, and Finnish speakers, but negative sentiments about Canadian French speakers and players from eastern European countries. Players and coaches also tended to be split on whether it was acceptable for other languages to be spoken in the locker room and on the ice. This work points to a fragmented in-group view of the acceptable language to use in professional hockey.

8:25

4aSCa2. Acoustics and ice hockey: The sociophonetic impact of Canadian English on American-Born players. Andrew R. Bray (Linguist., Univ. of Rochester, 513 Lattimore Hall, Rochester, NY 14626, andrew.bray@rochester.edu)

My research utilizes sociophonetic analysis to document the linguistic identity construction process that is ongoing in the sport of ice hockey. I argue that American-born players are constructing a hockey-based identity influenced by Canadian English (CE) due to the historical Canadian dominance of the sport. This identity incorporates Canadian Raising, FACE and GOAT monophthongization, both commonly attributed to CE and largely unexplainable based on players' regional dialects, and altered vowel production in hockey-specific terminology, most notably in the word *hockey* itself, unique to the hockey community. To document this variation, I analyze vowel formant values taken from sociolinguistic interviews with professional hockey players. I assess F1 and F2 values throughout vowel durations to establish if players are converging in production away from regional dialectal variants towards shared hockey-based variants. I argue these variants have gained indexical value linked to an emerging hockey-based identity that, although influenced by CE, is unique to the hockey community. In ongoing research, I aim to further document that this variation is most evident in hockey-specific terminology and that lexical diffusion occurs outwardly from these terms over time leading to players developing a more prevalent hockey-based identity as the sport gains more importance in their lives.

8:45

4aSCa3. There's no "I" in hockey: Identity work in hockey post-game interviews. Sarah Adams (Linguist., Univ. of Colorado Boulder, 433 Buchanan, University of Colorado Boulder, Boulder, CO 80310, saad1393@colorado.edu)

The community of practice represented by a professional sports team yields fascinating yet understudied sociolinguistic data on identity construction. Within the specific institutional context of hockey, the post-game interview between athlete and media personnel is a site for meaningful interactional analysis, where these interactions have tangible implications. Data are taken from the post-game interviews of the Colorado Avalanche professional men's ice hockey team during the 2020 postseason in Edmonton. Analysis of these interactions provides insight on the athlete's ability to use the interview to develop himself as a part of his team, or his "teamness." I identify the ways in which the team's culture is constructed and reinforced through specific linguistic practices. I further introduce the interviewer as a co-constructor who at once holds an agenda and makes space for the athlete to be agentive. The data illustrate the role of praise and blame, positioning evasion as an appropriate answer when the athletes avoid self-praise or blaming a teammate, as well as referent shifts and evaluation shifts, and passive constructions. I use research on turn design to analyze these practices and link the institutional role of the athlete to his appropriate performance of "teamness" through his use of language. For Speech Communication Technical Committee Best Student Paper Award.

9:05

4aSCa4. Sports and linguistics pedagogy. Robert Kennedy (Dept. of Linguist., Univ. of California, Santa Barbara, CA 93106-3100, rkennedy@linguistics.ucsb.edu)

The domain of organized sports offers a unique circumstance to explore a wide range of fundamental concepts in linguistics classrooms. We present an overview of a pedagogical model that engages students of linguistics using data from sports oriented settings, with frequent reliance on peer-reviewed literature. The course model incorporates three domains of language behavior: (a) lexis, and the interplay of semantic specialization and sociocultural indexation of specialized sport-oriented vocabulary; (b) sports announcer talk, a spoken broadcast register whose study enables a novel exploration of syntactic nuance; (c) participant interaction with each other and with the media, which demonstrates fundamentals of pragmatic inference in spoken communicative exchanges. Some course topics span more than one of these domains. Nickname creation, for example, is phonologically and phonetically driven, with gradient effects that adhere to constraints of brevity and clarity, yet linked to sociocultural indexicality. While no single textbook serves as an appropriate primary text for such a course, the beneficial outcomes for students are numerous. Aside from the primary goal of applying linguistic concepts to an engaging domain, students probe a variety of published works that examine sports/language interactions, demonstrating a broad sample of research methods and theories.

Session 4aSCb

Speech Communication: Speech Perception Poster Session II

Benjamin Munson, Chair
University of Minnesota, 115 Shevlin Hall, Minneapolis, MN MN

All posters will be on display and all authors will be at their posters from 10:00 a.m. to 12:00 noon.

Contributed Papers

4aSCb1. Differences in prominence-related acoustic measurements in L2 read speech as a function of L1 framing typology. Carissa A. Diantoro (Linguist., Univ. of Oregon, 1290 University of Oregon, Eugene, OR 97403, carissad@uoregon.edu), Hakyung Sung (Linguist., Univ. of Oregon, Eugene, OR), and Melissa M. Baese-Berk (Linguist., Univ. of Chicago, Chicago, IL)

Speech production in a non-native language is often influenced by their first language's (L1) phonology. One factor that might affect their production prosodically is typological differences in event construal, specifically motion events. Building on this concept, prior studies on gestures and spontaneous speech show that how speakers frame motion events in their L1 can transfer to their second language (L2) through grammatical patterns. However, no study has looked at how L1-L2 differences in framing motion events may affect speakers' L2 production prosodically. To explore whether those differences affect how L2 speakers prosodically produce motion events, this study collected read speech from two spoken corpora of L2 English learners. We chose L2 learners from different L1 backgrounds that differ in the way they grammatically construct motion events. English production from Korean and Turkish L1 speakers (which are typologically similar) were compared to that of L1 German speakers (which is typologically similar to English), along with productions by L1 English speakers. Duration and various acoustic measurements of pitch and intensity were extracted and analyzed. Results indicated some similarities among learners of the same typology along with differences between the L1 and L2 English speakers. Implications on conceptual transfer will be discussed.

4aSCb2. Distributional learning of non-native tone contrasts by older adults after training and overnight consolidation. Yin-To Chui (Div. of Humanities, The Hong Kong Univ. of Sci. and Technol., The Hong Kong University of Sci. and Technol., Clear Water Bay, Hong Kong, Hong Kong, ytchuiac@connect.ust.hk), Susu Lai, and Zhen Qin (Div. of Humanities, The Hong Kong Univ. of Sci. and Technol., Hong Kong, Hong Kong)

Despite decline in psychoacoustic and statistical learning (SL) abilities, older adults demonstrate remarkably intact perceptual learning in both L2 (tone-word learning) and L1 settings (perceptual adaptation to accented/noise-vocoded speech) but show limited transfer of learning to untrained stimuli. This study tests whether perceptual learning is maintained in an implicit statistical learning task where older adults learn L2 tonal contrasts through exposure to probability distributions of tonal tokens, which may pose higher requirements on both psychoacoustic and SL abilities, and whether sleep-dependent consolidation helps the generalization of perceptual knowledge. L1-Cantonese older adults learned to discriminate a perceptually difficult level-falling tone contrast following a pre-test, training, post-training overnight interval. Training stimuli were synthesized by interpolating naturally produced Mandarin level and high-falling tones into six equidistant steps. Participants either heard a bimodal (two-peak resembling level-falling categories) or unimodal distribution (single-peak) consisting of 256 tokens. ABX discrimination task was administered for testing, with tokens by two genders and on two pseudo-syllables to test generalization. Pilot data of 14 participants showed a trend of group effect with the bimodal group outperforming the unimodal group after training and sleep-dependent consolidation, showing that perceptual learning is maintained in a paradigm that relies heavily on psychoacoustic and SL abilities.

4aSCb3. Portuguese rhotic variation in the Brazilian cities of Salvador and São Paulo. Francis Jagiella (Lingusistics, Indiana Univ., 1020 E. Kirkwood Ave., Ballantine Hall 504, Bloomington, IN 47405, fjagiell@iu.edu)

Brazilian Portuguese has two rhotic phonemes: the alveolar flap /ɾ/ and the historically long version which previous publications call velar, uvular, or glottal fricatives, or alveolar trills and approximants. This variation occurs both within and across dialects. Deletion is also common, most notably in word-final position. For the current project, thirty-five participants from Salvador and ten participants from São Paulo were recorded reading predetermined stimuli of isolated words and sentences, creating 6,383 instances of the rhotic phoneme. Productions were classified as exhibiting deletion or for having voicing, frication, flapping, and place characteristics. The results indicate a range of surface forms of the phoneme more variable than previously cited, with palatal fricatives common in Salvador and several flap + fricative variants common in São Paulo, along with other less frequent forms. In Salvador, glottal fricatives predominate across the board with much higher rates of deletion. In São Paulo, glottal fricatives predominate in onset positions, but alveolar trills and approximants and flap + fricative variants predominate in coda position. While deletion is most common word-finally, it occurs in all environments where the phoneme is found.

4aSCb4. The production of tone and intonation in Mandarin-English bilingual children—A pilot study. Jie Yang (Commun. Disord., Texas State Univ., 200 Bobcat Way, Round Rock, TX 78665, j_y90@txstate.edu)

In addition to consonants and vowels, Mandarin Chinese uses two levels of prosodic contrast, tone at word level and intonation at utterance level, for linguistic message generation. When tone and intonation interact, Mandarin speakers use global and/or localized fundamental frequency (f0) adjustments to maintain linguistic intelligibility at both word- and utterance-level. Word-level tonal contrast does not exist in English. For English speakers, f0 changes at utterance level for intonation were not confined by word-level tonal requirements. The present study investigated how Mandarin-English bilingual Children manage these two levels of prosodic contrast within the respective phonology at different developmental stages. Six- and nine-year-old Mandarin-English bilingual children and young adults (seven per group) completed the speech production tasks. Stimuli were carrying sentences (statements and questions) ending with monosyllabic target words that are phonetically similar in Mandarin and English (e.g., [li] 丽, Lee). The carrying sentences were statements with falling intonation, questions with and without inversion in English, and questions with and without question particle (吗) in Mandarin. Average f0 and magnitude of f0 change were measured within words and over utterances and compared between languages and among age groups. Results indicated the influences of language and

developmental age. Work supported by Texas State University Research Enhancement Program.

4aSCb5. Stroop task response times: The bilingual advantage. Kathleen Siren (Speech-Language-Hearing Sci., Loyola Univ. Maryland, 4501 N. Charles St., Baltimore, MD 21210, ksiren@loyola.edu), Bridget Killmurray (Speech-Language-Hearing Sci., Loyola Univ. Maryland, Baltimore, MD), Sarah Bayer (UNC-Chapel Hill, Chapel Hill, NC), Sophia Launay-Fallasse, Meghan Stapp, and Tepanta Fossett (Speech-Language-Hearing Sci., Loyola Univ. Maryland, Baltimore, MD)

Bilingual individuals often must focus on one language while suppressing interference of another language and may therefore develop a specialized capacity to selectively attend to the language currently being used, a form of cognitive control. For this investigation, monolingual and bilingual individuals completed a 40-item modified Stroop task. In this task, participants were shown incongruent visual stimuli, with the names of various colors presented in a different ink color. Participants were instructed to respond verbally by naming the ink color instead of reading the written color word. Investigators scored verbal responses for accuracy and measured speed of response. Additionally, investigators measured duration of selected responses. Results demonstrate that monolingual and bilingual individuals were similarly accurate on the task. However, on average, bilingual speakers were faster at responding than monolingual speakers. Further, bilingual speakers were less variable in response time with responses that were shorter in duration than monolingual speakers. These results are discussed with reference to theories of differential brain development and cognitive function in individuals who simultaneously acquire more than one language.

4aSCb6. Age and category structure in phonetic category learning. Christopher C. Heffner (Communicative Disord. and Sci., Univ. at Buffalo, 122 Cary Hall, South Campus, Buffalo, NY 14214, ccheffne@buffalo.edu)

When learning new speech sound categories in a second language, listeners must decide which sounds belong to the same category and which belong to different categories. This learning process is contingent on perceptual systems that change across the lifespan, which may, in turn, make certain categories easier to learn depending on which systems are employed by learners. In the present study, learners across a variety of ages categorize speech sounds taken from German that vary in their category structure (i.e., the complexity of the categories being learned within the stimulus space). Participants aged 7 to 70 are recruited and run at community sites such as museums and libraries. The preliminary dataset indicates that the simpler category structure saw strong age effects (with poorer performance on the younger and older ends of the age range and better performance at intermediate ages), while differences across ages were much smaller for the more complex category structure. These findings suggest that learning simpler category structures may rely on age-dependent learning systems, while learning more complex categories may rely on systems that are less age-dependent.

4aSCb7. Relating assimilation data to identification data in second language learners. Kenneth J. de Jong (Dept. of Linguist., Indiana Univ., Dept. of Linguist., Ballantine Hall, Bloomington, IN 47405, kdejong@indiana.edu), Yu-Jung Lin (Foreign Lang., Literatures, and Cultures, College of the Holy Cross, Worcester, MA), Yen-Chen Hao (World Lang. and Cultures, Univ. of Tennessee, Knoxville, TN), and Hanyong Park (Linguist., Univ. of Wisconsin-Milwaukee, Milwaukee, WI)

Commonly cited models of bilingual phonology, e.g., the Speech Learning Model and the Perceptual Assimilation Model, project hypotheses about second language (L2) developmental patterns based on hypothetical assimilation patterns between categories in the learner's first language (L1) and L2, with the addition of a learning component which differentiates an L2 and L1 system. This paper relates two experiments with two participant groups, one native Korean, and one native Taiwan Mandarin. Both groups identify English obstruents /p b t d f v θ ð/ in four prosodic positions (onset, coda, and intervocalic pre-stress and post-stress) using their native non-Roman graphemes in the first experiment, and using English labels in the second. The assimilation results generate predictions for discrimination performance in the second experiment, assuming only the L1 categories. The two groups show large differences in assimilation pattern due largely to the non-sibilant fricative /f/ in Mandarin, in opposition to the Korean system which has only non-sibilant /h/. Analyses reveal a strong correlation between L2 discrimination patterns and predictions based solely on the L2-L1 assimilation results, suggesting that the perceptual system of L2 learners can be surprisingly well-captured as a single perceptual system with a bilingual array of segmental categories.

4aSCb8. The perception of (trans)masculine speech: Effects of stimulus acoustics and rater identity. Benjamin Munson (Speech-Language-Hearing Sci., Univ. of Minnesota, 115 Shevlin Hall, Minneapolis, MN MN, munso005@umn.edu) and Devin V. Dolquist (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN)

A large body of work has examined the acoustic characteristics of cisgender, heterosexual (cishet) men and women's speech, and the acoustic features that predict listeners' judgments of binary gender through speech. Considerably less work has examined the diverse ways that non-cishet men, non-cishet women, and people whose gender is neither exclusively male nor exclusively female express gender through speech. Moreover, there is relatively little work on how gender-diverse listener groups perceive gender through speech. This poster presents the results of a perception experiment designed to fill those gaps. Stimuli were multiple sentence productions of 20 masculine-presenting individuals, including cisgender men, transgender men, and transmasculine nonbinary people. These are described and analyzed acoustically in Dolquist (2023). Listeners were 88 cishet men, 62 cishet women, and 50 gender and sexuality diverse (GSD) individuals. Listeners identified gender identity, gender orientation (i.e., cisgender or transgender), and a variety of attributes for each stimulus. GSD listeners were more likely than cishet men and cishet women to identify voices as transgender or nonbinary, and were more likely to evaluate those voices favorably. Moreover, cishet men and women's judgments of gender were more strongly predicted by sex-dimorphic acoustic characteristics like f0 than were GSD listeners' judgments.

4aSCb9. Racialization and sentence intelligibility in older adults with hearing impairment. Benjamin Munson (Speech-Language-Hearing Sci., Univ. of Minnesota, 115 Shevlin Hall, Minneapolis, MN MN, munso005@umn.edu), Tatiana Lyons, Matthew B. Winn (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN), Peggy Nelson (Ctr for Applied/Translational Sensory Sci., Univ. of Minnesota, Minneapolis, MN), Enengy Schutt, and Alayo Tripp (Speech Lang. Hearing Sci., Univ. of Minnesota, Twin Cisties, Minneapolis, MN)

Previous studies have shown that the perception of a talker's racial identity affects the intelligibility of that person's speech to younger, normal-hearing listeners (Babel & Russell, 2015; McGowan, 2015; Tripp, Lyons, & Munson, 2022). These findings show that social perception influences intelligibility. These effects are thought to reflect varied experiences interacting with racially diverse individuals, attitudes and beliefs about different racial groups, or a combination of these factors. The current experiment examines whether the strength of the influence of perceived racial identity on intelligibility is similar across different age groups and levels of hearing acuity. Participants (n = 42, including 16 normal hearing [NH] younger adults, 11 NH older adults, and 15 older adults with hearing loss) were played sentences produced by four talkers who varied in racial identity, in audio-only and audiovisual conditions, in background noise and reported what they heard. Participants also completed a measure of attitudes toward the four talkers. Analyses are ongoing, and examine (1) whether the influence of presentation modality on intelligibility varies across the four talkers, (2) whether these talker differences in audiovisual benefit vary across the three listener groups, and (3) whether attitudes toward the talkers predict intelligibility. [Funding: NIH grant R21 DC018070]

4aSCb10. Preceding word information for predicting speech errors in English as foreign language speech. Ki Woong Moon (Linguist., Univ. of Arizona, 1200 E University Blvd, Douglass 318B, Tucson, AZ 85721, kiwoongmoon@arizona.edu)

Speech errors, including disfluency errors (e.g., filled pauses ("uh", "um"), repetition ("I me-mean right now.")), and mispronunciation of speech segments (e.g., "think" as /sɪŋk/) are natural occurrences in speech production and they can affect speech fluency and proficiency. Detecting these errors is important, especially in assessing second language (L2) learners. Non-native speakers often produce speech errors, even in read speech, due to increased cognitive load when simultaneously producing the current word and processing the upcoming word. By analyzing two L2 speech corpora having different types of errors (disfluency and mispronunciation), the study aims to assess the effectiveness of incorporating preceding word information in speech error prediction. The results indicate that longer syllable duration in the word preceding the mispronunciation error correlates with error production. The study extends its investigation to identify acoustic and lexical features influenced by the presence of speech errors. The study found that speech errors increase duration of speech. The study also found that words with more syllables may cause L2 speakers to produce speech errors.

4aSCb11. Quantitative assessment of English accent variation: Levenshtein distance and perceptual identification. Lian J. Arzbecker (Communicative Disord. and Sci., Univ. at Buffalo, 103D Cary Hall, Buffalo, NY 14207, arzbecker.1@osu.edu) and Ewa Jacewicz (Speech and Hearing Sci., The Ohio State Univ., Columbus, OH)

This is a perceptual study with a basis in acoustic phonetics. It investigates the extent to which accents of English can be quantified by the Levenshtein distance (LD), a string metric. LD defines string similarity by the minimum number of character edits required to transform one string into another. Narrowly transcribed speech samples of English provide the phonetic strings required for LD comparison. For the purposes of the current experiment, 24 speakers representing four English accent varieties are included: Midland American (control), British/Australian, Hindi-influenced, and Mandarin-influenced. High- and low-pass filters are used to augment or attenuate perceptual contribution of high- versus low-frequency information in the speech signal. Predictions stem from published correlations between LD and listeners' perceptional ratings of intelligibility and native-likeness. For a group of monolingual American English listeners, confusion is expected between Midland American and British/Australian accents due to similarly low LDs, further intensified by low-pass filtering. However, fewer confusions are predicted between Hindi- and Mandarin-influenced English, due to varying L1 influences and the absence/presence of tonal information. This research seeks to explore accent perception, investigating the relationship between manipulated aspects of the speech signal and listener identification.

4aSCb12. Beyond visual primes: Audiovisual integration of racial identity and sentence intelligibility. Eleanor Nickel (Sociology, Univ. of Minnesota, 909 Social Sci. Bldg., 267 19th Ave. S, Minneapolis, MN 55455, nicke288@umn.edu), Alayo Tripp (Linguist., Univ. of Florida, Minneapolis, MN), Enengy Schutt, and Benjamin Munson (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN)

There is no biological basis for racial categories. Yet, the presentation of racialized stimuli impacts psycholinguistic processing, suggesting that a talker's racial identity affects measures of intelligibility (Babel & Russell, 2015; McGowan, 2015). Previous work has used static visual primes, presenting images accompanied by acoustic sentence stimuli. Static visual primes may prime expectations of the person's speech, with unexpected pairings decreasing intelligibility. These findings show that social expectations affect speech intelligibility and have broad implications for real-world speech perception in high-stakes educational (Evans, Munson, & Edwards, 2017), social (Purnell, Baugh, & Idsardi, 1999), and legal (Rickford & King, 2016) contexts. Expanding our understanding of the psycholinguistic and psychophysical mechanisms behind these phenomena requires experimentation with dynamic, audiovisual speech rather than static visuals. The current experiment presents conflicting visual and auditory cues indexing the white and Black racial identities of the talkers. The talker who is seen (the 'visual' talker) and the racial identity of the talker who is heard (the 'audio' talker) are fully crossed. This design allows us to examine how intelligibility is influenced by visual racial identity, audio racial identity, and the audiovisual (mis)match between them. Data collection is currently in progress and includes groups of both white and Black listeners.

4aSCb13. Revisiting rapid accent adaptation with computational modeling. Samantha Chiu (Psychol. Brain Sci., Univ. of Iowa, 340 Iowa Ave. Iowa City, IA 52242, samantha-chiu@uiowa.edu), Leo Moore, and Ethan Kutlu (Dept. of Linguist., Univ. of Iowa, Iowa City, IA)

Studies on accent adaptation report that monolinguals can rapidly adapt to a novel foreign accent with short exposure or training (Baese-Berk *et al.*, 2013; Clarke & Garrett, 2004; Maye *et al.*, 2008). While most of these studies rely on self-reported monolinguals, there is great variability in how much language diversity monolinguals encounter regularly (Castro *et al.*, 2022). We return to this question of rapid accent adaptation in monolinguals through computational modeling where linguistic diversity is directly controlled. We simulate monolingual speech perception using PyTorch's Wav2-Vec2 model (Paszke *et al.*, 2019), pre-trained on the Librispeech corpus with American-accented English (Panayotov *et al.*, 2015). We replicate the experimental design of Baese-Berk *et al.* (2013) by fine-tuning this model on a single accent ($n = 150$ sentences across 5 talkers) and multiple accents ($n = 30$ sentences for each of 5 talkers). Preliminary results find that exposure to a single accent (68.6% correct) or multiple accents (69% correct) does not induce accent adaptation to a novel accent (no accent training = 69% correct). We predict that exposure to a single or multiple accents will increase accuracy but requires many additional hours of exposure. We discuss implications of our models against accent adaptation studies.

4aSCb14. Using computational models to study accent adaptation: A tutorial. Leo Moore (Linguist., Univ. of Iowa, 340 Iowa Ave. Iowa City, IA, leo-moore@uiowa.edu), Samantha Chiu (Psychol. Brain Sci., Univ. of Iowa, Iowa City, IA), and Ethan Kutlu (Linguist., Univ. of Iowa, Iowa City, IA)

One of the limitations in accent perception research is the difficulty of quantifying how much exposure is needed to observe adaptation to a novel accent. While numerous studies have investigated this issue (Baese-Berk *et al.*, 2013; Clarke & Garrett, 2004), it has been impossible to control for individual-level differences and solely focus on linguistic differences with human participants. However, computational models can be intentionally set in terms of these features to better understand the process of adaptation to novel accents. Here we discuss how machine learning models can be implemented using PyTorch, a framework in Python made by Meta. PyTorch allows for the easy creation of language learning models that can be used in speech-recognition experiments (Paszke *et al.*, 2019). Here we use the Wav2Vec2 model created by Meta, pre-trained on 960 hours of the Librispeech ASR corpus (Panayotov *et al.*, 2015). We discuss importing the model and fine-tuning it using samples of different accents. Next, we demonstrate building a decoder using a greedy algorithm to transform the model's classification into a transcript. Our goal is to compare simulations against data from human participants and tease apart the direct effects of individual variability on accent perception.

4aSCb15. Intelligibility of British and American English in different listening conditions. Elizabeth D. Young (Commun. Sci. and Disord., Univ. of Utah, 390 S 1530 E, Rm. 1218, Salt Lake City, UT 84112, liz.d.young@utah.edu), Kayleena Faulkner, Isabella McHugh, and Sarah H. Ferguson (Commun. Sci. and Disord., Univ. of Utah, Salt Lake City, UT)

Communication specialists frequently hear complaints from patients that unfamiliar accents are more difficult to understand, particularly in background noise. Previous work (Schilaty *et al.*, 2021) has indeed shown that when presenting American listeners with both American and British accents, intelligibility in noise is poorer for the unfamiliar accent across age and hearing status. However, Schilaty *et al.* (2021) examined relatively easy sentence materials (Basic English Lexicon [BEL] sentences; Calandruccio & Smiljanic, 2012) in only one type of noise. The current study expands on previous work by comparing the intelligibility of American and British

4a THU. AM

accents for a more difficult sentence set (Harvard sentences; Rothauser *et al.*, 1969) in young adult American listeners across two noise types (speech-shaped noise and 12-talker babble). Stimuli consisted of 180 Harvard sentences produced by two male talkers: one with an American (Northwestern) dialect and one with a Southern British dialect. Young adult listeners with normal hearing transcribed the sentences in quiet, in speech-shaped noise at −3 dB SNR, and in 12-talker babble at −3 dB SNR. Listeners also completed a dialect familiarity questionnaire. The results will shed additional light on the interactions between talker and listener dialect, dialect familiarity, and background noise type.

4aSCb16. Perceptual representation of speaker gender in Spanish-English bilingual listeners. Brandon Merritt (Speech, Lang., and Hearing Sci., The Univ. of Texas at El Paso, 734 S. Mesa Hills Dr., #72, El Paso, TX 79912, bmmerritt@utep.edu)

The perceptual representation of speaker gender in monolingual English listeners has been found to be gradient and 2-dimensional. However, both gender expression and listener attribution of a speaker's gender are known to vary by cultural and linguistic norms. Thus, listeners' attribution of a speaker's gender is expected to vary based on the cultural and linguistic practices of their community. This study examined the perceptual representation of speaker gender in Spanish-English bilingual listeners at the U.S./Mexico border of El Paso, TX. Twenty-four Spanish-English bilingual speakers of diverse gender identities (e.g., cisgender men, cisgender women, and transgender women) were audio recorded reading sentences from the English and Spanish versions of the Hearing in Noise Test. Nineteen cisgender Spanish-English bilingual listeners completed an auditory free classification paradigm, in which they classified speakers by perceived general similarity and gender identity in both Spanish and English conditions. Multidimensional scaling of listeners' classifications in each language revealed that listeners organized speakers in a more expansive perceptual space in English (3 dimensions) as compared to Spanish (2 dimensions). Dimension weightings indicated that, in Spanish, listeners placed more emphasis on Dimension 1 as compared to Dimension 2 when classifying speakers, while, in English, listeners equally weighted the 3 dimensions.

4aSCb17. Effects of quality of initial exposure stimuli on speech intelligibility. Seung-Eun Kim (Linguist., Northwestern Univ., 2016 Sheridan Rd., Evanston, IL 60208, seungeun.kim@northwestern.edu), Matthew Goldrick (Linguist., Northwestern Univ., Evanston, IL), and Ann Bradlow (Linguist., Northwestern Univ., Evanston, IL)

Listeners can use lexical information to drive adaptation to talker-specific speech characteristics with a small amount of exposure. This suggests that adaptation may be strongest when listeners are initially exposed to speech in a condition that facilitates word recognition. This was tested by examining the intelligibility of second-language (L2) speech-in-noise when stimuli were initially presented in Quiet versus low signal-to-noise ratio (SNR) conditions. Recordings of 10 L2 English talkers were tested. For each talker, 120 sentences were mixed with speech-shaped noise ranging from −4 dB to 8 dB in steps of 2 dB and in Quiet (15 sentences at each SNR). These stimuli were presented from lowest to highest SNR (forward; −4 dB to Quiet) in one group of L1 English listeners (n = 10/talker) and from highest to lowest SNR (reverse; Quiet to −4 dB) in the other (n = 10/talker). The results showed that early exposure to a clean signal led to better speech perception. At lower SNRs, word recognition accuracy in the reverse order was significantly higher than the forward order, resulting in a steeper slope of psychometric function of intelligibility in the forward versus reverse condition. The quality of initial speech input may influence talker adaptation, particularly in adverse listening conditions.

4aSCb18. Non-native clear speech increases intelligibility but not through improved word segmentation: Evidence from a visual-world eye-tracking study. Madeline Smith (Dept. of Linguist., Univ. of Texas at Austin, 305 E. 23rd St., Stop B5100, Austin, TX 78712, madelinesmith@utexas.edu), Zhe-chen Guo (Dept. of Commun. Sci. and Disord., Northwestern Univ., Evanston, IL), and Rajka Smiljanic (Dept. of Linguist., Univ. of Texas at Austin, Austin, TX)

Listener-directed hyperarticulated clear speech produced by native (L1) talkers improves word segmentation and reduces lexical competition. Less is known about whether non-native (L2) clear speech also confers such benefit. In a visual-world eye-tracking study, we investigated if L2 clear speech improves word segmentation and the time course of the benefit for native listeners. Forty L1 English participants heard sentences produced in conversational and clear styles by a highly intelligible L2 English / L1 Spanish speaker with a discernable non-native accent. Sentences contained a target word (e.g., *doll*) with which a corresponding competitor overlapped phonemically (e.g., *dolphin*), creating temporary ambiguity with the target and the following word's onset (e.g., *doll found*). Each recording was presented in quiet alongside pictures of the target, competitor, and two distractors. Participants were instructed to select the picture mentioned in the sentence they heard. No significant clear speech segmentation advantage was found; the proportion of looks to targets over competitors indicates similar time course of disambiguation in both conversational and clear speech. The results suggest that L2-accented clear speech with its deviations from the target-language-specific modifications and greater phonetic variability increases signal uncertainty resulting in no benefit for word segmentation even though word recognition was improved.

4aSCb19. Assessing retroflex and alveolar liquid perception and production in heritage Tamil speakers. Kamala Muthukumarasamy (Carleton Univ., 1125 Colonel By Dr., Ottawa, ON K1S 5B6, Canada, kamalamuthukumarasam@cmail.carleton.ca) and Chandan Narayan (York Unversity, Toronto, ON, Canada)

In this presentation, we explore factors affecting and the relationship between perception and production of the alveolar-retroflex liquid contrast ([l]-[èÙ]) in heritage Tamil speakers. In particular, we examine the connection between language and identity in the maintenance of this acoustically fragile contrast with similar spectral characteristics, shown through low perceptual salience. Tamil speakers (n = 18) completed an AX discrimination task with non-word Tamil VCVs was administered. D-prime, a bias free measure of perceptual distance, was computed from the discrimination data. Additionally, participants provided minimal pairs with the target consonants in an elicited production task. An F3-F2 (Hz) score was taken as a measure of productive salience, with alveolars having a larger difference than retroflexes. Quantitative results showed a high degree of variation in productive salience, as some speakers clearly produce the contrast while others did not. Perceptual distance was also variable, with some participants clearly showing categorical discrimination while others did not. Qualitative results revealed that a concrete ties to tangible culture and a strong linguistic identity can serve as an indicator of accuracy in perception and production. This research addresses whether acoustically fragile contrasts are realized in heritage Tamil and provides new insight into heritage phonological contrast retention and maintenance.

4aSCb20. Durational and spectral factors in judgements of American Raising. Elliott Moreton (Linguist., Univ. of North Carolina, Chapel Hill, Dept. of Linguist., CB #3155, Chapel Hill, NC 27599-3155, moreton@email.unc.edu), Jeff Lamontagne (French and Italian, Indiana Univ., Bloomington, IN), and Monica Nesbitt (Linguist., Indiana Univ., Bloomington, IN)

Canadian Raising and its relatives in the USA can respond to underlying voicing of flapped /t/ (Raised "writer" vs. un-Raised "rider") from the earliest stages (Fruehwald 2016). How so, if Raising is phonologized from a phonetic precursor sensitive only to phonetic features? Proposal (Bermúdez-Otero 2019): Raising responds to duration, not voicing: diphthongs are shortened before underlyingly voiceless sounds by a pre-existing lexical phonological rule of Pre-Voiceless Clipping, after which postlexical Raising transparently affects the shortened diphthongs. Experiment: Participants

(N = 141, US dialects) read wordlists with /ai/ and /ei/ in voiceless and voiced contexts, then sorted the words into groups judged to share a vowel (DiPaolo & Faber, 1990). Clipping and Raising were measured using duration, F1, and F2. Predictions: (1) across speakers, /ai/-Clipping and /ai/-Raising should be positively correlated. (2) /ai/-sorting should be better predicted by /ai/-Clipping than by /ai/-Raising, because lexical rules change phonemes, while postlexical ones are subphonemic. (3) /ei/-sorting should be positively correlated with /ai/-sorting, since Clipping affects all vocoids. Results: (1) /ai/-Clipping did *not* predict /ai/-Raising ($r = -0.082$); (2) /ai/-Clipping predicted word sorting marginally *worse* than /ai/-Raising (95% CI for $r_{clipping}$-$r_{raising} = (-0.50, 0.03)$). (3) /ei/ and /ai/ judgements *were* positively correlated ($r = 0.37$, 95% CI = (0.21, 0.51)).

4aSCb21. Perceptual adaptation to unfamiliar dialects in an eye-tracking task. Larisa Bryan (Ohio State Univ., 1712 Neil Ave., Columbus, OH 43210, bryan.368@osu.edu) and Cynthia G. Clopper (Ohio State Univ., Columbus, OH)

Variation in pronunciation of vowels across American English dialects often creates the strongest distinctions between dialects and leads to lexical ambiguity between minimal pairs within dialects. This lexical ambiguity is stronger for listeners with lifetime exposure to multiple dialects than to a single dialect. In the current study, we investigated the influence of brief, in-lab dialect exposure on lexical competition and perceptual adaptation. Midwestern American English participants were first presented with a familiarization short story told in one of two unfamiliar dialects (Southern American English or a novel accent) and then completed a visual-world eye-tracking task containing both acoustically ambiguous and non-ambiguous words from each dialect in a forced-choice task. The results showed evidence of greater lexical competition for target dialect minimal pairs following familiarization. This pattern of results is consistent with previous work suggesting greater lexical competition following lifetime exposure to dialect variation among geographically mobile listeners, suggesting similarities in the effects of brief in-lab and lifetime exposure to dialect variation on lexical competition and processing.

4aSCb22. How rhythmic cues influence perception of accent distance across L1 and L2 English speakers. Holly Lind-Combs (Speech and Hearing Sci., The Ohio State Univ., 1070 Carmack Rd., Columbus, OH 43210, lind-combs.1@osu.edu), Rachael F. Holt (Speech and Hearing Sci., The Ohio State Univ., Columbus, OH), Tessa Bent, and Malachi Henry (Speech, Lang. and Hearing Sci., Indiana Univ., Bloomington, IN)

Segmental and suprasegmental cues can both contribute to listeners' accent strength and distance judgments. This study examined the influence of a suprasegmental cue—rhythm—on listeners' judgments of accent distance. Rhythm was quantified using two measures of variability in vocalic interval duration: normalized pairwise variability index (nPVI) and VarcoV. Thirty adults ranked productions from female and male speakers from six L1 and six L2 English varieties (24 speakers) across six sentences based on their perceived distance from Midland American English (the local dialect). Linear mixed effect models predicted accent distance ratings from rhythm measures—specifically, differences in nPVI and VarcoV between the L1/L2 speakers and Midland speakers, with speaking rate and sentence length as covariates. Differences between L1/L2 speakers' and Midland speakers' nPVI and VarcoV scores did not significantly predict accent distance rankings. However, significant interactions between nPVI and certain accents (e.g., Irish, Southern American, Cantonese-accented English) suggest that rhythmic properties may significantly impact accent distance judgements for some accent varieties. These results provide further insight into how rhythmic cues contribute to accent perception. [Work supported by NSF grants 1941691 and 1941662].

4aSCb23. The contribution of suprasegmental cues to perceived L1 and L2 accent distance. Malachi Henry (Speech, Lang. and Hearing Sci., Indiana Univ., 1579 S Renwick Blvd, Bloomington, IN 47401, mjhenry5519@gmail.com), Tessa Bent (Speech, Lang. and Hearing Sci., Indiana Univ., Bloomington, IN), Rachael F. Holt, and Holly Lind-Combs (Speech and Hearing Sci., Ohio State Univ., Columbus, OH)

First (L1) and second language (L2) pronunciations that diverge more segmentally from the local accent are rated as more distant. However, fewer studies have addressed relations between suprasegmental features, such as intonation, and accent distance. We examined the relation between perceived accent distance and four suprasegmental measures: time normalized f0 Euclidean distance, mean F0, pitch range, and speaking rate. Thirty adults completed six ladder tasks in which listeners ranked talkers' perceived distance from the local accent. For each ladder, listeners heard one sentence produced by 24 talkers: one man and one woman from six L1 and six L2 English varieties. Each suprasegmental measure independently predicted distance rankings with a variety of significant interactions. Talkers with higher accent rankings had narrower pitch ranges, lower mean f0, faster speech, and greater f0 Euclidean distance. Furthermore, suprasegmental measures related to accent distance differentially based on syntactic structure. For example, as f0 Euclidean distance increased, accent rankings increased for interrogative sentences, but not for declarative sentences. These data support the need for future studies examining how suprasegmental features impact perception of L1 and L2 accents. [Work supported by NSF grants 1941691 and 1941662].

4aSCb24. Social network characteristics impact the recognition of degraded speech by adult cochlear implant users. Terrin N. Tamati (Dept. Otolaryngol., Vanderbilt Univ. Medical Ctr., 1608 Aschinger Blvd, Columbus, OH 43212, terrintamati@gmail.com), Emily M. Clausing (Ohio State Univ., Columbus, OH), and Victoria A. Sevich (Speech and Hearing Sci., The Ohio State Univ., Columbus, OH)

Social networks influence the quantity and quality of linguistic input experienced in everyday listening environments. In normal-hearing listeners, the variable linguistic input provided by larger and more diverse social networks has been shown to support speech processing. However, for adult cochlear implant (CI) users, limitations in the perception of linguistic and talker details may limit input variability benefits. The current study examined the effects of social network size and diversity on the recognition of spectrotemporally degraded speech by adult CI users. Twenty-six postlingually deafened adult CI users completed a detailed questionnaire about their regular communication partners. Social network size was calculated as the number of regular communication partners, and social network diversity was calculated as the degree of age, education, and accent heterogeneity among communication partners. Social network metrics were compared to vowel, word, and sentence recognition accuracy scores, also controlling for basic auditory ability. Results showed that social network age diversity was moderately to strongly correlated with word and sentence recognition accuracy. Social network size and other diversity metrics were not related to word or sentence recognition accuracy. These findings suggest that more diverse input from different age groups facilitates spoken word recognition in adult CI users.

4aSCb25. Neural oscillation to music processing in children with different language backgrounds. Maxfield Rodgers, Kristal Reyes, Faith Chai, Blessy Gill, Angela Cheng (St. John's Univ., Queens, NY), and Yan H. Yu (St. John's Univ., 4631 216 St., Bayside, NY 11361, yanhyu@gmail.com)

Musical experience and tonal language experience are both associated with structural and functional changes in the brain that underlie facilitated pitch perception. We investigated whether neuronal oscillations in response to music were correlated with musical expertise and whether they reflected the language backgrounds (tonal versus nontonal, monolingual versus bilingual). We measured music processing in bilingual children (5–10 years old) from Mandarin (a tone language) households and three groups of age-matched children from non-tone language households (Bilingual Spanish-English, monolingual mainstream American English (MAE), and African American English (AAE)). Event-related brain potentials (ERPs) were

recorded in an oddball paradigm with six types of music changes. We analyzed the phase locking and amplitude modulations of ongoing oscillations in the theta (4–8 Hz) alpha (8–14 Hz), beta- (14–30 Hz), and gamma- (30–80 Hz) bands to music changes. Preliminary results suggest that musical expertise was associated with strengthened phase locking of neural oscillations to most music features. Tonal language experience was also associated with more robust phase locking. Bilingual experience alone does not show any enhancement in neural oscillation. There were no clear advantages of music processing for the bilingual experience.

4aSCb26. Fundamental frequency patterns across gender, task, and language in Spanish-English bilinguals. June M. Contreras (Speech Lang. and Hearing Sci., Univ. of Texas at El Paso, 3532 Keltner, El Paso, TX 79904, jmcontreras2@miners.utep.edu) and Brandon Merritt (Rehabilitation Sci., The Univ. of Texas at El Paso, El Paso, TX)

Speaking fundamental frequency (SFF) patterns are known to vary by language and speaker gender. Relatively little data exist for SFF patterns among cisgender and transgender speakers, especially those who speak languages other than English. Further, speaking task (e.g., read versus spontaneous speech) may impact SFF patterns. This study examined SFF patterns in Spanish-English bilinguals of varying gender identities. Twenty-four speakers (8 cisgender men, 8 cisgender women, and 8 transgender women) recorded a read passage and spontaneous speech in Spanish and English. SFF measures of minimum, maximum, range, and median were found to be stable across speech tasks and languages. A significant effect of gender was found. Cisgender women on average produced the highest values of minimum, maximum, and median SFF and largest SFF range. Cisgender men produced the lowest values and smallest range. Transgender women produced SFF measures within an intermediate range of cisgender men and women. Compared to monolingual English speakers, Spanish-English bilingual cisgender women had a larger SFF mean for both Spanish and English, cisgender men had similar SFF means for both Spanish and English, and transgender women had a slightly larger SFF mean for both Spanish and English.

4aSCb27. Intelligibility, recall, and voice evaluation across accents. Line Lloy (Linguist., Univ. of BC, Vancouver, BC, Canada), Nikolai A. Schwarz-Acosta (Spanish and Portuguese, Univ. of California - Berkeley, Emeryville, CA), and Molly Babel (Linguist., Univ. of BC, 2613 West Mall, Vancouver, Vancouver, BC V6T 1Z4, Canada, molly.babel@ubc.ca)

Speech elicits variable responses from listeners. Voices can vary in their intelligibility, how well listeners can recall messages produced by the voice, and what kind of social evaluation it explicitly or implicitly evokes. These different responses may be due to individual-specific attributes within a voice or the accent it carries. For example, familiar or more standard language varieties may be more intelligible producing more easily recalled, and eliciting more positive social evaluations from listeners. The current study uses 35 English-speaking voices from 7 different language backgrounds that vary in familiarity and prestige to the listener ($n = 430$) population, which is a representative heterogeneous sampling from the local university community. Specific voices were chosen from a larger data set based on their acoustic similarity. Listeners either completed a speech transcription task (quantifying intelligibility) or a cloze task (quantifying recall) and all listeners provided an evaluation of the voices' likability and perceived comprehensibility. Bayesian data analysis is used to quantify and characterize the relationship between a voice's intelligibility and how well it is recalled, and whether this relationship is predicted by social evaluation and listener experience. These results have implications for theories of speech recognition and how listeners process accents.

4aSCb28. The Accent Atlas: Effects of long-term exposure to nonnative accents on adaptive speech perception. Yuting Gu (Univ. of California Irvine, Irvine, CA), Seth Cutler, Chigusa Kurumada (Univ. of Rochester, Rochester, NY), and Xin Xie (Univ. of California Irvine, SSPB 2223, University of California Irvine, Irvine, CA 92617, xxie14@uci.edu)

As mobility increases and virtual interactions grow, quickly adapting to diverse accents is essential for effective communication. While lab-based accent training promotes rapid adaptation to unfamiliar accents, whether prolonged environmental exposure to linguistic diversity yields similar benefits is unknown. We conducted a large-scale perceptual experiment involving 600+ participants across 15 U.S. states, utilizing a cross-modal word matching task to measure speech recognition in both non-native accented English and native English with background noise. Participants were recruited from linguistically diverse (LD) or linguistically homogeneous (LH) states. Linguistic diversity was defined based on the prevalence of non-English speaking residents (U.S. census data). Our findings were two-fold: firstly, consistent with prior studies, only exposure to non-native accented speech—but not exposure to speech in noise—improved recognition of novel talkers with the same accent; secondly, individuals from the LD states outperformed those from LH areas in initial accented speech perception. However, by the end of the experiment, benefits from in-lab accent exposure did not differ significantly between the two groups, suggesting that short-term training could mitigate long-term environmental influences. These results illuminate the interplay between daily linguistic environments and accent adaptation, enhancing our understanding of speech perception's adaptivity in a multilingual context.

4aSCb29. Perception of speaker identity for bilingual voices. Sylvia Cho (Linguist., Simon Fraser Univ., 8888 University Dr., Burnaby, BC V5A 1S6, Canada, sylvia_cho@sfu.ca)

Voice is often described as an "auditory face"; it provides important information concerning speaker identity (e.g., age, height, sex). The acoustic properties related to voice can also vary substantially within a speaker based on one's emotional, social, and linguistic states. Recent work suggests that biological components have the greatest impact in the acoustic variability found in voice, followed by language-specific factors and speaking style [Lee & Kreiman, J. Acoust. Soc. Am. 153, A295 (2023)]. The effects of such within- vs. between-speaker acoustic variability on the perception of speaker identity, however, have not been explored. The present study therefore examines the perception of speaker identity in bilingual voices. The prediction is that acoustic variability will also affect speaker identity perception: voices will be discriminated best for between-speaker samples, while within-speaker variability will not affect perception of speaker to the same extent. To test this prediction, listeners participated in a voice discrimination task using bilingual voice data produced by Korean heritage speakers across different languages (Korean, English) and speech styles (read, extemporaneous). The data will be analyzed to measure the effects of speaker, language, and speech style on voice discrimination. The results will be reported in relevance to the relationship between bilingualism and speech style on voice quality and speaker identity.

4aSCb30. Tongue bracing robustness under perturbation: Comparison of L1 versus L2 speech with and without bite-block. Abiodun Ibikunle (Linguist., Univ. of BC, Vancouver, BC, Canada), Marcell Maitinsky (Linguist., Univ. of BC, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, mlmtnsky@student.ubc.ca), Annabelle Purnomo, Chenxi Xu, Yadong Liu, and Bryan Gick (Linguist., Univ. of BC, Vancouver, BC, Canada)

Lateral tongue bracing is a universal speech posture observed across many languages [Gick *et al.*, 2017, JSLHR 60; Liu *et al.*, 2022, JIPA, Phonetica 79]. Previous work has found that under perturbation by 10 mm bite-blocks, there is less lateral contact during a speaker's L2 speech compared to their L1 speech [Bengtson *et al.*, 2023, J. Acoust. Soc. Am. 154]. However, the research did not investigate lateral contact without bite-blocks and the effect of this perturbation on posture remains unclear. The present study addresses this by comparing a speaker's L1 and L2 speech both with and without bite-block perturbation, with the hypothesis that posture during speech movement is more robustly maintained under perturbation with more language experience. Participants will read two short texts in their L1 and L2 separately, both with and without 10 mm bite-blocks, and F1/F2 of cardinal vowels will be compared between bite-block conditions for each language. L2 proficiency will be collected for each participant and results will be reported with relevance to posture and speech movement coordination for L1 versus L2, with implications discussed for proficiency. [Work supported by NSERC].

4aSCb31. Individual differences in perception of Thai and Korean stops. Melissa Baese-Berk (Linguist., Univ. of Chicago, 1115 E 58th St., Rosenwald Hall Rm. 203, Chicago, IL 60657, mmbb@uchicago.edu) and Charlie Nagle (Univ. of Texas, Austin, TX)

Differentiating between pairs of unfamiliar speech sounds is often challenging for adult listeners. However, this difficulty varies depending on multiple factors. For example, it is clear that the relationship between a listener's first language and the target language can impact the ease or difficulty of differentiating pairs of sounds. However, individuals also vary widely in their ability to differentiate these novel sounds, even if they share a language background. In this study, we compare native English listeners' ability to differentiate between initial stop consonants in Korean and in Thai. In addition to comparing individual performance across these two target languages, we also assess performance as a function of various linguistic and cognitive measures including phonological short-term memory, language learning aptitude measures, and L1 production and perception patterns.

THURSDAY MORNING, 16 MAY 2024 | ROOM 213, 8:00 A.M. TO 11:55 A.M.

Session 4aSP

Signal Processing in Acoustics, Acoustical Oceanography, Physical Acoustics, Computational Acoustics, and Underwater Acoustics: Bayesian and Machine Learning in Acoustics II

Yangfan Liu, Cochair
Purdue Univ., Ray W. Herrick Laboratories, Purdue University, 177, South Russell Street, West Lafayette, IN 47907-2099

Ning Xiang, Cochair
School of Architecture, Rensselaer Polytechnic Institute, 110 Eighth Street, Troy, NY 12180

Paul J. Gendron, Cochair
ECE, University of Masschusetts Dartmouth, 285 Old Westport Rd., Dartmouth, MA 02747

Chair's Introduction—8:00

Invited Papers

8:05

4aSP1. Application of machine learning on complete vehicle data for interior vibration prediction based on transfer paths in narrowband spectrum. Marcus Maeder (School of Eng. and Design, Tech. Univ. of Munich, Garching near Munich, Germany, marcus.maeder@tum.de), Martin Eser, and Steffen Marburg (School of Eng. and Design, Tech. Univ. of Munich, Garching, Germany)

Interior noise and vibration are critical quality criteria for vehicles in the automotive sector. The latter is essential in driving comfort, where many development resources are used for the design and the evaluation. As a result, testing the entire vehicle is very important to assess vibration and ride comfort. The assembly of the entire vehicle is carried out virtually using transfer functions derived either from simulations or from measurements. Since simulations of a complete vehicle are still challenging,transfer functions are usually measured with considerable effort. Therefore, limiting the necessary measurements to specific components is desirable to be efficient and to save costs. In this work, the authors use an artificial neural network to predict the interior vibration profiles of a complete vehicle based on full test drives at different speeds and road conditions. Triaxial acceleration measurements serve as the database. In addition, a criterion is proposed to select essential sensors for the learning process. The results show that if the data is handled carefully, many sensors can be discarded, and the network can predict accurate acceleration spectra for virtual sensors at various test conditions.

8:25

4aSP2. Implementation of supervised machine learning classification for detection and severity determination of electrified power train noise and vibration fault diagnosis. Joohyun Lee (Ray W. Herrick Laboratories/School of Mech. Eng., Purdue Univ., 3977 State Rd. 38 E, Apt 207, Lafayette, IN 47905, lee2243@purdue.edu), Yangfan Liu, J. S. Bolton, and Patricia Davies (Ray W. Herrick Laboratories/School of Mech. Eng., Purdue Univ., West Lafayette, IN)

Vehicles equipped with electrified powertrains produce lower sound and vibration levels compared to those equipped with internal combustion engine powertrains. This makes noise and vibration (N&V) from other non-engine components more perceptible. Gear growl is one of the newly observed N&V that brings concerns by the passengers and manufacturers. The understanding of signal characteristics and the threshold for determining whether gear growl requires attention remains limited. To address this, supervised machine learning classification is employed. Root-mean-square (RMS) and spectral entropy values are sufficient for the classification of vibration data with test accuracy of 0.983. However, the acoustic signal required more features due to background noise, making data linearly inseparable. Features that describe the characteristics of acoustic data are studied, extracted, and selected. Utilizing a support vector machine (SVM) for classification, the study achieves an average test accuracy of 0.918. Further, a multi-class classification model is implemented based on preliminary subjective listening studies, classifying different severities of gear growl. Further listening studies are suggested for improving multi-class classification performance. Methods described in this study mainly focus on the analysis of gear growl, but they can be generalized for N&V signal-based fault diagnosis applications.

Contributed Paper

8:45

4aSP3. Detecting ringed seal vocalizations in multiple environments using deep learning. Karlee Zammit (Earth and Ocean Sci., Univ. of Victoria, 504 Ridgebank Crescent, Victoria, BC V8Z4X9, Canada, karlee.zammit@gmail.com), William D. Halliday (Wildlife Conservation Society Canada, Whitehorse, YT, Canada), Fabio Frazao (Comput. Sci., Dalhousie Univ., Halifax, NS, Canada), Sebastien Fabbro (Phys., Univ. of Victoria, V8Z4X9, BC, Canada), and Stan Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada)

Deep learning methods have recently been successfully applied to create a variety of automated acoustic detectors in the field of marine bioacoustics. Automated detectors are essential for analyzing large volumes of passive acoustic monitoring (PAM) data since manual analysis is prohibitively time-consuming and costly. PAM is the primary method for obtaining data on species which are endemic to remote regions, such as the Canadian Arctic. Arctic ringed seals are listed as a Species of Special Concern in Canada due to a loss of critical habitat caused by the effects of climate change. Here, ResNet, a convolutional neural network architecture, is trained on thousands of examples of ringed seal vocalizations recorded at various locations within the Canadian Arctic to create the first practical automated ringed seal detector. The network achieves a precision of 0.89, recall of 0.80, and F1 score of 0.85 when tested on 215 five-minute recordings from sites included in the training process. To improve the generalizability of the detector for new locations, fine-tuning is performed using a small subset of annotated data from new sites. The detector will be available as an open-source tool for researchers to use as the basis for further development of new automated detectors.

Invited Paper

9:00

4aSP4. Application of deep learning methods for obstacle classification in ultrasonic surround sensing. Jona Eisele (Chair of Vibroacoustics of Vehicles and Machines, Tech. Univ. of Munich, Boltzmannstr. 15, Garching near Munich 85748, Germany, jona.eisele@tum.de), André Gerlach (Bosch Res., Robert Bosch GmbH, Renningen, Germany), Marcus Maeder (Chair of Vibroacoustics of Vehicles and Machines, Tech. Univ. of Munich, Garching near Munich, Germany), Andreas Koch (Inst. of Appl. Artificial Intelligence, Stuttgart Media Univ., Stuttgart, Germany), and Steffen Marburg (Chair of Vibroacoustics of Vehicles and Machines, Tech. Univ. of Munich, Garching, Germany)

Ultrasonic sensors are widely used in driver assist systems for close-range surround sensing. Based on the pulse-echo method, the distance to obstacles is calculated, particularly in scenarios such as parking and maneuvering. For advanced ultrasonic perception systems, however, the classification of obstacles should be considered, too. For instance, it is relevant to distinguish between traversable and non-traversable objects, as well as to detect pedestrians. Classifying obstacles by ultrasonic echoes remains an area of ongoing research, and, in contrast to ultrasonic ranging, implementation in current systems is rudimentary at best. This work examines the application of deep learning methods for ultrasonic-based object classification in driver assist systems. Features based on raw time signals, amplitude- and phase-based echo features as well as time-frequency images are compared. The preprocessing and feature extraction methods are evaluated based on experimental data captured in low and high clutter environments. Promising classification results are achieved using convolutional neural networks, demonstrating a significant improvement over traditional methods. Finally, we discuss the benefits of employing small aperture ultrasonic sensor arrays to increase classification robustness using beamforming methods.

Contributed Papers

9:20

4aSP5. Direction of arrival estimation with convolutional neural networks and multiple signal classification. Christopher J. Bell (Univ. of Rhode Island, The Fascitelli Ctr. for Adv. Eng., Kingston, RI 02881, cj.bell91@gmail.com), Kaushallya Adhikari (Univ. of Rhode Island, Kingston, RI), and Lauren Freeman (NUWC Newport, Newport, RI)

Recently, there has been a proliferation of applied machine learning (ML) research, including the use of convolutional neural networks (CNNs) for direction of arrival (DoA) estimation. With the large increase of research in this area, it is important to balance the performance and computational costs of CNNs with classical methods of DoA estimation such as Multiple Signal Classification (MUSIC). We outline the performance of both methods of DoA estimation for single source and two source cases and compare them to the Cramer-Rao lower bound (CRLB). For each source case, a CNN was trained for a perfect uniform line array (ULA), a perturbed ULA, and a ULA with missing sensors. The three cases are interesting studies as classical methods for DoA estimation such as MUSIC assume perfect array conditions, whereas the CNNs assume nothing about the structure of the data. The training data for each network was created by leveraging a signal model to create synthetic data at different SNRs. The results for each network are then compared to the results for MUSIC using the same signal case and array condition. The results indicate that for the single source case the CNN only performs significantly better than MUSIC for the perturbed array case. For the two-source case, the CNNs significantly outperform MUSIC for all ULA conditions, however, when compared to the CRLB it is shown that the CNN typically produces a biased estimate.

9:35

4aSP6. Physics informed sound field interpolation using an acoustic sensor network. Rashen Fernando (Elec. and Comput. Eng., Univ. of Illinois at Chicago, 2955 South Emerald St., Unit 1F, Chicago, IL 60616, pferna20@uic.edu), Manan Mittal, Yongjie Zhuang, Ryan M. Corey, and Andrew C. Singer (Elec. and Comput. Eng., Stony Brook Univ., Stony Brook, NY)

Sound field estimation is the process of analyzing and characterizing the distribution of sound waves in a particular physical space. The applications of sound field estimation extend to various areas, including the visualization of acoustic fields, interpolation of room impulse responses, identification of sound sources, capturing sound fields for spatial audio, and spatial active noise control, among other potential uses. In a previous study, a directionally weighted kernel has been used to estimate the sound field, where the priori information of source directions is employed to improve the estimation accuracy. In another separate study, spherical harmonics have been used to represent the sound field. However, the order of spherical harmonic coefficients was limited due to the limited number of microphones. This research introduces a novel method for sound field estimation using multiple microphones to sample a source-free volume. A physics-informed neural network is used to predict the spherical harmonics coefficients and locations of unknown sources to estimate the sound field.

9:50–10:05 Break

10:05

4aSP7. Development of the 2nd Cadenza challenge for improving music listening for people with a hearing loss. Michael A. Akeroyd (School of Medicine, Univ. of Nottingham, Nottingham, United Kingdom), Scott Bannister (Univ. of Leeds, Leeds, United Kingdom), Jon P. Barker (Univ. of Sheffield, Sheffield, United Kingdom), Trevor J. Cox, Gerardo Roa, Bruno Fazenda (Univ. of Salford, Salford, United Kingdom), Jennifer L. Firth (Univ. of Nottingham, Nottingham, United Kingdom), Simone Graetzer (Univ. of Salford, Salford, United Kingdom), Alinka Greasley (Univ. of Leeds, Leeds, United Kingdom), Rebecca Vos (Univ. of Salford, Salford, United Kingdom), and William M. Whitmer (Hearing Sci. - Scottish Section, Level 3, New Lister Bldg., Glasgow Royal Infirmary, Glasgow G31 2ER, United Kingdom, bill.whitmer@nottingham.ac.uk)

The Cadenza project is an ongoing project that aims to improve music quality for those with a hearing loss. The project is running signal-processing and machine-learning challenges to address different listening issues and scenarios. During the first round, the challenge focused on non-causal music source separation to allow remixing for those with hearing loss. This fed into an ICASSP 2024 challenge, which had crosstalk from loudspeaker reproduction included. There are three potential arms to our upcoming 2024 challenge based on reported issues from hearing-impaired music listeners: (1) low-latency causal audio source separation, (2) lyric intelligibility enhancement without loss of timbre or instrumental balance, and (3) loudness/dynamic range control. Each of these potential challenges raise questions as to the appropriate reference signals, as well as the practicalities of deriving the appropriate signals for an open-source machine-learning challenge. [Work supported by UK EPSRC Grant No. EP/W019434/1]

10:20

4aSP8. Development of AcousticsAI, a large language model chatbot trained on peer-reviewed acoustics literature. Nathan Leo (Eng. Sci. and Mech., The Penn State Univ., 212 Earth and Eng. Sci. Bldg, University Park, PA 16802, nal5367@psu.edu) and Christopher M. Kube (Eng. Sci. and Mech., The Penn State Univ., University Park, PA)

Large Language Models (LLMs) like ChatGPT are altering how we interact and extract useful knowledge. However, the ability to extract domain specific knowledge about acoustics is still lacking. This presentation highlights our efforts in developing AcousticsAI, which is a LLM chatbot trained specifically with peer-reviewed acoustics literature. AcousticsAI is being developed as an advanced tool to aid acousticians at the cutting edge of acoustics research. Being research focused, AcousticsAI must be able to interface the many different data streams extracted from peer-reviewed literature. This is accomplished through AutoGen, a multi-agent conversation system, enhances Large Language Models (LLMs) workflows by enabling conversable, customizable agents that incorporate human inputs and tools. AutoGen helps reduce the response correlation space, ensuring consistent and accurate LLM outputs with citations. By deploying specialized, context-specific models within the AutoGen framework, the study demonstrates improved precision and reliability in semantic reasoning over commonly used AI chatbots.

Invited Paper

10:35

4aSP9. Speech detection models for effective communicable disease risk assessment in air travel environments. Tenon Charly Kone (Flight Res. Lab., National Res. Council Canada, 1200 Montreal Rd., Ottawa, ON, Ottawa, ON K1A 0R6, Canada, tenoncharly.kone@nrc-cnrc.gc.ca), Sebastian Ghinet, Sayed Ahmed Dana, and Anant Grewal (Flight Res. Lab., National Res. Council Canada, Ottawa, ON, Canada)

In environments characterized by elevated noise levels, such as airports or aircraft cabins, travelers often find themselves involuntarily speaking loudly and drawing closer to one another in an effort to enhance communication and speech intelligibility. Unfortunately, this unintentional behaviour increases the risk of respiratory particles dispersion, potentially carrying infectious agents like bacteria

which makes the contagion control more challenging. The accurate characterization of the risk associated to speaking, in such a challenging noise environment with multiple overlapping speech sources, is therefore of outmost importance. Among the most advanced signal processing strategies that can be used to accurately determine who spoke when and with whom and for how long but most importantly how loudly, at one location, artificial intelligence-based speaker diarization approaches were considered and adapted for this task. This article details the implementation of speaker diarization algorithms, customized to extract speaker and speech parameters discreetly. Validation and preliminary study results are also provided. The algorithms calculate speech duration and sound pressure level for each sentence and speaker, aiding in assessing viral contaminant spread. The paper focuses on applying these algorithms in noisy environments, particularly in confined spaces with multiple speakers. The findings contribute to proactive measures for containing and managing communicable diseases in air travel settings.

Contributed Papers

10:55

4aSP10. Using convolutional neural networks to select the optimal wavelet for audio compression. Shaun Pies (Phys., Univ. of New Orleans, 2000 Lakeshore Dr., New Orleans, LA 70148, spies@uno.edu), Kendal Leftwich, and Juliette W. Ioup (Phys., Univ. of New Orleans, New Orleans, LA)

The vast amount of data generated and stored across domains is overwhelming. Noteworthy examples include the Event Horizon Telescope using 3.5 petabytes for the first black hole image, LIGO amassing 4.5 petabytes, and CERN with 487 petabytes. This research addresses the challenge of data storage and transmission using machine learning and wavelet analysis. The goal is to create an efficient data compression method that significantly reduces storage needs without compromising data integrity. Wavelets, mathematical tools which break down signals into coefficients, show promise for compression. Choosing the right wavelet is crucial; our research demonstrates that machine learning can effectively solve this problem. By training a convolutional neural network on spectrograms of audio files, we identify the optimal wavelet for data compression, achieving up to 50% compression with minimal loss.

11:10

4aSP11. Testing transfer learning for source ranging in a laboratory tank. Corey E. Dobbs (Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, cedobbs@byu.edu) and Tracianne B. Neilsen (Phys. and Astronomy, Brigham Young Univ., Provo, UT)

One difficulty in applying deep learning techniques to ocean acoustics is the spatially and temporally varying environmental properties. Another challenge is the lack of labeled data for training large networks. The overall goal of this work is to develop deep learning approaches that can be adaptable to different conditions. For example, a trained neural network will fail to generalize when the appropriate environmental variability is not included in the training data. To improve network performance, transfer learning can modify a pre-trained network to make predictions on data that was recorded under different conditions than the original dataset. In this work, a convolutional neural network was trained on acoustic data measured in a water tank, while the water was at room temperature, to predict source-receiver range. Transfer learning was used to update the pre-trained model with a smaller set of data measured at different water temperatures. The resulting model better generalizes to measurements at different temperatures. This approach illustrates how transfer learning can be used in ocean acoustics to improve generalizability in a specific area with less labeled data and lower computational cost. [Work supported by the Office of Naval Research, Grant N00014-22-12402.]

11:25

4aSP12. Machine learning for analysis of wind farm noise. Heather L. Lai (Eng. Programs, State Univ. of NY at New Paltz, 208 Eng. Innovation Hub, SUNY New Paltz, New Paltz, NY 12561, laih@newpaltz.edu), Anne C. Balant (Commun. Disord., State Univ. of NY at New Paltz, New Paltz, NY), and Chih-Yang Tsai (School of Business, State Univ. of NY at New Paltz, New Paltz, NY)

Two challenges associated with analyzing acoustical data from wind farms are: 1) separating turbine sounds from environmental sounds and 2) classifying acoustical samples into different types of wind turbine noise based on acoustic characteristics. Machine-learning methods for classifying general environmental sounds have been developed using large human classified databases (e.g., YAMNet), but only a few studies have targeted classification of wind farm noise (WFN) specifically. Techniques for classifying wind farm noise have focused on identification of amplitude modulation (AM) using both traditional methods such as low frequency peak prominence (IOA method) and machine learning methods using both targeted AM acoustics features and more general deep acoustic features. To address these two challenges, we are developing a multi-echelon machine learning framework to identify and classify noise from wind farms using publicly available windfarm data and open-source software. The first echelon provides an automated method for identifying WFN samples that are free of environmental sounds. The second echelon uses machine learning to classify these wind farm noises according to the degree of AM, prominent tones, and other factors that might contribute to the human response and are incorporated in the metrics used or potentially used to assess compliance with regulations.

11:40

4aSP13. Using deep learning for recreating binaural audio. Will Sloan (System and Comput. Eng., Carleton Univ., 1125 Colonel By Dr., Ottawa, ON K1S 5B6, Canada, willsloan@cmail.carleton.ca) and Amir Laghai (Dept. of Electronics, Carleton Univ., Ottawa, ON, Canada)

In recent years, there has been an increase in research around generating spatialized audio using a mono audio signal. Methods like using neural networks which combine image segmentation with object location for adding back the spatial qualities are often developed. Instead, our project focuses on taking arbitrary mono input sound and an input angle, and outputting spatial stereo audio of the input sound with the directionality of the angle. This is different from current implementations as it is a simpler approach to what spatial audio generation is, and it allows for the use in the model in new areas. Using a binaural microphone and a custom-made anechoic chamber, 120 hours of labelled binaural audio was recorded for use in our model. The audio consists of frequency sweeps, pink noise, and phonetically balanced speech. Our method is to predict a complex short-time Fourier transform mask which will contain the phase and amplitude within it. The model is an autoencoder based on the U-NET model, which is applied to the mono input before being compared to the labelled data for training. With this simpler approach, we hope to make spatial audio more accessible for a variety of applications.

Session 4aUW

Underwater Acoustics and Acoustical Oceanography: Scintillation in Acoustic Propagation

Brian T. Hefner, Cochair
Applied Physics Laboratory, University of Washington, Applied Physics Laboratory, University of Washington, 1013 NE 40th Street, Seattle, WA 98105

Dajun Tang, Cochair
Applied Physics Lab, University of Washington, 1013 NE 40th Street, Seattle, WA 98105

Chair's Introduction—8:00

Invited Papers

8:05

4aUW1. On the feasibility of using short-range, high-frequency transmissions to characterize the vertical-spectra of small-scale internal waves and turbulence in a bottom boundary layer. John A. Colosi, Timothy F. Duda (Woods Hole Oceanographic Inst., Woods Hole, MA), Matthew Alford, Ying-Tsong Lin, and Gunnar Voet (Marine Physical Lab., Scripps Inst. of Oceanogr., La Jolla, CA)

Using weak fluctuation theory, we examine the detectability of small-scale internal waves and turbulence via short-range, high-frequency transmissions to a vertical array. For internal waves the theory needs modification since the ranges of interest are shorter than the internal wave correlation lengths. The turbulence spectra are characterized by thermal dissipation, χ, kinetic energy dissipation, ε, and outer scale $L_{;0}$ (largest turbulent eddies) and regions of this space can be explored with different ranges and acoustic frequencies. Very little is known of the internal-wave structure when wave breaking starts. As an example, we consider an experimental arrangement atop the Atlantis 2 seamount where there is a strong, tidally driven boundary layer approximately 100-m thick.

8:25

4aUW2. Polynomial chaos expansions for modelling the statistics of acoustic propagation in random waveguides. Kevin D. LePage (NATO STO Ctr. for Maritime Res. and Experimentation, Viale San Bartolomeo 400, La Spezia, SP 19126, Italy, kevin.lepage@cmre.nato.int)

The use of Polynomial Chaos expansions to model the transfer of the statistical properties of the sound speed in ocean waveguides to those of acoustic waves passing through them is described. A perturbational framework approximating the interaction of acoustic normal modes with fluctuations around a background sound speed is used, with the fluctuations being represented vertically by empirical orthogonal functions and horizontally by correlation functions. The Polynomial Chaos expansions are derived to second order in the uncorrelated random variable representing the sound speed fluctuations for the generalized phase of the complex modal amplitudes, and to third order for the modal amplitudes themselves. The ability of the second order polynomial chaos expansion for the generalized model phase to accurately model acoustic propagation statistics is closely coupled to the accuracy of the adiabatic approximation in light scattering regimes. The weights of the Polynomial Chaos expansions are obtained using both direct (theoretical) and indirect (least-squares fit) methods. Divergence between these two sets of weights increases with combinations of increasing strength of sound speed fluctuations, increasing frequency, or increasing range, indicating where a higher order expansion may be necessary. Modelling results for the first and second moments of the acoustic intensity and the Scintillation Index in random waveguides are presented and compared to Monte Carlo results.

8:45

4aUW3. Probabilistic acoustic predictions leveraging NESBA and ARGO measurements. Bill Stevens (Portland State Univ., San Diego, CA) and Martin Siderius (Portland State Univ., 1900 SW 4th Ave., Ste. 160, Portland, OR 97201, siderius@pdx.edu)

Ocean acoustic propagation models used for sonar performance prediction often rely on global or regional ocean modeling and data assimilation, or monthly climatologic averages, for estimating the 4D ocean temperature and salinity (T/S) field. Ocean models rely heavily on satellite-based ocean surface measurements, e.g., sea surface temperature (SST) and height (SSH), plus typically vastly less resolved in space and time vertical water column measurements derived using ship-based and/or ARGO float instruments. Systems also exist that create 3D synthetic T/S fields from satellite SST/SSH measurements, e.g., the Modular Ocean Data Assimilation System (MODAS) and the more recent Improved Synthetic Ocean Profile (ISOP) system. It is unclear, however, how well these synthetic T/S fields capture significant ocean ducting conditions. Similarly, climatologic averages combine historical vertical T/S profile data in spatial cells by month; but the averaging process tends to blur out important features. This talk will address the use of 2021 New England Shelf Break Acoustics (NESBA) experiment and ARGO vertical profile data for: (1) testing the degree to which ocean models or climatology

4a THU. AM

capture important acoustic features under a range of conditions; and (2) providing more realistic sonar performance prediction means and uncertainties. [Work supported by the Office of Naval Research].

9:05

4aUW4. Abstract withdrawn.

9:25

4aUW5. Analysis of mid-frequency sound intensity fluctuations on the Washington shelf. Dajun Tang (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, dtang@uw.edu), Brian T. Hefner, John B. Mickett, Ramsey R. Harcourt, Guangyu Xu, Eric I. Thorsos, and Kumar R. Prakash (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

A joint Oceanography/Acoustics experiment was conducted 15 July–13 August 2022 in Washington's shallow coastal shelf waters to investigate mid-frequency sound intensity fluctuations and the oceanographic mechanisms driving them. Acoustic pulses centered at 3.5 kHz and 6 kHz were transmitted up-shelf from a stationary source and recorded by receivers on two moorings at 10 km and 20 km ranges. A ship-towed profiler, the Shallow Water Integrated Mapping System (SWIMS), repeatedly sampled the ocean between the source and receiver moorings, providing sound speed measurements as a function of range, depth, and time. In addition, five oceanographic moorings distributed along the acoustic path took additional ocean data over time. A roughly 15- dB ping-to-ping TL variability and scintillation index of near unity are found at the 20 km range. The intensity data show strong correlation between pings separated by a few minutes, beyond which it is uncorrelated. After smoothing, they also show trends at roughly 12-hour intervals. Assisted by environmental data and numerical modeling, the oceanographic mechanisms for these observed fluctuations are investigated. Implications for signal processing in the presence of such intensity fluctuations are also discussed. (Work supported by Office of Naval Research.)

9:45–10:00 Break

10:00

4aUW6. Measurements of mid-frequency acoustic signal variability in the Western Barents Sea. Altan Turgut (Naval Res. Lab., Acoust. Div., Washington, DC 20375, altan.turgut@nrl.navy.mil), Jeffrey Schindall (Naval Res. Lab., Washington, DC), Ewa Jarosz, and Ewa Wijesekera (Naval Res. Lab, Stennis Space Ctr., MS)

Two Vertical Line Arrays (VLAs) and one source moorings were deployed to study along- and cross-shelf acoustic propagation in the Western Barents Sea during October 12–19, 2022. An ITC 2010 source was used to transmit Linear Frequency-Modulated (LFM) acoustic signals covering a frequency band of 0.7–4.2 kHz. Several oceanographic moorings were also deployed to measure time series of ocean currents and sound-speed profiles. Analysis of the acoustic data showed strong effects of tidal currents and surface roughness on mid-frequency acoustic propagation. Along- and cross-shelf acoustic propagation data showed semidiurnal travel-time fluctuations due to tidal currents. In addition, a 15 dB increase in Transmission Loss was observed during the storm events due to rough surface scattering. Finally, an assessment of the Polar Front and its effects on acoustic propagation (both travel-time and amplitude) are studied. Several data and simulated examples are provided to demonstrate the 3D effects of a Polar front and nonlinear internal waves on mid-frequency acoustic propagation. [Work supported by the ONR.]

Contributed Papers

10:20

4aUW7. The Acoustic Laboratory for Marine Applications (ALMA) applied to fluctuating environment analysis. Samuel Pinson (École Navale, Lanveoc, France), Victor Quilfen (Shom, Brest, France), Florent Le Courtois (DGA TN, Ave. de la Tour Royale, Toulon 83000, France, florent.lecourtois@gmail.com), Gaultier Real (CMRE, La Spezia, Italy), and Dominique Fattaccioli (DGA TN, Toulon, France)

The Acoustic Laboratory for Marine Applications (ALMA) has been used to address problems in underwater acoustics, such as sound propagation in fluctuating environments. In this work, data from the ALMA-2016 at-sea campaign are used to analyze the ocean fluctuation's influence on sound propagation in a shallow-water waveguide. The experiment took place in November 2016 on the continental shelf of the eastern coast of the island of Corsica. A source and a receiver array were 9.3 km apart in a nearly constant water depth of 100 m. A thermistor chain was moored near the source to monitor sound speed fluctuations. The source emitted a variety of signals from which the chirp (1–13 kHz) is used to extract the waveguide eigenrays. To do so, a time-domain beamforming is performed on the match-filtered received signals with an automatic detection of local maxima in the time of arrival/direction of arrival (TOA/DOA) domain. A 2 min acquisition period of more than 13 h duration shows significant fluctuations in eigenray TOAs/DOAs. Qualitative comparisons with synthetic signals obtained from simulations permit reproduction of the observed eigenray fluctuations without including range dependence of the sound-speed profile. In addition, the joint analysis of the probability density function of the normalized acoustic intensity and of the thermistor chain data highlights the time dependence of the received signal characteristics.

10:35

4aUW8. Mid-frequency convergence zone propagation. Franklin H. Akins (Scripps, UCSD, 562 Arenas St., La Jolla, CA 92037, fakins@ucsd.edu), William Hodgkiss (Scripps, UCSD, San Diego, CA), and William Kuperman (Scripps, UCSD, La Jolla, CA)

Convergence zone propagation has a cycle distance of order 50 km with most of the path spent below the thermocline. Calculations using Garrett-Munk statistics and rays computed with a smoothed background sound speed profile over this single cycle predict propagation in the partially saturated regime and an integration time of 500 seconds (time spent in a single output bin of a discrete Fourier transform). Parabolic equation simulations with synthetic fine structure and ocean dynamics validate the ray-based calculations. Data of narrowband transmissions from 1.5 to 7.5 kHz from a shallow towed source (150 m) to a shallow drifting array (150 m) depth at a range of 59 km demonstrate smoothly varying phase in a beam permitting coherent integrations of 3.5 minutes. The shorter integration time in data is consistent with non-uniform motion estimated from GPS.

10:50

4aUW9. Issues in mid-frequency sound intensity fluctuation on the Washington Shelf. Brian T. Hefner (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, hefner@apl.washington.edu), Dajun Tang, John B. Mickett, Ramsey R. Harcourt, Guangyu Xu, Eric I. Thorsos, and Kumar Prakash (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Strong mid-frequency sound intensity fluctuations were found on the Washington shelf during an experiment in the summer of 2022. The Scintillation Index (SI) of integrated sound energy on single channels at 20 km range was near unity for both 3.5 kHz and 6.0 kHz signals. Accompanying oceanographic measurements from moorings and a towed profiling system revealed sound field variability at multiple time and spatial scales caused by linear and nonlinear internal waves, internal tides, and coastal trapped waves. While the 2022 dataset provides valuable information for addressing issues on intensity fluctuation, additional experimental data are needed to understand the impacts on active and passive mid-frequency sonar systems. To that end, a second Washington Shelf experiment is being planned for the summer of 2025. This talk will discuss the planned measurements and the range of issues to be addressed. These include the frequency-dependence of SI over 1–10 kHz, broadband fading, vertical and horizontal coherence, and the impacts of variability on propagation in subsurface ducts. These measurements are important for investigating fundamental oceanographic causes of sound fluctuation as well as for the design of appropriate signal processing methods. (Work supported by the Office of Naval Research.)

11:05–11:25 Panel Discussion

Session 4pAAa

Architectural Acoustics, Noise, Computational Acoustics, Signal Processing in Acoustics, and Structural Acoustics and Vibration: Artificial Intelligence and Machine Learning for Built Environments

Semiha Yilmazer, Cochair
Interior Architecture and Environmental Design, Bilkent University, Bilkent University, Faculty of Art, Design and Architecture, Department of Interior Architecture and Environmental Design, Ankara, 06800, Turkey

Dick Botteldooren, Cochair
Information Technology, Ghent University, Technologiepark 126, Technologiepark 126, Gent, 9052, Belgium

Chair's Introduction—1:00

Invited Papers

1:05

4pAAa1. Using large language models in the analysis of urban sound environments. Dick Botteldooren (Information Technol., Ghent Univ., Technologiepark 126, Technologiepark 126, Gent 9052, Belgium, dick.botteldooren@ugent.be), Tiebe Vercoutter, and Yuanbo Hou (Information Technol., Ghent Univ., Ghent, Belgium)

In recent years, Large Language Models such as Generative Pre-trained Transformers (GPT) have revolutionized Artificial Intelligence (AI). Such models are extremely successful in imitating general intuitive knowledge. Such knowledge is applied by humans in assessing the urban sound environment in different ways. Firstly, it helps to create expectations on the sound environment based on general knowledge of the place. Secondly, it allows to assess the plausibility and consistency of the verbal description of a sound environment. Hence, we propose to combine a GPT with a sound event and scene recognition AI to (1) contrast the recognized sounds with expectations based on geographical information on traffic infrastructure and points of interest near the measurement location; (2) create verbal soundscape annotations including perceived liveliness, calmness, etc. Prompt engineering, that is pre-conditioning and asking precise questions to the GPT, requires some domain knowledge and precise definition of the objective. Results show that labeling of sound events is improved, and in particular labels that would not be used by a human can be excluded by including contextual knowledge on the location with GPT. They also show that a plausible soundscape description is obtained.

1:30

4pAAa2. A non-linear model approach for predicting soundscape perception of study areas. Semiha Yilmazer (Faculty of Art, Design and Architecture, Dept. of Interior Architecture and Environ. Design, Bilkent Univ., Ankara 06800, Turkey, semiha@bilkent.edu.tr) and Zekiye Şahin (Interior Architecture and Environ. Design, Bilkent Univ., Ankara, Turkey)

A large majority of the studies use linear regression-based models for soundscape modeling due to their easy applicability. Only a few studies have chosen non-linear structures, such as neural networks. Moreover, students' perceptions of soundscape quality in study areas have yet to be explored. We aimed to predict soundscape perception of study areas by applying neural network models. We also compared our results with models applying linear approaches. Perceptual dimensions were obtained by applying the Principal Component Analysis. In this study, we used the data from a two-phase experiment to find Turkish soundscapes' affective quality attributes. The experiment was conducted in a well-insulated, quiet laboratory room at Bilkent University, and most of the participants were university students and Bilkent University faculty members.

1:55

4pAAa3. Mining user reviews, crowdsourced noise measurements, and metadata to inform design thresholds for acoustic conditions in restaurants. Samuel H. Underwood (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, 1110 S 67th St., Omaha, NE 68182-0816, samuelunderwood@huskers.unl.edu) and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, Omaha, NE)

High levels of noise are a well-known source of occupant discomfort in restaurants. While much previous work has focused on assessing subjective perceptions of restaurant soundscapes through survey instruments, less attention has been given to mining online user reviews for keywords related to psychoacoustic predictors. In this study, a dataset of online user reviews for over 2,500 restaurants, bars, and coffee shops in New York City has been obtained and compared against crowd-sourced measurements of noise levels within the same establishments. Analyses of review scores and the occurrence rate of acoustically relevant keywords in review text have been conducted using machine learning techniques to suggest design thresholds for restaurants across different categorizations. Results from this work can be used by consultants and owners to design restaurant soundscapes which may be more favorably reviewed.

2:20

4pAAa4. Toward audio-based sensing for pedestrian detection. Yiwei Ding (Georgia Inst. of Technol., 840 McMillan St. NW, Atlanta, GA 30332, yding402@gatech.edu), Chaeyeon Han, Pavan Seshadri (Georgia Inst. of Technol., Atlanta, GA), Bon Woo Koo (Toronto Metropolitan Univ., Toronto, ON, Canada), Noah Posner, Subhro Guhathakurta (Georgia Inst. of Technol., Atlanta, GA), and Alexander Lerch (Music, Georgia Inst. of Technol., Atlanta, GA)

The detection and counting of pedestrians plays a central role for the design of smart cities. Although the use of cameras for this task has been shown to have high accuracy, they come at a high cost and are susceptible to challenges such as poor lighting, fog, and obstructed views. Our study investigates audio-based pedestrian detection, combining potentially low cost sensors with advanced machine learning based audio analysis algorithms. With an audio sensor installed along the walkway, machine learning algorithms can tell from the audio whether there is a pedestrian or not, or how far the pedestrian is from the sensor. Results show the general feasibility of audio-based pedestrian detection but fall short of reaching the accuracy levels of video-based detection.

2:45–3:00 Break

Contributed Papers

3:00

4pAAa5. Toward smart acoustic spaces: embedded machine learning for sound event detection and classification in the built environment. Tre DiPassio (Elec. and Comput. Eng., Univ. of Rochester, 402 Comput. Studies Bldg., 160 Trustee Rd., Rochester, NY 14627, tredipassio@rochester.edu), Michael C. Heilemann, Benjamin R. Thompson, Jenna Rutowski, and Mark Bocko (Elec. and Comput. Eng., Univ. of Rochester, Rochester, NY)

Edge computing integrated into the built environment will enable the development of smart acoustic spaces in which the flexural vibrations of the walls and screens that define a space may be employed to generate and monitor sound and serve as unobtrusive and convenient audio interfaces. Integration of data processing and neural decision making on compact, low-power embedded hardware-software systems will make it possible to mediate real-time, personalized acoustic interactions of a user with their environment without the need to connect to the cloud, thus providing privacy and security. In this presentation we show how embedded machine learning (ML) models combined with vibroacoustic control and monitoring of elastic panels may be employed to perform various tasks such as speech recognition, sound source localization, or the detection of acoustic signatures of specific events such as a fall or other health emergencies. The selection of vibroacoustic features, the associated signal processing requirements, and the computational resources utilized by the ML models for various acoustic tasks will be discussed. We also highlight how the distributed modal response of extended flat panels can simplify the sensing and signal processing requirements in such systems.

3:15

4pAAa6. Extracting and classifying psychoacoustic features from online restaurant reviews. Martin J. Mann (School of Sci. and Eng., Baldwin Wallace Univ., 275 Eastland Rd., Berea, OH 44017, mmann18@bw.edu), Samuel H. Underwood, and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, Omaha, NE)

Recent work has shown that poor acoustic conditions persist in many restaurants. Owners who receive negative reviews of their establishment's soundscape may struggle to interpret the subjective customer responses into actionable corrective measures. Therefore, further work is needed to taxonomize the acoustically relevant keywords and phrases that occur in user reviews. In this study, an open-source database of restaurant, bar, and coffee shop reviews from across the United States has been obtained. Sentiment analysis and keyword count are used to extract positive, negative, and neutral subjective reviews and subjective features related to acoustics. The resulting subjective features are then categorized, weighed, and linked to objective acoustic parameters using machine learning techniques. Results from this study suggest that owners and consultants may be able to utilize online customer reviews to monitor acoustical comfort and ascertain the nature of an acoustical problem, not only in restaurants, but in any business sector where customers submit online user reviews.

3:30

4pAAa7. A general overview of methods for generating room impulse responses. Mihai-Vlad Baran (Music Res., McGill Univ., 1199 Rue Bishops, 807, Montreal, QC H3G 0A7, Canada, vlad.mihai.baran@gmail.com), Richard King, and Wieslaw Woszczyk (Music Res., McGill Univ., Montreal, QC, Canada)

The utilization of room impulse responses has proven valuable for both the acoustic assessment of indoor environments and music production. Various techniques have been devised over time to capture these responses. Although algorithmic solutions have been in existence since the 1970 s for generating synthetic reverberation in real time, they continue to be computationally demanding and in general lack the accuracy in comparison to measured authentic Room Impulse Responses (RIR). In recent times, machine learning has found application in diverse fields, including acoustics, leading to the development of techniques for generating RIRs. This paper provides a general overview, of approaches and methods for generating RIRs, categorized into algorithmic and machine learning techniques, with a particular emphasis on the latter. Discussion covers the acoustical attributes of rooms relevant to perceptual testing and methodologies for comparing RIRs. An examination of disparities between captured and generated RIRs is included to better delineate the key acoustic properties characterizing a room. The paper is designed to offer a foundational literature base for those interested in RIR generation for music production purposes, with future work considerations also explored.

3:45

4pAAa8. Robotization of "*in situ*" acoustic measurements. Gabriel Leroux (MJM Acoust. Consultants, Longueuil, QC, Canada), Émile Gagnon, Lionel Birglen (Dept. of Mech. Eng., Polytechnique Montréal, Montréal, QC, Canada), and Nicolas Leveque (MJM Acoust. Consultants, 753 Ste-Hélène, Longueuil, QC J4K 3R5, Canada, nleveque@mjm.qc.ca)

Currently, "*in situ*" acoustic measurements require a high level of human involvement. Measurements are carried out by technicians or engineers, and most of the time, two people are required. MJM Acoustical Consultants Inc. has explored the possibility of using an autonomous and semi-autonomous robotic solution to assist in the measurement process. Developed in partnership with university Polytechnique Montreal, the research project consists of developing a software platform and hardware for a mobile robot capable of moving around, locating itself and performing measurements using a Type I sound level meter. The prototype generates its own local area network (LAN), making it possible to send commands through high-density partitions such as brick walls or concrete floors remotely from a laptop. Localization system such as a laser remote sensing system allows the robot to navigate around obstacles. Its low noise level when stationary makes it ideally suited to conduct common standard tests such as impact noise isolation, airborne sound attenuation between rooms and background noise level measurements. Through tests carried out on field sites, the solution developed constitutes an effective instrumentation to make acoustic measurements sessions more efficient with a reduced level of human operation.

4:00–4:30

Session 4pAAb

Architectural Acoustics and Structural Acoustics and Vibration: Building Envelope Sound Isolation II

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

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

Invited Papers

3:25

4pAAb1. Acoustically rated fenestration products. Casey Mahon (390 Industrial Blvd., Sauk Rapids, MN 56379, cmahon@stcloudwindow.com)

If noise is the enemy, then windows, doors, and other fenestration systems form a front-line of defense in preserving the peace. Starting with basic design elements and the material components of products manufactured to achieve an elevated level of attenuation, we will explore what it takes to reach even higher, especially as acoustic performance moves beyond simple punched openings to storefronts, curtain walls, and other specialty installations. We will see that certified test reports are something every bona fide product manufacturer should have for documentation. In summary, this presentation will explore a check list that acoustic professionals should consider for any project where noise attenuation is a primary performance objective.

3:45

4pAAb2. The effect of angle of incidence on exterior façade isolation. Wayland Dong (Paul S. Veneklasen Res. Foundation, 1711 Sixteenth St., Santa Monica, California, CA 90404, wdong@veneklasen.com) and John Lo Verde (Paul S. Veneklasen Res. Foundation, Santa Monica, CA)

A common acoustical task is the design and specification of the acoustical performance of the exterior façade system to protect building occupants from road and railroad noise. This involves calculating the outdoor-to-indoor noise reduction of the façade. The incident sound wave from a roadway will often approximate a plane wave from a specific angle of incidence, but the only data usually available is the diffuse field transmission loss of the assemblies as measured in the laboratory. While it is understood that there can be significant differences between the transmission loss at a given angle of incidence compared to the diffuse field value, a detailed and quantitative understanding of the effect is lacking. Measuring the transmission loss from plane waves at varying angles of incidence is difficult, and a combination of theoretical analysis and field measurements may allow a better understanding of the relationship and hence more accurate calculations of the interior noise level from transportation sources. The literature is reviewed, and data from field measurements is analyzed to begin to evaluate this effect.

4:05

4pAAb3. Case study of noise intrusion into glass facade residential building. Dana S. Hougland (Shen Milsom & Wilke, LLC, 1801 Wewatta, Fl. 11, Denver, CO 80202, dhougland@smwllc.com)

A combined residential and hotel building with a glass facade adjacent to a major transportation corridor provided challanges for the owner and designers in selecting an appropriate quantitative metric for evaluating the subjectively acceptable level of noise intrusion. Surveys at a series of buildings with similar proximities were evaluated to compare subjective impressions to quantitative projections.

4:25

4pAAb4. An overview of green roof acoustic performance. Shiang-I Juan (Taiwan Bldg. Technol. Ctr., National Taiwan Univ. of Sci. and Technol., 43 Keelung Rd. Sec. 4, TBTC, Taipei 106, Taiwan, S.Juan@mail.ntust.edu.tw) and Lucky S. Tsaih (Dept. of Architecture, National Taiwan Univ. of Sci. and Tech., Taipei, Taiwan)

This comprehensive review provides an in-depth examination of the acoustic performance of green roofs within urban and built environments over the past 15 years. Green roofs, characterized by their vegetative cover and substrate layers, offer a multifaceted solution to address challenges associated with urban noise pollution. The objective of this review is to synthesize existing literature, emphasizing key studies and methodologies employed to assess the noise reduction and sound isolation capabilities of green roofs. Factors influencing green roof acoustic performance, such as plant species selection in the plant layer, soil distribution, moisture content, and compaction level in the vegetation layer, as well as the overall roof design in terms of shapes and building configuration, are discussed in detail. Additionally, the impact of overall roof design, including shapes and building configuration, is explored. The relevant sound absorption coefficient and insertion loss values, drawn from the existing literature, will be reported. By systematically analyzing empirical findings, this overview offers valuable insights into the potential of green roofs as sustainable components for noise mitigation in building construction, contributing to the development of more resilient and harmonious urban landscapes.

4:45

4pAAb5. Developing and validating analytical tools to predict the aero-acoustic performance of bespoke noise mitigation solutions. Viken Koukounian (Parklane Mech. Acoust., 3-1050 Pachino Court, Burlington, ON L7L 6B9, Canada, viken@parklanemechanical.com)

The characterization of empirical knowledge from observations is, generally, referred to as the Scientific Method. Most engineering best practices rely on ‘observations’ (e.g., laboratory testing) to make informed decisions for future projects and/or applications. While this approach is still considered to be the ‘state of the art’ in most fields of acoustics, it is especially relevant in the studies of architectural acoustics and environmental noise. Since it is not possible to test every permutation of product (and/or its configurations) and environmental conditions, the results of only common geometries and parameters are reported in databases for use by professionals. However, the described framework is rigid and prohibits true optimization of a solution which should consider the solution’s performance requirements, as well as the various constraints (site, project, multi-physics). A more sophisticated approach is possible, where analytical tools are developed to predict the multi-physics performance of products (such as silencers, plenum-silencers, louvers and enclosures). Numerical methods are relied on to validate these new tools against empirical data. The presented work demonstrates the optimization of a product’s aero-acoustic performance according to multi-variable constraints.

5:00

4pAAb6. Residential outdoor-to-indoor acoustic conditions: Method and pilot data. Iara B. Cunha (National Res. Council Canada, 1200 Montreal Rd., M-27, Ottawa, ON K1A 0R6, Canada, iara.batistadacunha@nrc.ca), Jennifer A. Veitch, Ashley Nixon, Markus Mueller-Trapet, Sabrina Skoda, and Jeffrey Mahn (National Res. Council Canada, Ottawa, ON, Canada)

Health and well-being of people is directly influenced by the indoor environmental quality of the buildings they occupy. Noise exposure, for example, is not only a cause of annoyance but it is established as a risk factor for the development of cardiovascular diseases and it is a cause of sleep disturbance. As part of a broader research project focused on establishing guidance for suitable interior conditions for adults as they age, the National Research Council of Canada is developing a field study to measure, among other environmental conditions, outdoor-to-indoor noise conditions at homes. Objective parameters such as sound pressure levels will be monitored for a period of time and compared to residents’ health and well-being assessments through sensors and surveys. The methodology and instrumentation used for the acoustic measurements will be presented as well as the early results from pilot tests. The future outcomes of this study will contribute to guidance for both new-build and retrofit scenarios of dwellings for aging in place.

5:15

4pAAb7. Effects of time varying aircraft noise on indoor speech interference. Adam Collins (400 – 2100 Derry Rd. West, Mississauga, ON L5N0B3, Canada, Adam.Collins@Stantec.com)

Aircraft noise can create noticeable speech interference inside buildings adjacent to airports. Outdoor aircraft take off noise measurements were completed and used to study time varying effects on indoor environments. Preliminary results suggest changing levels of speech interference over the duration of aircraft take off events.

Session 4pBAa

Biomedical Acoustics and Physical Acoustics: Droplets Strike Back II

Virginie Papadopoulou, Cochair
Biomedical Engineering, The University of North Carolina at Chapel Hill, 116 Manning Drive, 9004 Mary Ellen Jones Building, CB 7575, Chapel Hill, NC 27599-7575

Mario L. Fabiilli, Cochair
University of Michigan, 1301 Catherine Street, 3226A Med Sci I, Ann Arbor, MI 48109

Kevin J. Haworth, Cochair
Department of Internal Medicine, University of Cincinnati, 231 Albert Sabin Way, CVC 3939, Cincinnati, OH 45267

Invited Papers

1:00

4pBAa1. Enhancing cancer treatment through solid tumor fractionation with nanodroplet-mediated histotripsy and immunotherapy. Bar Glickstein (Bio Medical Eng., Tel Aviv Univ., Tel Aviv, Israel) and Tali Ilovitsh (Bio Medical Eng., Tel Aviv Univ., Ramat Aviv, Tel Aviv, Israel, Ilovitsh@tauex.tau.ac.il)

Checkpoint inhibition holds great promise in enhancing the body's immune response against cancer for durable and widespread anti-tumor effects. However, its efficacy in solid tumor therapy faces challenges such as limited immune cell infiltration and an immunosuppressive tumor microenvironment. This study introduces a cancer therapeutic platform, combining nanodroplet (ND)-mediated low-frequency histotripsy with anti-PD1 (aPD1) checkpoint inhibition. NDs, with their ability to penetrate capillaries and extravasate into tumor tissues, present a promising feature for noninvasive tumor treatments. The proposed method involves a two-step approach for low-energy ND-mediated histotripsy. First, NDs are activated volumetrically using a rotating imaging probe into cavitating gas bubbles. Subsequent low-frequency ultrasound implodes vaporized NDs, inducing tumor fractionation, tissue necrosis, and enhanced immune-cell infiltration, creating a T-cell–inflamed tumor. The goal is to amplify immunotherapy, advancing cancer treatment by targeting both the tumor and its microenvironment. Demonstrated in a breast cancer mouse model, the two-step approach showed significant lesions and tumor debulking. Combining aPD1 checkpoint inhibition further enhanced immune-cell infiltration and reduced tumor growth. Our findings demonstrate the potential of NDs with low-frequency ultrasound and immunotherapy for efficient noninvasive tumor treatment.

1:25

4pBAa2. Direct emulsification of liquid perfluorobutane: an improved method to formulate superheated perfluorocarbon nanodroplets. Adam Woodward, Robert Mattrey (Radiology, UTSouthwestern Medical Ctr., Dallas, TX), and Caroline de Gracia Lux (Radiology, UTSouthwestern Medical Ctr., 5323 Harry Hines, Dallas, TX 75390-8514, Caroline.Lux@UTSouthwestern.edu)

This talk will describe a simple and robust method to produce perfluorobutane (PFB) nanodroplets by direct sonication, and will compare the properties of the resultant emulsions with those obtained with the current advocated microbubble condensation method. We found that nearly 100% of particles produced by direct emulsification of PFB liquid at low temperature are liquid PFB-filled, whereas the emulsion produced by microbubble condensation if the microbubbles are not washed, contains 2000 times more non-PFB filled than PFB-filled particles. Consequently, when such suspensions are used to target receptors, the abundance of non-PFB particles will act as competitive inhibitors, and when used to extravasate to reach extravascular targets, the lower count of PFB-filled particles will require larger doses decreasing efficacy and increasing side effects. Adopting this direct formulation could be a game changer for all applications when experimental outcome is dependent on nanodroplet concentration, stability, purity and size. As a result, this method should accelerate the translation of these ultrasound activatable nanodroplets to both image and potentially treat diseased tissues.

1:50

4pBAa3. Ultrasound responsive multi-layered emulsions for drug delivery. Aaqib H. Khan (Chemical Eng., Indian Inst. of Technol. Gandhinagar, IIT Gandhinagar, Palaj Campus, Simkheda, Gandhinagar, Gujarat 382355, India, aaqib.khan@iitgn.ac.in), Sapna Bisht, Nishita Mistry, Karla Patricia Mercado-Shekhar (Biological Eng., Indian Inst. of Technol. Gandhinagar, Gandhinagar, Gujarat, India), and Sameer V. Dalvi (Chemical Eng., Indian Inst. of Technol. Gandhinagar, Gnadhinagar, Gujarat, India)

Vaporizable double emulsions, characterized by a central aqueous core, have demonstrated effectiveness in encapsulating hydrophilic drugs. This study aims to investigate the potential of incorporating an additional oil-layer in the double emulsions to encapsulate hydrophobic drugs. Vaporizable multi-layered emulsions were produced in three steps using perfluoropentane (PFP), phosphate-buffered saline (PBS), and sunflower oil. Curcumin, a natural anti-inflammatory drug, was dispersed in the oil phase. Krytox, polyglycerol polyricinoleate, and bovine serum albumin (BSA) were used as surfactants. PFP was sonicated in PBS (1:6) for 1 minute to create emulsion-1. Subsequently, emulsion-1 (1:4) was homogenized in oil to make emulsion-2. Emulsion-2 was homogenized in BSA (1:4) to yield emulsion-3 at 8000 rpm for 30 seconds. The vaporization pressure threshold was determined using 2 MHz focused ultrasound with a single-element transducer (f/# of 1.27, 0.5% duty cycle). B-mode imaging was conducted using a Verasonics Vantage 128 system with an L11-5v array to determine the droplet vaporization threshold, which was found to be 6.7 MPa. Curcumin-loading (0.87 ± 0.1 mg) was significantly higher in the multi-layered emulsions than in single-layered BSA-shelled microbubbles (0.019 ± 0.004 mg) ($p < 0.00001$), indicating that multi-layered emulsions exhibit higher drug loading capacity.

2:05

4pBAa4. The effect of decafluorobutane lipid nanoemulsion concentration on oxygen scavenging via acoustic droplet vaporization. Nour Al Rifai (Internal Medicine, Univ. of Cincinnati, 231 Albert Sabin way, Cardiovascular Ctr. 3950, Cincinnati, OH 45267-0586, alrifang@ucmail.uc.edu), Kateryna Stone (Internal Medicine, Univ. of Cincinnati, Cincinnati, OH), Diviyashree Kasiviswanathan (Medical Sci. Baccalaureate Program, Univ. of Cincinnati, Cincinnati, OH), and Kevin J. Haworth (Internal Medicine, Univ. of Cincinnati, Cincinnati, OH)

Acoustic droplet vaporization (ADV) is the phase transition of liquid droplets into gas microbubbles via ultrasound. Oxygen diffuses from the surrounding fluid into the microbubbles. This study determined the efficiency of oxygen scavenging using a lipid decafluorobutane (DFB) nanoemulsion. DFB nanoemulsions were prepared using high-shear pressure homogenization. Nanoemulsion size distributions and concentrations were measured ($n = 10$) using a Beckman Coulter Multisizer 4. Oxygen scavenging in 95% oxygenated water was measured in a flow phantom prior to introducing DFB nanoemulsion at concentrations of 0.05×10^{-4}, 0.25×10^{-4}, 0.5×10^{-4}, 2.5×10^{-4}, and 5×10^{-4} mL/mL and with ADV nucleated by an EkoSonic catheter driven with 47 W electrical power. The DFB nanoemulsion had a modal diameter of 920 ± 60 nm, polydispersity index (PDI) of 0.11 ± 0.03, and concentration of $3.11 \times 10–2 \pm 0.01 \times 10–2$ mL/mL. No statistical significant differences was detected in droplet size distribution metrics for at least 5 h at room temperature and 23 days at 4C. The baseline oxygen partial pressure (pO2) of the water was 553 ± 8 mmHg for all experiments. Peri-ADV pO2 dropped to 505 ± 16, 398 ± 44, 348 ± 11, 231 ± 27, 241 ± 7 mmHg for increasing concentration. A significant difference in pO2 was seen between thelowest and highest concentration ($p = 0.0097$). The computed ADV transition efficiency was highest at 0.25×10^{-4} mL/mL.

2:20

4pBAa5. Characterization of acoustic droplet vaporization in tissue-mimicking, fibrin-based hydrogels for microrheology applications. Anuj Kaushik (Radiology, Univ. of Michigan, Basic Radiological Sci., Medical Sci. I, A-wing, 6446B Ste. 1301 Catherine St., Ann Arbor, MI 48109-2026, ankaushi@umich.edu), Bachir A. Abeid, Jon B. Estrada (Dept. of Mech. Eng., Univ. of Michigan, Ann Arbor, MI), Mario L. Fabiilli, and Mitra Aliabouzar (Radiology, Univ. of Michigan, Ann Arbor, MI)

Ultrasound-induced vaporization of phase-shift droplets (PSDs) into microbubbles, termed acoustic droplet vaporization (ADV), has expanded biomedical applications. Here, we explored the potential of ADV as a high-resolution, on-demand microrheometer using theoretical and experimental methodologies. This approach could be used to characterize tissue elasticity at spatial resolutions unrealizable with conventional *in situ* techniques, thus assisting with identification of underlying pathologies. For theoretical studies, bubble dynamics were combined with appropriate material constitutive models accounting for viscoelasticity of the surrounding tissue. Experiments were conducted on submicron- and micron-sized, perfluoropentane PSDs embedded in fibrin-based hydrogels with varying shear elastic moduli (0.2 kPa–11 kPa) to mimic soft tissues. Experimental studies included optical high-speed imaging and acoustic interrogations to record radial dynamics and acoustic emissions of the ADV-bubbles, respectively. Radial dynamics and the acoustic emissions of microbubbles produced by ADV depended significantly on fibrin elasticity. For example, an increase in fibrin elastic modulus from ~0.8 kPa to ~6 kPa reduced the maximum expansion radius of the generated ADV-bubbles by 50%. This increase in elasticity significantly impacted both linear (e.g., fundamental) and nonlinear (e.g., subharmonic) acoustic responses of the ADV-bubbles, by up to 15 dB. These findings open doors to a novel ADV-assisted tissue characterization technique.

2:35–3:00 Break

Invited Papers

3:00

4pBAa6. Perfluorocarbon nanodroplets for ultrasound imaging and drug delivery. Geoffrey P. Luke (Thayer School of Eng., Dartmouth College, 15 Thayer Dr., Hanover, NH 03755, geoffrey.p.luke@dartmouth.edu)

Perfluorocarbon nanodroplets can be triggered to undergo a liquid-to-gas phase transition with an externally applied acoustic pulse. The resulting gaseous bubbles provide strong ultrasound contrast, enabling visualization of their distribution. This talk will focus on recent advances in imaging methods that enable multiplex and molecular imaging using the nanodroplets. In addition, a double-emulsion nanodroplet will be presented that simultaneously encapsulates hydrophobic and hydrophilic drugs. These dual-drug loaded nanodroplets can be used for on-demand delivery of combination chemotherapy and/or immunotherapy to a site of interest.

3:25

4pBAa7. Acoustic cluster therapy: An overview of research in cancer treatment. Jeffrey C. Bamber (Joint Dept. of Phys., Inst. of Cancer Res. and Royal Marsden NHS Foundation Trust, 15 Cotswold Rd., Sutton, London SM2 5NG, United Kingdom, jeff.bamber@icr.ac.uk), Nigel Bush (Joint Dept. of Phys., Inst. of Cancer Res. and Royal Marsden NHS Foundation Trust, London, United Kingdom), Udai Banerji (Drug Development Unit, Inst. of Cancer Res. and Royal Marsden NHS Foundation Trust, London, United Kingdom), Nina Tunariu (Radiology, Royal Marsden NHS Foundation Trust, London, United Kingdom), Mark O'Leary (Joint Dept. of Phys., Inst. of Cancer Res. and Royal Marsden NHS Foundation Trust, London, United Kingdom), Chanthirika Ragulan (Div. of Molecular Pathol., Inst. of Cancer Res., London, United Kingdom), Per Sontum (Oslo, Norway), Catharina de Lange Davies (NTNU, Trondheim, Norway), Melina Mühlenpfordt (Exact Therapeutics, Oslo, Norway), Erik Wennerberg (Div. of Radiotherapy and Imaging, Inst. of Cancer Res., London, United Kingdom), Anguraj Sadanandam (Div. of Molecular Pathol., Inst. of Cancer Res., London, United Kingdom), Alan Melcher (Div. of Radiotherapy and Imaging, Inst. of Cancer Res., London, United Kingdom), Amir Snapir, Andrew Healey, and Svein Kvåle (Exact Therapeutics, Oslo, Norway)

This presentation provides an overview of research on acoustic cluster therapy (ACT) in cancer treatment. ACT is a promising approach to enhancing the delivery of drugs to cancer cells that involves the use of intravenously injected clusters of microbubbles and microdroplets, which are activated by diagnostic ultrasound at the tumor site. Upon activation, the clusters expand to form bubbles of 20–40 μm diameter. When modulated by low-pressure (MI $\sim$ 0.2) low-frequency ($\sim$500 kHz) ultrasound while they sit in the tumor capillaries, these bubbles generate biomechanical effects that improve drug extravasation and interstitial permeation. They dissolve after 5–10 min. The technique has been used to temporarily open the blood brain barrier, improve the effectiveness of various chemotherapeutics in a range of tumor models, and provide ACT-imaging biomarkers that predict therapeutic outcome. It has exhibited no adverse effects to-date in a phase I/II clinical trial and improved clinical response of colorectal cancer liver metastases to chemotherapy. Ongoing research is investigating the potential synergistic effects of ACT with immunotherapy.

Contributed Papers

3:50

4pBAa8. Optimization of ultrasound contrast agent and treatment duration for drug delivery to methicillin-resistant Staphylococcus aureus diabetic wound biofilms in mice. Kelly VanTreeck (Biomedical Eng., UNC Chapel Hill, 116 Manning Dr., Chapel Hill, NC 27514, kevantre@unc.edu), Jamie Liu, Ashelyn Sidders, Kuan-Yi Lu, Amanda Velez, Phillip Durham, Duyen Bui, Michelle Angeles-Solano (Univ. of Chapel Hill, Chapel Hill, NC), Paul A. Dayton (Biomedical Eng., UNC Chapel Hill, Chapel Hill, NC), Sarah Rowe (Univ. of Chapel Hill, Chapel Hill, NC), and Virginie Papadopoulou (Biomedical Eng., UNC Chapel Hill, Chapel Hill, NC)

Chronic wounds are frequently infected with bacterial biofilms which pose a barrier to drug diffusion, drug uptake, and facilitate drug tolerance, thereby prolonging the healing process in chronic wounds and increasing the likelihood of relapse. A major contributor to relapse is persister cells, a subset resilient to high antibiotic doses. Previous work by our group showcased significantly improved gentamicin efficacy in a diabetic murine wound model using ultrasound, nanodroplets, and an anti-persister drug. Here, we aim to optimize nanodroplet formulation for persister cell targeting and treatment duration for clinical translation. In a methicillin-resistant Staphylococcus aureus (MRSA) diabetic wound model in SKH-1 hairless mice ($n = 5$), wounds were treated twice daily with gentamicin, ultrasound, and nanodroplets containing oxygen or palmitoleic acid (PA). These components aim to increase oxygenation and enhance persister uptake, respectively. Additionally, treatments with gentamicin, PA nanodroplets, and ultrasound for 2, 5, and 10 minutes were investigated ($n = 3$). Preliminary findings indicate PA most effectively potentiates antibiotic activity, 10-minute treatments improve efficacy by 1-log compared to 5-minutes, and 2-minute treatments achieve comparable efficacy to 5-minutes. These initial findings suggest a promising strategy for enhancing antibiotic efficacy and targeting persister cells in biofilms, with consideration for resource-efficient treatment times for clinical application.

4:05

4pBAa9. Localized compaction of collagen hydrogels using acoustic droplet vaporization promotes osteogenic differentiation of mesenchymal stem cells. Somnath Maji (Radiology, Univ. of Michigan, 1301 Catherine St., Med Sci. I, Rm. 6428, Ann Arbor, MI 48109, somnath.2812@gmail.com), Mitra Aliabouzar, Carole Quesada, Aidan Macpherson, Man Zhang (Radiology, Univ. of Michigan, Ann Arbor, MI), Brendon M. Baker, Renny Franceschi (Biological Chemistry, Univ. of Michigan, Ann Arbor, MI), and Mario L. Fabiilli (Univ. of Michigan, Ann Arbor, MI)

The lineage specification of mesenchymal stem cells (MSCs) is dependent on matrix stiffness, with osteogenesis promoted on or within stiff microenvironments. Unlike native extracellular matrix, conventional hydrogels have relatively static and uniform mechanical properties, thereby limiting the study of how spatiotemporally controlled mechanical cues impact MSC behavior. Here, we spatiotemporally control MSC differentiation in 3D hydrogels by actively modulating matrix stiffness using acoustic droplet vaporization (ADV). An acoustically responsive scaffold (ARS) was generated by co-encapsulating MSCs and perfluorohexane-based emulsion (13 μm) within type I collagen. ADV was used to generate bubbles within the ARS. Bubble diameter and the width of the compacted matrix region increased over time following ADV. Atomic force microscopy measurements demonstrated that the hydrogel region proximal to the bubble was significantly stiffer than the distal regions. Significantly greater levels of Runx2 and osteocalcin expression were observed in MSCs proximal to bubbles compared to distal on days 7 and 14. Higher alkaline phosphatase activity also validated the above findings. The significant upregulation of these

osteogenic biomarkers suggests ADV-induced, mechanical changes in ARSs were sufficient to enhance osteogenic differentiation of MSCs. This approach is a significant step towards controlling the 3D differentiation of MSCs in a spatially localized, non-invasive, and on-demand manner.

4:20

4pBAa10. Simultaneous plane-wave and ultra high-speed imaging of acoustic droplet vaporization. Laura Taylor (Imperial College London, London, United Kingdom), Qiang Wu (Univ. of Oxford, Oxford, United Kingdom), Kai Riemer (Imperial College London, London, United Kingdom), Luca Bau (Univ. of Oxford, Oxford, United Kingdom), Christopher Dunsby, Meng-Xing Tang (Imperial College London, London, United Kingdom), and Eleanor P. Stride (Univ. of Oxford, Inst. of Biomedical Eng., Oxford OX3 7DQ, United Kingdom, eleanor.stride@eng.ox.ac.uk)

Super-heated sub-micrometre perfluorocarbon droplets have been shown to offer advantages over microbubbles for ultrasound localization microscopy (ULM). These include better penetration of the microcirculation and enabling higher frame rates using methods such as acoustic wave sparsely activated localization microscopy (AWSALM). The variability of droplet behaviour has however been found to be substantially higher than that of microbubbles and this poses challenges for ULM image construction. The aim of this study was to use simultaneous high speed optical and acoustic imaging to capture the droplet vaporisation process and subsequent bubble dynamics. Droplets composed of octofluoropropane and decafluorobutane were injected into a polyethylene tube submerged in a 37 °C water bath and located at the focus of an ultrasound probe and microscope objective. Optical images were captured at 10 Mfps of the droplet response to ultrasound pulses of varying lengths and amplitude. The results indicate that the number of activated droplets is impacted by increasing either the ultrasound pressure or number of half-cycles as well as by the preceding activation ultrasound pulses; and that different droplet subpopulations can be activated by varying pulse parameters. These data indicate the potential for optimising droplet activation of ULM applications.

THURSDAY AFTERNOON, 16 MAY 2024 — ROOM 212, 1:00 P.M. TO 5:05 P.M.

Session 4pBAb

Biomedical Acoustics, Structural Acoustics and Vibration, Computational Acoustics, Acoustical Oceanography, and Physical Acoustics: Wave Propagation in Complex Media: From Theory to Applications II

Pierre Bélanger, Cochair
1100 Rue Notre Dame O, Montréal, H3C1K3, Canada

Guillaume Haiat, Cochair
Multiscale modeling and simulation laboratory, CNRS, Laboratoire MSMS, Faculté des Sciences, UPEC, 61 avenue du gal de Gaulle, Creteil 94010, France

Contributed Papers

1:00

4pBAb1. Heterogeneous ultrasonic wave properties in leg cortical bones of Thoroughbreds. Shuta Kodama (Lab. of Ultrasonic Electronics, Faculty of Sci. and Eng., Doshisha Univ., 1-3 Miyakodani, Tatara, Kyotanabe, Kyoto 610-0321, Japan, ctwj0327@mail4.doshisha.ac.jp), Taisei Tsubata (Doshisha Univ., Kyotanabe City, Kyoto Prefecture, Japan), Norihisa Tamura, Hiroshi Mita (JRA Equine Res. Inst., Shimotsuke, Tochigi, Japan), Ko Chiba (Dept. of Orthopedic Surgery, Graduate School of Biomedical Sci., Nagasaki Univ., Sakamoto, Nagasaki, Japan), and Mami Matsukawa (Doshisha Univ., Kyotanabe, Kyoto, Japan)

Safe and inexpensive diagnosis of leg bone disease in the field is very important for Thoroughbreds and large animals, because large animals are difficult to transport, and prompt diagnosis is required. The quantitative ultrasound (QUS) is attracting attention for the evaluation of leg bones because it can give us information of elasticity, which is related to the bone quality. For the development of a QUS system of animal leg bones, understanding of characteristic ultrasonic wave properties (velocity and attenuation) is a key issue. Therefore, wave properties in Thoroughbreds' leg bones were obtained experimentally using an ultrasonic pulse technique. Precise measurements showed clear heterogeneity of ultrasonic wave properties due to the location. Opposed to the other large animals like bovines, high wave velocities were found in the medial and lateral parts, which possibly tells the effects of hard training in the fields. From a clinical perspective, the medial and lateral parts are less prone to diseases such as periostitis. The attenuation was greater in the posterior part, especially in the porous area near the trabecular bone. We also found that attenuation depended on bone matrix. Further investigation is necessary to discuss on the heterogeneity of wave properties in detail.

1:15

4pBAb2. Assessment of cortical bone phantom properties using ultrasonic guided waves transduced with a multi-element transducer. Aubin A. Chaboty (PULÉTS, École de Technologie Supérieure, 1100 Rue Notre Dame O, Montréal, QC H3C1K3, Canada, aubin.chaboty.1@ens.etsmtl.ca), Vu-Hieu Nguyen (Laboratoire Modélisation et Simulation Multiechelle, Univeristé Paris-Est Créteil, Créteil, France), Guillaume Haiat (Laboratoire Modélisation et Simulation Multiechelle, Ctr. National de la Recherche Scientifique, Créteil, France), and Pierre Bélanger (PULÉTS, École de Technologie Supérieure, Montréal, QC, Canada)

The past decade has seen extensive exploration of alternative methods for the early diagnosis of osteoporosis through the assessment of the the bone quality. Previous research on axial transmission of ultrasonic guided waves demonstrated their sensitivity to the intrinsic properties of elongated cortical bones. This study highlights the capacity of low-frequency guided waves to ascertain bone properties through the inversion of dispersion curves. The proposed inversion scheme relies on dispersion curves simulated using the semi-analytical iso-geometric analysis (SAIGA) method. The model incorporates the excitability of ultrasonic guided wave modes to ensure that the inversion uses both the dispersive trajectories and amplitudes of the modes. Two models were examined: (1) a cortical bone phantom plate covered with a soft tissue mimicking material and (2) a quasi-cylinder cortical bone phantom surrounded by a soft tissue mimicking material. A proprietary axial transmission multielement ultrasonic transducer, designed for exciting guided waves under 500 kHz, was employed to capture experimental dispersion curves on phantoms through the 2D-FFT. The mechanical properties of the bone phantoms were inferred by minimizing disparities between experimental and simulated dispersion curves. Inverse properties exhibited an error of less than 4% compared to reference values. The axial transmission probe demonstrated its proficiency in accurately measuring modes propagating inside the cortical layer.

1:30

4pBAb3. Using an instrumented hammer to detect the rupture of the pterygoid bone plate: an animal study. Manon Bas dit Nugues (MSME, CNRS UMR 8208, CNRS, 61 Ave. du Général de Gaulle, Créteil 94000, France, manon.bas-dit-nugues@u-pec.fr), Giuseppe Rosi (Université Paris-Est Créteil, CNRS, MSME ; Université Gustave Eiffel, MSME, Creteil, France), Charles-Henri Flouzat-Lachaniette (INSERM U955, IMRB Université Paris-Est ; APHP, Hôpital Henri-Mondor, Service de Chirurgie Orthopédique et Traumatologique, Créteil, France), Roman H. Khonsari (APHP, Hôpital Necker-Enfants Malades, Service de Chirurgie maxillo-faciale et chirurgie plastique, Laboratoire 'Forme et Croissance du Crâne', Paris, France), and Guillaume Haïat (MSME, CNRS UMR 8208, CNRS, Créteil, France)

Craniofacial osteotomies involving pterygomaxillary disjunction are common procedures in maxillofacial surgery to detach the pterygoid plates from the palatal bones. Surgeons still rely on their proprioception to determine when to stop impacting the osteotome, which is important to avoid complications such as dental damage and bleeding. Our group has developed a technique consisting in using an instrumented hammer that can provide information on the mechanical properties of the tissue located around the osteotome tip. The aim of this study is to determine whether a mallet instrumented with a force sensor can be used to predict the crossing of the osteotome through the pterygomaxillary plate. 31 osteotomies were carried out in 16 lamb skulls. For each impact, the force signal obtained was analyzed using a dedicated signal processing technique. A prediction algorithm based on an SVM classifier was applied to the database. We showed that the device can detect the crossing of the osteotome, sometimes before its occurrence. The prediction accuracy of the device was 94.7%. The method seemed to be sensitive to the thickness of the plate and to crack apparition and propagation. These results pave the way for the development of a per-operative decision support system in maxillofacial surgery.

Invited Paper

1:45

4pBAb4. Abstract withdrawn.

Contributed Papers

2:05

4pBAb5. k-Wave II: Introduction, progress, and road map. Bradley Treeby (Dept. of Medical Phys. and Biomedical Eng., University College London, London WC1E 6BT, United Kingdom, b.treeby@ucl.ac.uk), Antonio Stanziola, David Stansby, Devaraj Gopinathan (Medical Phys. and Biomedical Eng., Univ. College London, London, United Kingdom), Jiri Jaros (Brno Univ. of Technol., Brno, Czechia), and Ben Cox (Medical Phys. and Biomedical Eng., Univ. College London, London, United Kingdom)

k-Wave is an open-source toolbox for simulating the propagation of acoustic and ultrasound waves in complex and tissue-realistic media that is widely used across academia and industry. In this talk, we present an introduction to k-Wave II. k-Wave-II is a major re-write of the original k-Wave toolbox developed with the following aims: (1) re-engineering the code base to leverage object orientated programming, making it simpler to use for model inversions and coupled physics problems, (2) extending the algorithms to facilitate general boundary conditions on arbitrary surfaces, and to increase performance for narrow-band simulations, (3) improving the development and release process to incorporate good practice and advance long-term sustainability, and (4) improving training, user engagement, and support. We will introduce the new toolbox, discuss current progress, and a provide a road map for future development.

2:20

4pBAb6. Wavefield boundary interactions in a multi-domain environment—Experimental results. Michelle E. Swearingen (Construction Eng. Res. Lab., US Army ERDC, PO Box 9005, Champaign, IL 61826, michelle.e.swearingen@usace.army.mil), Oliver-Denzil Taylor (ECS Southeast, LLC, Baton Rouge, LA), Alanna Lester (Cold Regions Res. and Eng. Lab., US Army ERDC, Lyme, NH), Abigail Stehno (Coastal Hydraulics Lab., US Army ERDC, Vicksburg, MS), Michael J. White (Construction Eng. Res. Lab., US Army ERDC, Champaign, IL), Christa Woodley (Environ. Lab., US Army ERDC, Vicksburg, MS), and Mihan McKenna (US Army ERDC, Vicksburg, MS)

Acoustic and seismic signals may be generated and received in land, air, and water domains within a multi-domain environment such as a littoral (nearshore) zone. As the energy moves between domains, the waves undergo complex transformations at the media boundaries. Separate consideration of each physical domain (land, air, water) leads to an incomplete understanding of the full cross-domain wavefield, leading to significant challenges in interpreting signals and creation of accurate models that realistically couple multiple domains. To address this knowledge gap, a team at the US Army Engineer Research and Development Center constructed a simplified, uniform levee to enable the measurement of signals that had

passed through undisturbed boundaries. Sensor types were medium-specific, with microphones in the air, hydrophones in the water, and accelerometers in the soil. Using electric bridge wire detonators (EBW's), impulsive signals were generated in each medium type and measured on sensors in all three media. This presentation begins with an overview of the experimental design, discusses the medium-specific measurement results, and concludes with a discussion of joint multi-domain data analysis results. Approved for public release: distribution is unlimited.

2:35

4pBAb7. Rapid computation of steady state acoustic fields in heterogeneous and nonlinear media using the Acoustic Field Propagator. Ben Cox (Univ. College London, University College London, Gower St., London WC1E 6BT, United Kingdom, b.cox@ucl.ac.uk), Ratan Saha (Indian Inst. of Information Technol. Allahabad, Allahabad, India), Antonio Stanziola, and Bradley Treeby (Univ. College London, London, United Kingdom)

The Acoustic Field Propagator is a method for rapidly computing steady state complex acoustic fields in homogeneous media from spatially arbitrary single frequency sources. Here, an extension of the Acoustic Field Propagator to the computation of steady state fields in arbitrary heterogeneous, absorbing and nonlinear media, using nested iterations of the Convergent Born Series and a multi-frequency Helmholtz equation expansion, will be described.

2:50–3:05 Break

3:05

4pBAb8. A static memory method for modelling time-fractional power law absorption. Matthew J. King (Dept. of Medical Phys. and Biomedical Eng., Univ. College London, Malet Pl. Eng. Bldg., Gower St., London WC1E 6BT, United Kingdom, matthew.king@ucl.ac.uk), Timon S. Gutleb (Dept. of Mathematics, Univ. of BC, London, United Kingdom), Ben Cox (Dept. of Medical Phys. and Biomedical Eng., Univ. College London, London, United Kingdom), and Bradley Treeby (Univ. College London, London, United Kingdom)

The attenuation of ultrasound propagation through tissue is known to follow a frequency power-law in the time domain. This can be modelled as a loss operator in the equation of state within the Euler equations that takes the form of a fraction time derivative operator. This can be treated through a second order accurate transfer between the fractional time derivative and a fractional Laplacian spacial operator in order to avoid storage of the full time history. This is used for example by the k-Wave toolbox. As an alternate finite history methods have been suggested. Building on recent work, here, we re-write the time fractional derivative as a finite sum using a recursion relation. This allows the time fractional derivative to be directly computed with a static memory requirement. For homogeneous media, the advantage of this over the fractional Laplacian methods may not initially be obvious with a base higher memory requirement and computational cost with only a small increase in accuracy. However, upon introducing a heterogeneous medium with regions of different power law attenuation; the increased computational cost of the time fractional method is contained exclusively within the pre-computation, while the increased cost for the fractional Laplacian is applied to each time step.

Invited Papers

3:20

4pBAb9. Elastodynamic property closures of elastic waves in polycrystalline materials. Christopher M. Kube (Eng. Sci. and Mech., The Penn State Univ., 212 Earth and Eng. Sci. Bldg, University Park, PA 16802, kube@psu.edu)

In polycrystalline materials like many metals, grainy microstructures significantly influence elastodynamics. Bulk waves scattering at grain boundaries cause attenuation and speed variation in waves. This depends on grain characteristics, including local elasticity and spatial properties. These are modeled statistically to homogenize the microstructure or elastodynamic fields, within bounds. For example, the elasticity of a homogenized medium can't exceed that of individual grains. 'Property closure' is the range within which microstructures yield specific properties like Young's modulus. This presentation introduces property closures for modeling elastic wave properties, specifically attenuation and velocity, in metal alloys. The focus is on the modeling framework and integrating microstructure statistics. An exciting prospect of this work is in linking measurable wave properties with other physical quantities dependent on microstructure.

3:40

4pBAb10. Ultrasonic scattering from polycrystals with transversely isotropic material texture. Nathanial Matz (Mech. and Mater. Eng., Univ. of Nebraska-Lincoln, Lincoln, NE), Waled Hassan (Rolls-Royce Corp., Indianapolis, IN), and Joseph A. Turner (Mech. and Mater. Eng., Univ. of Nebraska-Lincoln, University of Nebraska-Lincoln, Lincoln, NE 68588, jaturner@unl.edu)

During laser-based metal additive manufacturing (AM), the melted powder is subjected to rapid cooling and the resulting microstructures often have material texture. In many cases, the texture is uniaxial (i.e., transversely isotropic) such that one symmetry axis defines the material response. Therefore, ultrasonic inspection methods that exploit the scattering from the microstructure are complicated by the resulting texture which affects the coherent propagation and scattering. In this presentation, a generalized approach is described in which the covariance of the elastic modulus tensor for a uniaxial ensemble of cubic crystals is expressed in terms of a fundamental set of constants. These constants are determined for an arbitrary texture from synthetic polycrystals created using DREAM.3D. With this information, calculations for wave velocity, attenuation, and diffuse scattering can be made efficiently for any wave type and propagation direction relative to the material symmetry axis. Results are compared with analytical expressions for simplified cases and then more generalized textures are examined. Finally, prospects for characterization of components created using metal AM are discussed within the context of input data from electron backscatter diffraction measurements. These results are expected to provide insight regarding the inversion of measurement data for texture characterization.

Contributed Paper

4:00

4pBAb11. The effects of nonlinear approximation to mass operator on the wave propagation in polycrystalline materials. Marzieh Bahreman (212 Earth and Eng. Sci. Bldg., Penn State Univ., University Park, State College, PA 16802, mmb7533@psu.edu), Anubhav Roy, and Christopher M. Kube (212 Earth and Eng. Sci. Bldg., Penn State Univ., University Park, PA)

Existing theoretical models utilize the first or third-order smoothing approximations (FOSA and TOSA) to predict attenuation and wavespeed in polycrystalline materials. The FOSA-based model involves a Born approximation, limiting its application to the stochastic (low-frequency) scattering regime. This restriction is a significant factor contributing to its disagreement with FE-based models. The model based on TOSA was subsequently developed, incorporating two additional multiple scattering terms, and increasing the degree of heterogeneity. To extend into the high-frequency regime, the TOSA-based model was solved using numerical integrations. This presentation highlights the analytical solution of the Dyson equation based on the non-linear approximation, wherein the mass operator expression incorporates the mean dyadic Green's function. Evaluating this equation with consideration of higher-order correlations addresses potential truncation errors associated with FOSA and TOSA approximations and encompasses the full range of frequencies from Rayleigh to the geometric asymptote. This presentation will also demonstrate a comparison of the results obtained from the nonlinear approximation with those from FOSA and TOSA across a range of polycrystalline metals.

Invited Paper

4:15

4pBAb12. Characterization of complex stiffness and thickness of isotropic, viscoelastic plates using multi-modal Lamb waves. Clément Despres (Université de Sherbrooke, Sherbrooke, QC, Canada), Patrice Masson (Université de Sherbrooke, Sherbrooke, QC, Canada), Eric Ducasse (I2M, Université de Bordeaux, Bordeaux, France), Michel Castaings (I2M, Université de Bordeaux, Bordeaux, France), and Nicolas Quaegebeur (Université de Sherbrooke, Mech. Eng. Dept., 2500 blvd Universite, Sherbrooke, QC J1K 2R1, Canada, Nicolas.Quaegebeur@USherbrooke.ca)

This paper presents an approach exploiting the sensitivity of Lamb waves for characterizing the mechanical and geometrical properties of plates. The analytical sensitivity functions are first derived in the case of an isotropic plate and are integrated into an iterative inverse problem to optimize its mechanical and geometrical properties based on a zero-finding approach (Gauss-Newton algorithm for a multivariable problem). This method is validated numerically for a viscoelastic plate and shows high accuracy and low computational cost when compared to existing methods. The design af an experimental setup based on dedicated air-coupled transducers is proposed by optimizing the shape and size of the transducers in order to increase sensitivity of the measred signals with respect to the mechanical parameters of the structure of insterest. Experimental validation demonstrates the ability of the methodto assess simultaneously the viscoelastic properties and the thickness of isotropic plate-like structures.

Contributed Papers

4:35

4pBAb13. Shear wave propagation in a fiber-laden viscoelastic waveguide under prestress: Inverse modeling challenges. Lara Nammari (Biomedical Eng., Univ. of Illinois Chicago, 851 S. Morgan St., Rm. 218 SEO (MC 063), Chicago, IL 60607, lnamma2@uic.edu), Dieter Klatt, and Thomas Royston (Biomedical Eng., Univ. of Illinois Chicago, Chicago, IL)

The functional role of skeletal muscle and the hierarchal microstructure and arrangement of fibers within it results in anisotropy and inhomogeneity in both material properties and imposed stresses. Dynamic elastography reconstruction methods for estimating muscle tissue viscoelastic properties that are based on assumptions of homogeneity, isotropy and only bulk wave motion may produce inaccurate estimates. Biases may be introduced in reconstruction by homogenizing muscle with axially aligned fibers and approximating it as transversely isotropic. The significance of these biases, and their interplay with imposed stresses and confounding waveguide effects due to small cross-sectional dimensions, is quantified with a series of numerical finite element and experimental elastography studies on cylindrically-shaped fiber-laden muscle phantoms, with varying fiber dimensions. Specifically, numerical simulations of elastography are conducted on 4-, 60- and 113-fiber models with the aligned fiber cross-sectional area fixed (reduced fiber diameter as fiber count increases) and comprising approximately 40% of the cross-sectional area of the cylindrical phantom that is also undergoing tensile axial pre-loading. Fiber elastic moduli twice that of connective tissue are considered. Experimental studies of the 4-fiber model are used to validate the numerical model. [Funding support: NSF 1852691 and NIH AR071162]

4:50

4pBAb14. Rayleigh-Lamb wave propagation in a prestressed transversely isotropic viscoelastic waveguide: Inverse modeling challenges. Alexandra Vorobyeva (Biomedical Eng., Univ. of Illinois Chicago, Chicago, IL), Qifeng Wang (Northwestern Univ., Evanston, IL), Dieter Klatt (Biomedical Eng., Univ. of Illinois Chicago, Chicago, IL), Kenneth Shull (Northwestern Univ., Evanston, IL), Eric J. Perreault (Northwestern Univ., Evanston, IL), and Thomas Royston (Biomedical Eng., Univ. of Illinois Chicago, 851 S. Morgan St., Rm. 218 SEO (MC 063), Chicago, IL 60607, troyston@uic.edu)

The functional role and structure of skeletal muscle results in anisotropy in both material properties and imposed stresses, as well as waveguide effects. Dynamic elastography reconstruction methods for estimating muscle tissue viscoelastic properties that are rooted in assumptions of isotropy and bulk wave motion may produce inaccurate estimates. The superposition of axially-aligned orthotropy (transverse isotropy) in material properties and axially-aligned prestress conditions due to passive stretch or muscle activation makes it difficult to independently discern how much of the apparent anisotropy is due to the muscle material or the imposed stress field. Furthermore, this stress field may result in large strain conditions that require use of higher order terms in the stress-strain relationship. The significance of this confounding condition and strategies for decoupling material and stress-based anisotropy are investigated with a series of numerical finite element and experimental elastography studies using scanning laser Doppler vibrometry and magnetic resonance elastography. Shear and Rayleigh-Lamb wave motion is studied in a polymeric muscle phantom that is in the shape of a rectangular rod and has either isotropic or transversely isotropic material properties under zero stress conditions. [Funding support: NIH AR071162]

Session 4pEA

Engineering Acoustics: General Topics in Engineering Acoustics

Pranav Agrawal, Chair
Civil and Environmental Engineering, University of California, Los Angeles, 580 Portola Plaza, 5731 Boelter Hall, Los Angeles, CA 90095-1593

Contributed Papers

1:00

4pEA1. The feasibility of building an impedance tube to measure sound absorption coefficients of materials. Nicholas Zomparelli (Aercoustics Eng. Ltd., 1004 Middlegate Rd., Ste. 1100, Mississauga, ON L4Y 1M4, Canada, nicholasz@aercoustics.com)

This presentation focuses on the design and build of an impedance tube to measure sound absorption coefficients of materials. A practical approach was taken to develop an impedance tube from readily available ‘off the shelf’ materials, with the ultimate goal of determining feasibility and accuracy of such a build. The design theory from multiple sources will be discussed, as well as a detailed breakdown of the dimensions and materials used for the apparatus. Measurement methodologies, testing set-up and data processing techniques are also presented to contextualize the results. Finally, comparisons of the tube results are made to existing known sound absorption coefficient data of materials to ultimately assess the accuracy of the low-cost impedance tube and any deficiencies. Methods to improve the design of the tube are also explored, as well as future design improvements.

1:15

4pEA2. PCB-based miniature vibro-tactile display for the visually impaired. Maijie Xiang (Mech. Eng., Tufts Univ., 485 Foley St., unit 1706, Somerville, MA 02145, Maijie.Xiang@tufts.edu), Robert D. White (Mech. Eng., Tufts Univ., Medford, MA), and Jonathan Bernstein (Draper Lab, Cambridge, MA)

An estimated 2.2 billion people worldwide are visually impaired (WHO, 2019), yet current Braille readers are limited to one or two rows of text and cannot display images. Thus, a low-cost, electronically refreshable tactile display for accessing graphical and textual information is needed. We are developing such a display using a printed circuit board (PCB) as the substrate and bottom electrode array, and a metalized Kapton film as the vibrating membrane. Punched foam tape is used as a spacer to create a gap between the board and film so the pixels can be actuated electrostatically. The current prototype has 6 pixels at 3.5 mm spacing, Vibration amplitudes of 8 microns peak-to-peak were achieved in a prototype using a drive voltage of 600V at 200 Hz. This is sensible to the human touch. Further optimization requires matching the mechanical impedance of the finger, and maximizing vibration amplitude in the preferred sensible frequency range between 50 and 300 Hz. Finite element analysis (FEA), laser vibrometry, and nano-newton force-displacement measurements have been used to characterize the system. Future versions will increase amplitude, reduce drive voltages, and increase display resolution to larger arrays of more closely spaced pixels through further miniaturization.

1:30

4pEA3. Speakers—As a sensor for detecting acoustic loads with Artificial Intelligent (AI). Noori Kim (Purdue Univ., West Lafayette, IN), Hui Jun Kim (Dong-eui Univ., Bu, Korea (the Republic of)), Tobias S. Yoo (Purdue Univ., West Lafayette, IN), and Sung-Hee Kim (Dong-eui Univ., 176 Eomgwangro, Busanjin-gu, Busan, Korea (the Republic of), sh.kim@deu.ac.kr)

The potential of using speakers as a sensor to detect ear canal conditions was demonstrated previously. This research contains our ongoing and continuous effort to utilize a single speaker as a sensor by measuring electrical impedance varying acoustic loads. Electrical impedance data (magnitude and phase) from six different acoustic load conditions were collected as features for machine learning (ML) model training. To enhance the learning performance, the data were pre-processed and augmented with normalization and level-shifting techniques, respectively. The raw data were converted to images to optimize the learning performance to classify acoustic loads from the impedance measurement. Several forms of images were experimented such as magnitude only, overlapped magnitude and phase, and rectangular form. A total of 2100 data (350 each) were used with CNN-based State of the Art (SOTA) models such as AlexNet, ResNet, and DenseNet. Both binary and multiclass classifications were performed, showing 0.9716 average accuracy and 0.907 accuracy, respectively. This innovative single-speaker approach using impedance as ML features is poised to revolutionize traditional acoustic sensing research by harnessing the limitless power of AI.

1:45

4pEA4. Experimental verifications of perforation-modulated slow-wave phenomenon. Sihui Li (Mech. Eng., Hong Kong Polytechnic Univ. Shenzhen Res. Inst., Hong Kong), Xiang Yu (Mech. Eng., Hong Kong Polytechnic Univ. Shenzhen Res. Inst., Hong Kong), and Li Cheng (Mech. Eng., Hong Kong Polytechnic Univ. Shenzhen Res. Inst., Hung Hom, Hong Kong 852, Hong Kong, li.cheng@polyu.edu.hk)

Slow wave phenomenon has been theoretically and numerically predicted inside a in a retarding based on the so-called sonic black hole (SBH). A perforation-modulated SBH (PMSBH) retarding structure is then proposed, in which the two physical processes (sound velocity reduction and absorption) can be balanced to enhance the black hole effects by modulating the perforation parameters. This work presents an experimental effort to confirm the theoretically predicted slow wave phenomenon as well as the ABH-induced sound absorption inside a PMSBH. Alongside some theoretical development, this study brings forward the concept of tunable design to improve the performance of SBH structures, which can benefit the design of sound wave manipulation and noise control devices.

2:00

4pEA5. A fluid filled MEMS hydrophone. Georgios Karamanis (Mech. Eng., Tufts, 574 Boston Ave. Rm. 318, Medford, MA 02155, georgios.karamanis@tufts.edu), Jonathan Anderson, Lalitha Parameswaran (Adv. Mater. and Microsystems, MIT Lincoln Lab., Lexington, MA), James Vlahakis (Mech. Eng., Tufts, Medford, MA), Livia Racz, Daniel Freeman (Adv. Mater. and Microsystems, MIT Lincoln Lab., Lexington, MA), and Robert D. White (Mech. Eng., Tufts Univ., Medford, MA)

MEMS hydrophones are of interest to provide a small form factor acoustic sensing capability in water. The majority of previous work on MEMS hydrophones (e.g., Bernstein 1997, Moon, 2010, Gu, 2018) use air backed diaphragms in order to provide increased sensitivity. However, this limits the operating depth of the device due to a reduction in the diaphragm burst pressure. In this work we investigate an architecture that has the potential to increase the operating depth by filling the backing cavity with a heavy fluid. We have designed architectures with multiple acoustic ports (Moon, 2010) and coupled piezoelectric diaphragms, in a bid to maintain sufficient sensitivity while incorporating the filling fluid. The sensing elements are Parylene coated, repackaged Vesper VM1000 aluminum nitride MEMS microphones. It was reported (Travaglione, 2018) that such a device can sense underwater sound, but characterization was limited. In our prototypes, the microphone die are co-packaged onto modified printed circuit boards which include the acoustic coupling elements and required preamplifiers. We report on computational predictions of sensitivity and resolution, and provide initial test results as part of our ongoing investigation. Support from OUSD (R&E) Innovation and Modernization Office.

2:15

4pEA6. The simulation and fabrication of the void head mass by an increase in the bandwidth ultrasonic transducers. Reza Rahnama (Mater. Sci. and Eng., Shiraz Univ., Iran Shiraz, Shiraz 7157743364, Iran (the Islamic Republic of), rezarahnamac@gmail.com)

To achieve the bandwidth of the frequency bandwidth, a new transducer of its main components is developed. The better performance of a transducer is proportional to its head weight. In this study, the internal part of the transducer head is hollow, which reduces the weight of the transducer head and is a factor in increasing the frequency bandwidth. In this paper, the finite element method (FEM) has been used to validate the results obtained from the effects of structural changes in the head applied to the transducer. In this paper, to achieve higher frequency bandwidth, the Tonpilz transducer structure was studied via analysis and computational results using the COMSOL Multiphysics software. The optimized transducer has higher frequency bandwidth, which confirms the effect of using the void of the transducer head mass.

2:30

4pEA7. Sound absorption performance of double layer sandwich structure containing micro-perforations and extended necks. Zacharie Laly (Dept. of Mech. Eng., Sherbrooke Univ., 2500, boul. de l'Université, Sherbrooke, QC J1K 2R1, Canada, zacharie.laly@usherbrooke.ca), Chris Mechefske (Dept. of Mech. and Mater. Eng., Queen's Univ., Kingston, ON, Canada), Sebastian Ghinet, Tenon Charly Kone (Aerosp., National Res. Council Canada, Ottawa, ON, Canada), Noureddine Atalla (Dept. of Mech. Eng., Sherbrooke Univ., Sherbrooke, QC, Canada), and Behnam Ashrafi (Aerosp. Manufacturing Technol. Ctr., National Res. Council Canada, Montreal, QC, Canada)

In this study, a double layer sandwich panels made of micro-perforated plate and perforated plate with extended necks is proposed and its acoustic attenuation is studied using the finite element method. The top panel of the structure is micro-perforated and is connected to a first honeycomb structure cells. The internal panel is perforated, and a neck is connected to each perforation and extends into each cell cavity of the second honeycomb structure. The proposed material design is made of 33 micro-perforations and 33 extended necks and has excellent mechanical stiffness due to the honeycomb structure performance. This study shows how to design the parameters of the necks and the micro-perforations to widen the sound absorption frequency band. With identical micro-perforation diameters and identical necks, the sound absorption coefficient presents two resonance peaks where the surface impedance is close to the impedance of air. The number of the sound absorption peaks increases when different micro-perforation diameters and different necks are used while the sound absorption of conventional double layer sandwich structure is zero over the entire frequency range. The proposed material design can be used to attenuate the noise in various engineering fields that require good bending stiffness.

2:45–3:00 Break

3:00

4pEA8. Characterizing the absolute sound levels of ambient noise in a water tank. Benjamin L. White (Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, bw392@byu.edu), Corey E. Dobbs, and Tracianne B. Neilsen (Phys. and Astronomy, Brigham Young Univ., Provo, UT)

The underwater acoustic measurement tank used at Brigham Young University is affected by several external sound sources which, although easily identified or hypothesized about on a superficial level, must be investigated in order to determine their specific effects on data collection. Therefore, our goal is to characterize the absolute sound levels of these external acoustic sources. The measurements are conducted using B&K and TC4038 hydrophones. Data are collected over frequency bands within the "flat" range of these hydrophones, facilitating the consistent identification of ambient sounds across several bands. Measurements are taken at various points of the tank in order to determine to what extent tank location affects the ambient sound. Data in Volts are converted to Pascals by applying sensitivity values. To evaluate the accuracy of levels obtained, the same method is applied to the signal from a known source and compared to the levels obtained via a deconvolution approach. The conclusions of this study are applicable in obtaining sound levels in our own tank and in other environments (e.g., aquariums). [Undergraduate Research supported by the College of Physical and Mathematical Sciences, Brigham Young University]

3:15

4pEA9. Comparative numerical investigation of flow-induced noise characteristics of high-speed trains using high-resolution compressible Large Eddy Simulation. Kwongi Lee (School of Mech. Eng., Pusan National Univ., 2, Busandaehak-ro 63 beon-gil, Busan, Korea (the Republic of), dlrnjsrl93@pusan.ac.kr), Ceholung Cheong (Ctr. for Adv. Refrigeration, Air-Condition and Energy, Busan, Korea (the Republic of)), and Jaehwan Kim (Hyundai Rotem, Uiwang-si, Korea (the Republic of))

The South Korean Ministry of Land, Infrastructure and Transport has developed a "Comprehensive Plan for 400km/h High-Speed Rail" aimed at enhancing the operational speed of the high-speed railways, currently running up to 300km/h. A short-term goal is to elevate the operating speed of high-speed trains to 370 km/h, up from 320 km/h. Because aerodynamic noise proportionally escalates with the 6th powers of the speed, aerodynamic noise becomes more significant at higher speeds. Consequently, there's a pressing need for design solutions that reduce aerodynamic noise in high-speed trains. This study involves an aeroacoustic analysis using real-scale models of the current model and the preliminary design targeting 370 kph operation. Each model's 8-car formation has been simplified to a 5-car setup. A challenge in predicting aerodynamic noise is the generation of detailed sound sources in the near field and precise noise propagation in the acoustic field. For that, a three-dimensional compressible Large Eddy Simulation technique is employed, utilizing high-resolution grids. This allows for concurrent computation of the external flow and acoustic fields for a real-scale, high-speed train in an open environment. The analysis comprehensively examines the aerodynamic and aeroacoustic properties of each train car, including the major contributors to aerodynamic noise in high-speed trains. The radiated noise is predicted using the Ffowcs Williams and Hawkings equation and is further examined in relation to vortex sound sources.

3:30

4pEA10. Modular design for optimized acoustic properties of multi-layer system. Tao Yang (Tech. Univ. of Munich, Munich, Germany), Martin Eser (Tech. Univ. of Munich, Boltzmannstrasse 15, Munich 85748, Germany, m.eser@tum.de), Marcus Maeder (Tech. Univ. of Munich, Garching near Munich, Germany), and Steffen Marburg (School of Eng. and Design, Tech. Univ. of Munich, Garching, Germany)

It has been demonstrated that multilayer systems have the potential for excellent sound absorption, effectively attenuating sound across a wide range of frequencies and offering a versatile solution for addressing acoustic challenges in various environments. A key consideration is the design of a multi-layer structure and the parameters of each layer to achieve optimal sound absorption performance. Modular design is a crucial aspect of this work, involving the creation of smaller, units that can be customized and combined to achieve specific sound absorption properties. This study applies numerical optimization, inversion, and matrix methods to multilayer sound absorber systems, aiming to provide customized parameters for the modular design of fibrous materials with exceptional sound absorption properties. The effectiveness of the modular design will be validated and improved through impedance tube measurements of the manufactured multilayer systems, ensuring that the resulting sound absorbers meet the desired acoustic performance across diverse real-world scenarios.

3:45

4pEA11. Wind shadowing of a sonic anemometer in low Reynolds number flows. Julia Huckaby (Mech. Eng., Tufts Univ., 200 College Ave., Medford, MA 02155, julia.huckaby@tufts.edu), Robert D. White (Mech. Eng., Tufts Univ., Medford, MA), Ian Neeson (VN Instruments, Brockville, ON, Canada), Don Banfield, and Anthony Colaprete (NASA Ames Res. Ctr., Mountain View, CA)

Computational Fluid Dynamics (CFD) simulations have been carried out to better understand the wind shadowing effects on a sonic anemometer system. The anemometer will measure wind speed on the surface of Mars and the stratosphere of Earth.The wind velocity vector is measured through the difference in acoustic time of flight in 3 directions. However, the structure itself disrupts the flow and creates an internal wake. To understand this and generate correction factors, CFD simulations were run under Mars surface conditions while varying incident angle and flow speed. Spalart-Allmaras turbulence models were employed in both steady and unsteady Reynolds Averaged Navier Stokes (RANS) frameworks and compared to a laminar model. Due to the low Reynold's number ($<$150) we observe no shedding at speeds below 10 m/s. At speeds below 10 cm/s the flow diverts around the structure and sensitivity drops. At intermediate speeds near 5 m/s the wind shadow ratio (WSR) varies substantially with incident wind angle. Design changes to the structure allowed us to improve the worst WSR from 38% to 47%. We were also able to quantify correction factors. Validating comparisons to water tunnel and wind tunnel measurements will be presented.

4:00

4pEA12. Analysis of the linear regime of a power amplifier for underwater acoustics experiments. Natalie Bickmore (Phys. and Astron., Brigham Young Univ., N283 ESC, Provo, UT 84602, njbickmore@gmail.com), Tracianne B. Neilsen, and Corey E. Dobbs (Phys. and Astron., Brigham Young Univ., Provo, UT)

When transmitting sound in a laboratory water tank, a power amplifier is needed to boost the generated signal. If the amplitude of the generated signal is too high, the power amplifier introduces harmonics into the transmitted signal. Measurements were taken to determine the maximum input amplitude for which the TEGAM 2350 power amplifier, which provides x50 amplification, has a linear response. To evaluate when harmonics are generated, acoustical measurements were taken in our acrylic 1.22 m by 3.66 m tank filled with 0.5 m of water. Custom LabView software generated a chirp with a specified amplitude. This signal was sent through the power amplifier and an impedance matching transformer to a B&K 8103 transmitter. The sound was received on the B&K 8103 hydrophones. Both were attached to two different UR10e robots that could be positioned throughout the tank. The receiver hydrophone was moved to three different locations in the tank for a variety of chirp amplitudes to find the maximum amplitude without harmonic generation. Results of these measurements are shown for three frequency bands: 10–50 kHz, 50–100 kHz, and 100–150 kHz . [Undergraduate research supported by the College of Physical and Mathematical Sciences, Brigham Young University]

4:15

4pEA13. Design and construction of the Texas Christian University impedance tube. Claire Elrod (Mech. Eng., Texas Christian Univ., 28840 W Bowie St., Fort Worth, TX 76109, claire.elrod@tcu.edu) and Hubert S. Hall (Mech. Eng., Texas Christian Univ., Fort Worth, TX)

The two-microphone impedance tube test method is a well-established and widely used technique for determining the acoustic absorption coefficient and impedance ratio of materials. This method uses two closely spaced microphones to simultaneously measure the incident and reflected sound waves. A two-microphone impedance tube measurement system made of 6061-T6 Aluminum with a diameter of 3 inches, a 0.5 inch wall thickness, and microphones spaced 2.7 inches apart has been constructed for undergraduate research at Texas Christian University (TCU). These geometrical values suggest a usable frequency range of 50 Hz to 2637.77 Hz as referenced in ASTM Standard E1050-19. Validation of the system was achieved by taking measurements on Owen Corning Type 705 pressed fiberglass board with a 1-inch thickness and comparing them to absorption data provided by the manufacturer. Additional validation measurements were taken without a test sample in place. All validation tests suggest that the TCU impedance tube is an accurate measurement system.

4:30

4pEA14. Acoustic characteristics of OLED panel gaming monitor. Sungtae Lee (LG Display, 245 LG-ro, Paju-si, Gyunggi-do, Korea (the Republic of), owenlee@lgdisplay.com) and Hyungwoo Park (ICT, Dong-Seoul Univ., Seongnam-si, Gyeonggi-do, Korea (the Republic of))

Previous studies have presented that OLED panels can be fully utilized as high-quality audio devices. Through continuous research, methods for improving the sound quality of panel speakers and implementing multi-channel sound have been studied. OLED panel speakers have undergone continuous improvement, allowing this technology to be used in gaming monitors, and research is underway to replace existing desk soundbars or headphones. This OLED panel speaker technology improves picture and sound quality not only for games but also for various contents. In particular, information delivery methods that utilize direct sound coming through the screen, such as CSO(Cinematic Sound OLED), are sophisticated, accurate, and provide the best sense of realism. Sound generated close to the human auditory organ, such as headphones or earphones, impairs hearing health. The proposed method has the advantage of matching the focus of the screen and the sound, as well as the connection between the human auditory organ and the point of sound generation. It has the advantage of being able to maintain a certain distance. In particular, content such as games needs to have a fast response speed of the display, as well as the location and focus of the sound to be clearly defined. In this study, it is introduce the results of analyzing the acoustic characteristics of the OLED panel speaker using the proposed method and utilize this to contribute to commercialization.

4:45

4pEA15. Extending the spatial extent of steady state broadband noise signals using time reversal. Rylee Russell (Phys. & Astron., Brigham Young Univ., BYU, N283 ESC, Provo, UT 84602, rsrussell01@gmail.com) and Brian E. Anderson (Phys. & Astron., Brigham Young Univ., Provo, UT)

Exciting structures with broadband noise can be used to assess structural health by how it responds to the noise. Here time reversal (TR) is used to focus high-amplitude acoustic energy with the goal to use it to excite a structure. Equalization methods help achieve a desired spectral shape at the location of the TR focus. It is desired to increase the spatial extent of the focusing to excite larger regions. This study explores methods to expand the spatial extent first through the superposition of focusing to multiple locations and second by using a spatial inverse filter. These methods widen the spatial extent of the focusing but can come at the cost of the desired spectrum and/or overall level. The overall goal is to see if TR excitation yields higher amplitude equalized noise than the standard broadcast of it.

Session 4pED

Education in Acoustics, Biomedical Acoustics, Musical Acoustics, Physical Acoustics, and Structural Acoustics and Vibration: Teaching Acoustics With (or Without) Math

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

Jill A. Linz, Cochair
Physics, Skidmore College, 815 N Broadway, Saratoga Springs, NY 12866

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

Invited Papers

1:00

4pED1. Using musical acoustics to motivate math learning. Andrew A. Piacsek (Phys., Central Washington Univ., 400 E. University Way, Dept. of Phys., Ellensburg, WA 98926-7422, andy.piacsek@cwu.edu)

A course on musical acoustics, at the high school or introductory college level, can be an effective vehicle for improving student math skills. More importantly, it provides an opportunity to develop an appreciation for the utility of mathematics among skeptical students, including those from groups, such as women and minorities, who have historically been steered away from quantitative studies. Having an answer for the rhetorical "What is math good for?" can boost a student's motivation for learning math and open the door to consideration of STEM careers. This presentation gives an overview of how math skills are explicitly addressed throughout a 10-week (one quarter) general education course, "Physics of Musical Sound," taught at Central Washington University. Examples of specific in-class exercises will be described, as well as how math is incorporated into collaborative term projects. A discussion will be provided of what has worked, and not worked, throughout 25 years of teaching this course.

1:20

4pED2. Using music to motivate learning mathematics in acoustics courses. Gordon P. Ramsey (Phys. Dept., Loyola Univ. Chicago, Loyola University Chicago, Chicago, IL 60660, gramsey@luc.edu)

Mathematics often intimidates students, especially if they feel unprepared for the math level in the course. However, when they see how math and music are intimately related, their attitude often changes. Many fundamental aspects of music are described by math. These include note frequencies, chords and scales, consonance and dissonance, and instrument tuning. The reactions to music of consonance and dissonance are related to the relative ratios of the notes in a chord or note sequence in a piece. The connection between music and math forms the basis for designing instruments that play the music. Instruments must be designed to replicate the note frequencies, while their geometry allows them to be being playable by humans. Mathematics is at the heart of instrument design. I will suggest ways to present these concepts to students for their greater appreciation of how acoustics and math are related.

1:40

4pED3. Improving quantitative literacy through a general education acoustics course. Kurt R. Hoffman (Phys., Whitman College, 345 Boyer Ave. Hall of Sci., Walla Walla, WA 99362, hoffman@whitman.edu)

Most Colleges and Universities have some form of quantitative reasoning requirement for graduation. A narrow view of this requirement is that students need to do some mathematics in the course to demonstrate some facility at calculating quantities. Additionally, students often discuss their discomfort with mathematics when discussing their anxiety about registering in a non-majors physics course. In this presentation, I will discuss strategies used in a musical acoustics course for non-science majors to address proportionalities, graphical representations, and approximations to help students get past equations to think about the underlying physical relationships. Often, I find that students actually are much better at using mathematics than their self-assessments. By giving students different contexts for mathematical thinking, the quantitative elements make more sense to them and calculations become less intimidating. The examples I will discuss focus on using classroom-based experiments to identify relationships between dependent and independent variables on topics such as musical scales, simple oscillators, and fourier analysis.

2:00

4pED4. Building a mathematical model for a simple harmonic oscillator that uses educational methods found in both physics education research and in the language disciplines that make it accessible to undergraduate students in an introductory musical acoustics course. Jill A. Linz (Phys., Skidmore College, 815 N Broadway, Saratoga Springs, NY 12866, jlinz@skidmore.edu)

At the basis of any course in acoustics is the fundamental idea of the simple harmonic oscillator. The term alone is confusing to students with little to no background in physics or math. For courses in musical acoustics at the undergraduate level, this topic is often minimized due to the lack of preparation. This, in turn, results in a more superficial approach to the advanced topics. While deriving the mathematical treatment from first principles is out of reach to these students, approaching the math itself as a language where they are building a description of a simple mass and spring system in their new, yet somewhat familiar, language can be accomplished through pictures, graphs and hands on activities. Students begin to build a vocabulary of "words" that can be strung together in "sentences" that tell the story of how the motion of a mass on a spring is produced. Emphasis is placed on the analogous comparison of physical properties by relating variables such as amplitude and frequency to that of volume and pitch. This model can then be used as a building block to the understanding of how sound is produced, propagated and perceived.

2:20–2:35 Break

2:35

4pED5. Teaching acoustic and electromagnetic waves. David A. Brown (ECE/CIE, Univ. of Massachusetts Dartmouth, 151 Martine St., UMass Dartmouth, Fall River, MA 02723, dbAcousticsdb@gmail.com)

Students have observational experience with waves by speaking and hearing, seeing, and feeling but their use of mathematics may be delayed until introductory physics at middle school, high school, and/or college. Students following a university physics or engineering bachelor's degree are usually required to take a course on electromagnetic wave theory because of the foundational science and/or applications such as communications and observation. Regrettably, it is common that a mathematical description of various wave propagations may be acquired but without a thorough understanding of the underlying physics – especially in electromagnetics. Thus, we have tried to develop an introductory graduate level course that teaches electromagnetics and optic waves and its applications on the foundation of first understanding acoustics. This serves as a preparation for research or vocation. Topics include traveling waves, standing waves, wave impedance, radiation patterns, interference, interferometry, sonar and radar, imaging, waveguides in more. Examples of introducing the mathematics of solution first and then derivations of wave equation is presented. The goal is that mathematics should be a useful language to communicate physical information and that acoustic demonstrations should reinforce learning in multiple physical modalities.

2:55

4pED6. Learning to read an acoustics equation as a descriptive sentence. Daniel A. Russell (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg, University Park, PA 16802, dar119@psu.edu)

Teaching theoretical acoustics at the graduate level requires the use of mathematics, and often the math can be difficult enough that students get lost in the math and can't see the underlying physical concepts the equations are describing. What many students don't realize is that a mathematical equation is just a shorthand notation that condenses a long and complicated sentence into a few symbols. Being able to read and interpret an equation as a sentence (which is more than just reading the names of the symbols or variables) is a valuable skill that can help students grasp the underlying physical concepts. This talk will provide several examples from fundamental acoustics of how helping students learn to read and interpret mathematical equations as sentences can help them reveal and grasp the physical concepts the equations are describing.

Contributed Papers

3:15

4pED7. When is an approximate solution better than an "exact" solution? Steven L. Garrett (Salinas Union High School District, 1736 Lowell St., Seaside, CA 93955, sxg185@psu.edu)

The solution to any problem always has two parts: (1) The numerical "result," with units and possibly a related uncertainty, that is usually provided by a calculation (analytic or numerical) and (2) an intuitively satisfying explanation of that "result." The "exact" results will be presented for the frequency of the first standing-wave mode of a mass-loaded string (*i.e.*, pendulum) as well as the "exact" result for the frequency shift that occurs when the stiffness of a piano string is combined with the string's tension as the restoring forces. The modal frequency of the mass-loaded string will then be approximated by synchronizing the frequencies of a complete sinusoidal half-wavelength connected to a very short pendulum. The exact solution for the "stiff string" problem that was given by Morse in his second edition of *Vibration and Sound* will be shown to be incorrect. A simple approximation, using Rayleigh's method, will provide the correct result. Both approximate methods create an intuitively satisfying explanation for each result. These examples will also confirm John von Neumann's assertion (in 1947) that "Truth is much too complicated to allow anything but approximations."

3:30

4pED8. Early development of a set of interactive tools for undergraduate acoustics education. Noah J. Parker (Acoust., Penn State, 201 Appl. Sci. Bldg., University Park, PA 16802, np.acoustics@gmail.com) and Daniel A. Russell (Graduate Program in Acoust., Penn State Univ., University Park, PA)

Using interactive learning materials has been found to increase the efficiency of teaching math and physics concepts fundamental to acoustic phenomena; this presentation will discuss work done to explore their use in undergraduate acoustics courses. During the Spring 2024 semester at Penn State, three undergraduate courses in acoustics were offered, each aimed at different types of students at differing levels of mathematical rigor. These were: a general education elective for freshmen from non-STEM majors, a sophomore/junior level course required for communication science disorders majors, and a junior/senior elective for STEM majors. Observation of each class during the semester identified a variety of topics for which a web-based interactive tool could improve student understanding and engagement. This talk will discuss some of the topics identified as potentially benefiting from an online interactive tool, with a specific focus on one interactive webpage which may serve as an example template for the style and delivery of future developments. This specific interactive webpage

explains basic trigonometric functions in a manner approachable by students at any level of mathematical confidence, by illustrating geometric implications with interactive graphs, explaining with a conversational tone, and addressing the etymology of some of the mathematical terms.

3:45

4pED9. A tutorial and assignment on the analysis and syntheis of music to provide acoustic foundations for courses in the Psychology of Music and in Music Cognition. Annabel J. Cohen (Psych., Univ. of PE, Dept. of Psych., University of PE, 550 University Ave., Charlottetown, PE C1A 4P3, Canada, acohen@upei.ca)

Students in courses in the Psychology of Music and in Music Cognition benefit from an understanding of aspects of musical acoustics. A tutorial assignment has been developed to provide hands-on experience in synthesizing acoustic elements and analyzing real music using Audacity®, freely downloadable software. The tutorial emphasizes visualizing waveforms and spectrographs of target sounds. For the synthesis component, students generate sine tones related by small integers on separate tracks and combine the tracks to create complex waves. Students must answer questions regarding frequency, amplitude, pitch, loudness, and sound quality. They are also asked to synthesize a sequence of successive notes of the chromatic scale, so as to understand the significance of the semitone ratio $2^{1/12}$ (i.e., 1.05946), and they are asked to synthesize a sequence of successive notes from the diatonic scale. For the analysis component of the tutorial, students download a favourite piece of music, provide waveform and spectrographic representations, and answer questions about the representation of pitch, loudness, time and timbre. They also locate auditory repetitions and identify different sections of their piece. The students are asked to summarize what they have learned from the assignment. The assignment can be adapted for work in teams, and a detailed grading rubric facilitates student evaluation.

4:00

4pED10. Considerations for mathematical backgrounds and goals of music students in introductory acoustics courses. Daniel Choi (Newcomb & Boyd, 303 Peachtree Ctr. Ave. NE, Ste. 525, Atlanta, GA 30303, dchoi@newcomb-boyd.com)

Many music institutions include an introductory course in acoustics as prerequisites for certain classes and/or requirements for graduation of a degree program. With the topic involving science, there are mathematical concepts that are naturally introduced to the students. For instruction of music students this may result in challenges as the range of their mathematical background is generally larger. Aside from difficulties in the coursework, it is also common where students will not utilize the mathematical concepts from the course throughout their career. Therefore, it is crucial to survey the typical level of the students' interests and skills in mathematics to align it with the course objectives and determine the methods of instruction. This presentation will focus on modifying the level of mathematical instruction based on the goals of typical music students and the real-world applications of the learned content.

4:15

4pED11. Viewpoints on math in acoustics education and their reflections in the acoustics program's curriculum at Purdue University. Yangfan Liu (Ray W. Herrick Laboratories/School of Mech. Eng., Purdue Univ., Ray W. Herrick Labs., 177 S, Russell St., West Lafayette, IN 47907, yangfan@purdue.edu), Junfei Li, J. S. Bolton, and Patricia Davies (Ray W. Herrick Labs., Purdue Univ., West Lafayette, IN)

Faculties in the acoustics and noise control program at Purdue university will share their viewpoints on the roles of mathematics in acoustics education and math instruction practices in the acoustics curriculum at Purdue university. It is suggested to consider mathematical training from two prospective: (1) teaching math techniques and (2) cultivation of a mathematical mindset. Opinions will be shared on how each math training aspect benefits acoustics students with engineer careers and research careers as well as the long-term influences of math education on industry-academia collaboration. We will also briefly discuss how these views on math education are reflected in Purdue's current acoustics curriculum and could guide further improvements of the curriculum.

Session 4pID

Interdisciplinary and Student Council: Hot Topics in Acoustics

Ian C. Bacon, Cochair
Physics & Astronomy, Brigham Young University, 333 W 100 S, Provo, UT 84601

Chirag Gokani, Cochair
Walker Department of Mechanical Engineering, University of Texas at Austin, 10000 Burnet Road, Austin, TX 78758

Heui Young Park, Cochair
Pennsylvania State University, State College, PA 16801

Ann Holmes, Cochair
Psychological & Brain Sciences, University of Louisville, 2082 Douglass Blvd., Apt 5, Louisville, KY 40205

Chair's Introduction—1:00

Invited Papers

1:05

4pID1. Robustness of vortex wave based communications to operational variables. Mark Kelly (GA Tech, 313 Oakland St., Decatur, GA 30030, mkelly75@gatech.edu) and Chengzhi Shi (Univ. of Michigan, Atlanta, GA)

Orbital angular momentum (OAM) based acoustic communications are a promising method of increasing bandwidth in underwater acoustic communications networks as their unique phase patterns form an orthogonal basis set on which communications protocols may be based. The underwater acoustic environment, however, is highly complex and dynamic. These complexities make developing systems capable of long-range communications challenging. Methods for using BELLHOP's ray tracing software to simulate the time series of vortex-wave based communications signals are presented. Additionally, this study explores the robustness of the inner product deconvolution method of decoding OAM-based communications against various operating conditions, considering both environmental and platform variations. The effects on channel cross-talk are assessed against sound speed profile uncertainty, position errors, Doppler Effect, and turbulence in the water column. These analyses shed light on the operational implementation of OAM-based communications systems.

1:20

4pID2. AI-based headphones for augmenting human hearing. Bandhav Veluri, Malek Itani, Tuochao Chen (Univ. of Washington, Seattle, WA), and Shyamnath Gollakota (Univ. of Washington, Paul G. Allen School of CSE, Seattle, WA 98195, gshyam@cs.washington.edu)

Consider being able to listen to the birds chirping in a park without hearing the chatter from other hikers, or being able to block out traffic noise on a busy street while still being able to hear emergency sirens and car honks. We introduce semantic hearing, a novel capability for hearable devices that enables them to, in real-time, focus on, or ignore, specific sounds from real-world environments, while also preserving the spatial cues. Results show that our system can operate with 20 sound classes and that our transformer-based network has a runtime of 6.56 ms on a connected smartphone. In-the-wild evaluation with participants in previously unseen indoor and outdoor scenarios shows that our proof-of-concept system can extract the target sounds and generalize to preserve the spatial cues in its binaural output.

1:35

4pID3. Reciprocal dynamics in nonlinear systems. Andrus Giraldo (Korea Inst. for Adv. Study, Seoul, Korea (the Republic of)), Ali Kogani (Concordia Univ., Montreal, Montreal, QC, Canada), and Behrooz Yousefzadeh (Concordia Univ., Montreal, 1455 De Maisonneuve Blvd. W., Rm. EV-4.139, Montreal, QC H3G 1M8, Canada, behrooz.yousefzadeh@concordia.ca)

Nonreciprocal vibration transmission is an important problem from a fundamental perspective and because of the additional functionalities that it enables in mechanical or acoustic devices. A common realization of nonreciprocal dynamics relies on implementation of nonlinear internal forces within the system. In this talk, we identify and discuss different manifestations of nonreciprocity in nonlinear systems. As a necessary condition, nonreciprocal dynamics is realized in nonlinear systems with broken mirror symmetry. We show that a second symmetry-breaking parameter can counteract the original asymmetry and ultimately restore reciprocal dynamics in a system with broken mirror symmetry, even near the system resonances. Thus, breaking the mirror symmetry is a necessary but insufficient condition for realizing nonreciprocity in nonlinear systems. We also highlight the contribution of phase to nonreciprocal vibration transmission by discussing response regimes that are characterized by nonreciprocal phase shifts. Our findings showcase the potential of asymmetry to serve as an additional design parameter for devices that operate based on nonreciprocity.

1:50

4pID4. Hearing your voice in the crowd: How do bats echolocate in dense swarms? Laura Kloepper (Dept. of Biological Sci., Univ. of New Hampshire, 230 Spaulding Hall, Durham, NH 03824, laura.kloepper@unh.edu), Stephen P. Blackstock (Univ. of Texas at Austin, Austin, TX), Amaro Tuninetti (Dept. of Biological Sci., Univ. of New Hampshire, Providence, RI), Dieter Vanderelst (Univ. of Cincinnati, Cincinnati, OH), and Michael R. Haberman (The Univ. of Texas at Austin, Austin, TX)

In order to navigate their environment and find food, bats rely on comparing returning echoes to their broadcast signal. Through decades of research, we have a good understanding of how single bats accomplish this task, but we still don't know how bats in dense groups can echolocate without interference from others. A spectral Jamming Avoidance Response (JAR) has long been proposed as one strategy, in which bats flying together shift frequencies to avoid interference. Recent work, however, questions this strategy, and suggests bats may not use any JAR for sensing in groups. Bats have long served as bioinspiration for active sensing devices, so understanding strategies for collective sensing in bats has implications for applications of synthetic sonar. In this talk, I will review the "hot topic" of bat biosonar in swarms and show new data from our research that suggests spectro-temporal variation may play a role for sensing in even the largest, densest bat swarms.

THURSDAY AFTERNOON, 16 MAY 2024 — ROOM 205, 1:00 P.M. TO 5:00 P.M.

Session 4pNS

Noise, Physical Acoustics and Engineering Acoustics: Methods for Community Noise Testing and Analysis II

Alexandra Loubeau, Cochair
NASA Langley Research Center, MS 463, Hampton, VA 23681

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

Duncan Halsead, Cochair
Aercoustics Engineering Ltd, 1004 Middlegate Rd, Mississauga, L4Y 0G1, Canada

Contributed Papers

1:00

4pNS1. Automotive tire facility noise characteristics. Peter VanDelden (RWDI, 600 Southgate Dr., Guelph, ON N1G 4P6, Canada, peter.vandelden@rwdi.com), Lorenzo Carboni, and Slavi Grozev (RWDI Windsor, Windsor, ON, Canada)

The environmental noise influence of facilities that change, replace or repair tires has notable acoustic characteristics, including the sound from air impact wrenches. Air impact wrenches are a ubiquitous tool at such facilities, where they are used to remove and replace the nuts that hold wheels on cars and trucks. The sound is commonly emitted through open doorways. Tire facility sound data was evaluated for impulsive characteristics using procedures described in the newest version of ISO 1996-3. The dynamic nature of air impact wrench sound also allowed the ISO 1996-3 approach to be evaluated.

1:15

4pNS2. Acoustic detection and classification of all-terrain vehicles and automobiles in pennsylvania state parks. Carter A. Paprocki (Acoust., Penn State, 201 Appl. Sci. Bldg., University Park, PA 16802, cap6296@psu.edu), Andrew Barnard (Acoust., Penn State, University Park, PA), and Peter Newman (Recreation, Park, and Tourism Management, Penn State, State College, PA)

The preservation of today's natural environment is more important than ever before; with the encroachment of civilization, protected areas are necessary. This protection must go beyond just saving wildlife but also protecting natural soundscapes for visitors to experience "dispersed low-density outdoor recreation." However, some visitors use the wilds for motorized recreation such as Off-Road Vehicles, All Terrain Vehicles, and Side-by-Side Vehicles producing elevated levels of noise which could impact the other

visitors. In order to determine how the protected areas were used, listening stations were placed at high-traffic areas to monitor the soundscape for one summer. This dataset was analyzed to understand the number and impact of anthropogenic noise sources. This research develops a training dataset that could be implemented with supervised machine-learning models to automate event detection and classification.

1:30

4pNS3. Sound levels of the UVU pedestrian bridge. Jacob Sampson (Phys., Utah Valley Univ., 800 W University Parkway, Orem, UT 84058, jacobbsampson@gmail.com), Isaac Setter (Phys., Utah Valley Univ., Orem, UT), Brian D. Patchett (Phys., Utah Valley Univ., Orem, UT), Abolfazl Amin, Joshua Goates (Utah Valley Univ., Orem, UT), and Bonnie Andersen (Phys., Utah Valley Univ., Orem, UT)

Utah Valley University provides a pedestrian bridge to travel from a train station to the school over Interstate-15. Due to traffic noise, people using the bridge could be exposed to sound levels that damage their hearing. Sound level measurements have been made at several locations on the bridge with two different instruments, the first being an Extech (Knoxville, USA) SL400 noise dosimeter and the other being a Larson Davis (Depew, USA) 831C sound level meter. Sound levels have ranged from 55 to 102 dB, depending on time of day and location on the bridge. In addition to our experimental results a COMSOL model was also developed to simulate the interactions of the freeway noise with the bridge, demonstrating how the geometry of the bridge affects the noise exposure of pedestrians. The Occupational Safety and Health Administration (OSHA) limit for non-occupational noise exposure of 100 dB is 15 minutes, suggesting that pedestrians that linger on the bridge could be exposed to damaging levels of sound. This research seeks to better understand and quantify the noise levels that pedestrians experience on the bridge.

1:45

4pNS4. Acoustic measurements of wind farm noise inside homes and comparison with the ISO 9613 standard. Umberto Berardi (Toronto Metropolitan Univ., 350 Victoria St., Toronto, ON M5B 2K3, Canada, uberardi@ryerson.ca), Gino Iannace, Amelia Trematerra, and Antonella Bevilacqua Bevilacqua (Università della Campania Luigi Vanvitelli, Parma, Italy)

To achieve the energy transition objectives, many wind farms are installed to produce electricity by exploiting wind energy. The production of electricity through wind farms is widespread in the world. One of the problems complained about by people who live near wind farms is the noise emitted by the rotation of the blades. This work reports acoustic measurements inside homes near wind farms. The acoustic measurements were performed for different wind speeds and directions. Furthermore, to verify the effectiveness of the simulation method of the propagation of the noise generated by wind farms, the measured values of the sound levels were compared with those obtained from the numerical simulations using the provisions of the ISO 9613 standard, and the wind turbines are considered point-like sound sources. The comparison between the measurements carried out in the field with the theoretical simulations has highlighted that the calculated levels are lower than the measured ones, this means that in the acoustic impact forecasting procedures, the noise emitted by the wind farms is underestimated and is one of the reasons why the Wind farms in operation are a source of complaints from the populations living nearby.

2:00

4pNS5. Empowering residents of communities near wind farms to record and document wind farm noise through the use of a phone app. Heather L. Lai (Eng. Programs, State Univ. of NY at New Paltz, 208 Eng. Innovation Hub, SUNY New Paltz, New Paltz, NY 12561, laih@newpaltz.edu), Anne C. Balant (Commun. Disord., State Univ. of NY at New Paltz, New Paltz, NY), and Kimberly A. Riegel (Phys., Farmingdale State College, Farmingdale, NY)

The impact of disturbances caused by wind farm noise (WFN) on residents can be difficult to predict. In talking with advocates for residents living near wind farms, it has become clear that some residents are negatively impacted by WFN, but the specific features that cause the greatest impacts are not well understood. A few studies have been published correlating in-home recordings of WFN with annoyance ratings by occupants in nearby residences, and some local municipalities have recorded SPLs at residences (such as an upstate NY town appointed volunteer noise officer who measures sound pressure levels on the property of concerned residents), but the vast majority of residents' experiences go undocumented. In this talk, we will present a plan for using a cell phone app, "Auditive," originally developed for collecting users' responses to urban noise to provide residents living near wind farms with the opportunity to document their experiences with WFN while making simultaneous recordings. These recordings in conjunction with the user ratings will allow identification of qualitative aspects of WFN that cause the most annoyance from a large sample of individuals and may also provide a sense of agency for residents who are most negatively affected.

Invited Papers

2:15

4pNS6. Improvements to a mobile app for community noise and assessment data collection. Kimberly A. Riegel (Phys., Farmingdale State College, 2350 Broadhollow Rd., Farmingdale, NY 10803, riegelk@farmingdale.edu) and Jody Resko (Social Sci., Queensborough Community College, Bayside, NY)

A mobile phone application was developed to collection health history, real time noise levels and participants' perceptions. The goal of this application is to collect data that can directly link community response, noise levels and community health from a large number of participants. This will help identify where noise is causing issues related to physical and mental health on a community scale. In order to identify these effects at a community level large numbers of responses are critical. Ease of use is essential to ensure that participants remain willing to participate in the study and continue to submit samples. This needs to be balanced with the usefulness of the data collected by the public. Previously, a beta version of the app was deployed to evaluate the useability and effectiveness of data collection. Several issues with the data collection were identified and improvements to the app were implemented. Evaluation of the data and data visualization options will be presented using the most recent version of the application.

2:35

4pNS7. Relationship between intra-community noise tolerance distributions and fitted dose-response funtions. Richard D. Horonjeff (48 Blueberry Ln., Peterborough, NH 03458, rhoronjeff@comcast.net)

Increasingly, researchers are employing fitted exposure-response functions to not only characterize the relationship between sound level and prevalence of annoyance but also to compare the relative noise tolerance of individual communities. The sound level at which 50% of the population is annoyed has been advanced as a community noise tolerance metric. This paper combines prior research and recent simulations to reframe the discussion in terms of plausible distributions of individual (personal) noise tolerances within a given community; the integrals of those (density) distributions become the (cumulative) observed exposure-response relationships. Various

tolerance distributions are developed, their resulting cumulative exposure-response relationships calculated, and the ability of commonly-employed functional forms to represent those cumulative relationships presented. The results of the current investigation strongly suggest that data sets which include annoyance fractions of 50% have the greatest chance of accurately predicting the 50%-annoyed sound level. The further the data set lies from the 50%-annoyed point the more dependent an extrapolated estimation becomes on the fitted functional form accurately representing the true cumulative relationship. Errors of estimation are shown for a number of tolerance distributions, functional forms and experimental data ranges. Finally, some remedial recommendations are offered.

2:55–3:10 Break

3:10

4pNS8. A unified framework for creating soundscape perception indices based on the SSID Protocol. Andrew Mitchell (Inst. for Environ. Design and Eng., Univ. College London (UCL), Central House, 14 Upper Woburn, London WC1H 0NN, United Kingdom, andrew.mitchell.18@ucl.ac.uk), Francesco Aletta, Tin Oberman, and Jian Kang (Inst. for Environ. Design and Eng., Univ. College London (UCL), London, United Kingdom)

The soundscape approach provides a basis for considering the holistic perception of sound environments, in context. While steady advancements have been made in methods for assessment and analysis, a gap exists for comparing soundscapes and quantifying improvements in the multi-dimensional perception of a soundscape. To this end, there is a need for the creation of single value indices to compare soundscape quality which incorporate context, aural diversity, and specific design goals. Just as a variety of decibel-based indices have been developed for various purposes (e.g., LAeq, LCeq, L90, Lden, etc.), the soundscape approach requires the ability to create novel indices for different uses, but which share a common language and understanding. We therefore propose a unified framework for creating both bespoke and archetypal single index measures of soundscape perception based on the soundscape circumplex model, allowing for new indices to be defined in the future. The implementation of this framework is demonstrated through the creation of a public space typology-based index using survey data collected under the SSID Protocol. Indices developed under this framework can enable a broader and more efficient application of the soundscape approach in design, planning, and regulation.

Contributed Papers

3:30

4pNS9. Investigation of whistle noise impacting a residential subdivision. Gregory Dennis (Valcoustics Canada Ltd., 41 Corianne Ave., Whitby, ON L1M2H9, Canada, greg.pwdennis@gmail.com)

There have been complaints from residences within a residential subdivision in New Tecumseth, Ontario, of train noise from the Canadian Pacific Railway (CPR) line that passes directly east of the development. The complaints relate to sound levels experienced within the dwellings during train pass-bys, most notably from the use of whistles at the grade level crossing southeast of the site. Sound level measurements were completed to quantify the indoor and outdoor sound levels at several dwellings closest to the rail line. The measured sound levels were compared to predictions made using the Ministry of the Environment, Conservation and Parks (MECP) noise model Sound from Trains Environmental Analysis Method (STEAM). This investigation discusses shortfalls in STEAM, which resulted in the sound levels (from whistle noise) being under predicted at the dwellings. The study also compares the sound levels predicted using the Federal Railway Administration (FRA) horn noise model to those measured on site and predicted using STEAM.

3:45

4pNS10. Revised comprehensive new definition of noise. Daniel Fink (The Quiet Coalition, The Quiet Coalition, P.O. Box 533, Lincoln, MA 01733, DJFink@thequietcoalition.org)

A revised comprehensive new definition of noise is proposed. Noise: a) For living things, noise is unwanted and/orharmfulnd. b) In engineering and electronics, noise is any unwanted disturbance within a useful frequency band, such as undesired lectric waves in a transmission channel or device. c) In scientific measurements, noise is erraic, intermittent, or statistically random oscillation. The revised comphrehensive new definition builds on the Acoustical Society of America/American National Standards institute definition to include technical considerations, and acknowledges the harmful effect of noise on plants. It updates the noise definition presented at the 2019 Acoustical Society of America winter meeting, *noise is unwanted and/or harmful sound*. Unlike the standard definition, *noise is unwanted sound*, that new definition emphasized that unwanted sound is harmful, able to cause adverse auditory and non-auditory health effects, and that wanted sound can also caus auditory damage. The Proceedings of Meetings on Acoustics article based on that presentation has been cited 37 times. The prevvious new definition opens the 2021 American Public Health Association policy statement, *Noise as a Public Health Hazard*, was adopted for use by the International Commission on Biological Effects of Noise (ICBEN) in 2023, and added to the ICBEN Constitution.

4:00

4pNS11. Reduction of noise impact from HVAC equipment that affects students in an Ecuadorian university. Felix Ramon Silva Tumbaco, David Andres Lince Correa, Galo Durazno (Escuela Superior Politecnica del Litoral, Guayaquil, Ecuador), and Carlos Yoong (Wood PLC, 2020 Winston Park Dr #700, Oakville, ON L6H6X7, Canada, carlos.yoong@woodplc.com)

The Faculty of Mechanical Engineering in the Coastal Polytechnic School located in Guayaquil, Ecuador currently presents an issue associated to the high noise levels from HVAC equipment located in close vicinity of classrooms. During the design and installation of the air conditioning system, noise control was never considered. This paper explores the different solution options that are pratical in an academic context while also taking into account local products, costs and environment. By taking into account the local guidelines, a group of students conducted the project from the field work to the reporting. These students then presented this study as their Capstone Project.

4:15

4pNS12. Sound and noise exposure dosimetry at large-scale music events and festivals. Wannes Van Ransbeeck (Ghent Univ., Ghent, Belgium), Nele De Poortere, Marcel Kok (dBControl, Zwaag, Netherlands), and Sarah Verhulst (Ghent Univ., Technologiepark 126, Zwijnaarde 9052, Belgium, s.verhulst@ugent.be)

Regulations governing sound exposure at amplified-music events aim to protect the audience from hearing damage. However, current level monitoring practices often rely on a single measurement position that is considered representative of individual exposure, overlooking crucial variation. This study addresses this lacking insight by analyzing individual dosimeter-based sound-level exposure at music events and its relation to existing guidelines. Dosimeters secured to 42 individual participants (19 females, aged 18–25) recorded octave bands and A/C-weighted sound levels at six large-scale music events (+20.000 visitors). Individual exposure durations varied

between 4.4 and 22 hours. Comparisons of exposure were made against local, German and WHO regulations. At four out of six events, individual doses surpassed the WHO's 100 dBA recommendation, with most subjects also exceeding the ISO1999 occupational limit and the WHO's 100 dBA-for-4-hours threshold (16Pa2h). Equivalent exposures ranged between [85.2,104.5] dBA (LA,eq) and [97.1,119.6] dBC (LC,eq) among participants. Additionally, LC,peak values fluctuated between [133.6,143.5] dBC. These findings highlight a discrepancy between the fixed-location noise exposure monitoring, as per country-specific legislation, and the actual individual exposure experienced by event attendees. Our results can inspire safe-listening guidelines for such music events, or help in deciding which single position measure is most representative. Work supported by UGent BOF-IOP EarDiMon and dBControl

4:30–5:00 Panel Discussion

THURSDAY AFTERNOON, 16 MAY 2024 ROOM 206, 1:00 P.M. TO 5:25 P.M.

Session 4pPA

Physical Acoustics, Engineering Acoustics and Structural Acoustics and Vibration: Novel Methods and Applications in Nondestructive Evaluation

Luz D. Sotelo, Cochair
Purdue University, 2550 Northwestern Ave, 1900D, West Lafayette, IN 47906

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

Joseph A. Turner, Cochair
Mechanical and Materials Engineering, University of Nebraska-Lincoln, University of Nebraska-Lincoln, Lincoln, NE 68588

Chair's Introduction—1:00

Contributed Papers

1:05

4pPA1. Measurement of acoustic nonlinearity and anelasticity in metals with phase-sensitive nonlinear reverberation spectroscopy and noncontacting electromagnetic-acoustic transduction. Ward Johnson (National Inst. of Standards and Technol., 325 Broadway, MS 647, Boulder, CO 80305, wjohnson@boulder.nist.gov)

Nonlinear reverberation spectroscopy (NRS), in each of its various implementations, focuses on characterization of resonant acoustic nonlinearity and anelasticity through measurements of time-dependent changes in resonant frequency and vibrational amplitude during free decay after acoustic excitation. A general advantage of NRS over stepped-frequency nonlinear resonance techniques is the relatively short duration of free decay and associated small temperature drift during data acquisition. A recent innovation in NRS employs noncontacting electromagnetic-acoustic transduction and phase-sensitive superheterodyne reception to provide additional advantages of eliminating contributions to nonlinearity and loss from contacting transduction, simplifying signal analysis, and improving signal-to-noise ratios. Measurements of resonant axial-shear modes in a 7075 Al cylinder demonstrate precision of fractional frequency shifts on the order of 0.1 ppm, exceeding by two orders of magnitude the greatest reported precision of measurements achieved with nonlinear resonant ultrasound spectroscopy (NRUS). The high precision and resolution of the technique enable sensing of differences in nonlinearity and anelasticity associated with residual porosity at industrially relevant levels of less than half a percent in commercially pure additively manufactured aluminum.

1:25

4pPA2. Developments in an acoustic resonance test for the detection of manufacturing anomalies in hydroelectric generator stator windings. Kevin Venne (Hydro-Québec, 1800 Bd Lionel-Boulet, Varennes, QC J3X 1S1, Canada, venne.kevin2@hydroquebec.com), Mathieu Kirouac, Hélène Provencher, Mélanie Lévesque, and Mathieu Soares (Hydro-Québec, Varennes, QC, Canada)

To meet the ever-increasing demand for electricity, Hydro-Québec (HQ) is seeking to simultaneously increase the power of its generating stations while improving its service quality. Thus, the company has tasked its

4p THU. PM

research institute to investigate innovate methods to meet the aforementioned goals. Of interest in the current study is the development of an acoustic resonance test (ART) to improve the quality control (QC) in the manufacturing of hydroelectric generator stator windings. Since HQ and its suppliers are investigating new fabrication methods for stator windings to meet the required timelines and increased power requirements, QC is required to ensure the service quality of its new hydroelectric generators. Typical manufacturing anomalies found in stator windings are delamination and air pockets between insulation layers. Such anomalies can result in an acceleration in the degradation of the winding insulation, which reduces the service quality of hydroelectric generators. To benchmark the ART method, the results of the suspected locations of the anomalies along the stator windings were compared with an acoustic camera and the locations were dissected and inspected under microscope for validation. Ten different stator windings were tested and two metrics (variations in both force and frequency responses) were found to indicate the location of delamination sites in the stator windings and corroborated with the result of both the acoustic camera and the dissections.

Invited Paper

1:40

4pPA3. Acoustoelastic characterization of aluminum plates using zero group velocity Lamb modes. Rosa E. Morales (Lawrence Livermore National Lab., 7000 East Ave. Livermore, CA 94550, morales31@llnl.gov), Niket Pathak (Mech. Eng., Univ. of Colorado, Boulder, Boulder, CO), Jordan Lum (Lawrence Livermore National Lab., Livermore, CA), Christopher M. Kube (Eng. Sci. and Mech., The Penn State Univ., University Park, PA), Todd Murray (Mech. Eng., Univ. of Colorado, Boulder, Boulder, CO), and David M. Stobbe (Lawrence Livermore National Lab., Livermore, CA)

Acoustoelasticity, a characteristic of material anharmonicity, gives rise to a link between wave propagation velocity and the stress state in materials. Ultrasonic techniques to monitor this coupling, particularly with high sensitivity and in a noncontact manner, can have widespread application both in the quantification of applied and residual stress and in the characterization of nonlinear material behavior through measurement of higher order elastic constants. Here, we use a laser ultrasonic technique to excite and detect zero group velocity (ZGV) Lamb wave resonances in aluminum plates under uniaxial loading. A laser line source is used to excite these resonances at different orientations with respect to the applied load and the signals are detected using an interferometer. The effects of stress and source orientation on ZGV resonance frequencies are validated using the theory of acoustoelastic Lamb wave propagation. In addition, a model-based inversion technique is used to extract Murnaghan's third-order elastic constants from measurements of the stress dependence of the first two ZGV modes generated parallel and perpendicular to the applied load. Laser generation and detection of ZGV resonances is shown to be an effective and powerful approach for the noncontact and nondestructive acoustoelastic characterization of elastic waveguides.

Contributed Papers

2:00

4pPA4. Ellipsometry study of surface acoustic waves for viscoelastic material characterization: Estimation of complex Lamé coefficients versus the frequency. Aziz Bouzzit (CY Cergy Paris, 5 mail Gay Lussac, Neuville sur Oise, Ile de France 95000, France, aziz.bouzzit@cyu.fr), Loïc Martinez, Andres Arciniegas mosquera, Stéphane Serfaty, and Nicolas Wilkie-chancellier (CY Cergy Paris, Neuville sur Oise, Ile de France, France)

Adequate regarding material characterization, surface acoustic waves (SAWs) have a unique elliptic polarization that can be distinguished by two parameters. The first parameter, the H/V value, is the ratio between the in-plane and out-of-plane components of the particle's motion while it undergoes the surface wave. The second parameter is the orientation angle θ of the elliptic motion, defined as the angle between the major axis of the ellipse and the horizontal axis. The present paper proposes a method for estimating the viscoelastic isotropic material properties from these two parameters. A complete characterization is done by identifying the variation of the complex Lamé coefficients versus the frequency. This is achieved by employing the Quaternion Fourier Transform (QFT) on the quantitative measurement of the polarization. The propagative characteristics of the SAW, represented by the complex wavenumber, are extracted using the Prony algorithm. The inverse problem is based on the theoretical models of the propagation of SAW on viscoelastic materials. The proposed method is implemented and tested on space-time signals extracted from numerical simulation and experimental setup.

2:15

4pPA5. Buried object localization by spectral analysis of surface wave reflections. David Baumann (Elec. & Comput. Eng., Lake Superior State Univ., Sault Ste Marie, MI), Emily Hagelthorn, Andrew Heiny (Mech. Eng., Lake Superior State Univ., Sault Ste Marie, MI), Robert Hildebrand (Mech. Eng., Lake Superior State Univ., 650 W Easterday Ave. Sault Ste Marie, MI 49783, rhildebrand@lssu.edu), Morgan Kelly, Haluk Kucuk (Mech. Eng., Lake Superior State Univ., Sault Ste Marie, MI), Dustin Mangone, Omar Nobani (Elec. & Comput. Eng., Lake Superior State Univ., Sault Ste Marie, MI), and Masoud Zarepoor (Mech. Eng., Lake Superior State Univ., Sault Ste Marie, MI)

A proposed method of buried object localization, based upon spectral analysis of surface wave reflections, is investigated numerically. This arises from a hypothesis to the effect that reflectivity would be maximized at some intermediate wavelength, shorter ones associated with disturbances too shallow to substantially excite the object as a secondary (reflecting) source, and longer ones with disturbances involving such volumes of earth in motion that the object become insignificant as a reflector. Discernment of that intermediate wavelength of maximum reflectivity might, thereby, provide an index of the object's depth. Horizontal distance from echo return time, and azimuth from phase array techniques, could complete a localization methodology. Presented, accordingly, are 2D FEA simulations, in which narrow-banded surface wave trains are excited by a point source on the surface of an elastic medium, a void provided at some horizontal distance and depth, the strength of reflected surface motions thereafter examined in relation to frequency. These bear out the hypothesis, finding maximal reflectivity for a surface wavelength about two-thirds of the object depth. Proposed is that this could serve as the basis for a horizontal stand-off detection method, especially for applications like landmines, for which it may be undesirable to scan from above.

Invited Papers

2:30

4pPA6. Development of a scanning acoustic microscopy-based structural prior for microtexture region characterization. Laura Homa (Univ. of Dayton Res. Inst., Dayton, OH), Tyler Lesthaeghe (Univ. of Dayton Res. Inst., Dayton, OH), Matthew Cherry (Air Force Res. Lab., Fairborn, OH), Erik Blasch (Air Force Res. Lab., 875 N. Randolph, Ste. 325, Arlington, VA 22203, erik.blasch.1@us.af.mil), and John Wertz (Air Force Res. Lab., Beavercreek, OH)

Nondestructive evaluation (NDE) plays a crucial role in ensuring aircraft availability. The current NDE paradigm often relies on mono-modal testing and signal-over-threshold criteria to provide robust defect or damage detection, not characterization. One example is found in the risk-based management of surface-breaking cracks in metal, where cracks of a given size can be detected by eddy current testing (ECT) with a calculable probability. Yet, there are cases where detection proves insufficient. Consider the case of microtexture regions (MTR) found in certain titanium alloys, which can increase the risk of cold dwell fatigue failure when found above a certain size and in specific orientations relative to the surrounding material. At present, the size and orientation of MTR cannot be characterized using only one NDE modality. In this work, a data fusion-based solution to MTR characterization is developed. First, two inspection methods—scanning acoustic microscopy (SAM) and ECT—are selected, where each method is individually capable of only partial characterization. Then, matching component analysis is used to develop a surrogate forward model relating MTR orientation to ECT output. This data is then inverted using boundaries provided from the SAM data as a structural prior.

2:50–3:10 Break

Contributed Papers

3:10

4pPA7. A novel ultrasonic technique for the inspection of a plate heat exchanger. Pooja Dubey (George W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., 2 Rue Marconi, Georgia Tech-Europe, Metz 57070, France, pooja.dubey@gatech.edu) and Nico Declercq (George W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., Metz, France)

Biofilm formation in process industries, water treatment devices, and drinking water pipe networks pose a significant risk to public health and cause a variety of operational issues. One of the major challenges is the inspection of biofilms in heat-exchanging devices such as plate heat exchangers. A plate heat exchanger (PHE) is integral to food industries, water treatment plants, and others. Conventional techniques for cleaning biofilm formation in these plates (foulant) include chemical cleaning, steam, and hydro-blasting. However, they are inefficient and labor-intensive because of limitations such as increased handling risk, over-cleaning, and corrosion. Thus, developing novel techniques for real-time monitoring of biofilm growth on these devices is critical for efficient working. Previous studies were restricted to the development of ultrasound-assisted heat exchangers to reduce the deposits in these plates. However, only a few studies have investigated ultrasound as a probable monitoring tool. Thus, this research aims to explore the nonlinear ultrasonic parameters using the second harmonic generation technique as a real-time tool for monitoring biofilms in PHE. The proposed research will help design more effective ultrasonic-assisted plate heat exchangers to achieve maximum heat transfer efficiency.

3:25

4pPA8. Thickness characterization of test specimens using frequency-modulated ultrasonic signal generated via. fluidic oscillator. Viswa R. Sunkavalli (Zerstörungsfreie Prüfmethoden für das Bauwesen, Bundesanstalt für Materialforschung und -prüfung, Unter den Eichen 87, Berlin, Berlin 12205, Germany, viswa.sunkavalli@bam.de), Christoph Strangfeld, and Stefan Maack (Zerstörungsfreie Prüfmethoden für das Bauwesen, Bundesanstalt für Materialforschung und -prüfung, Berlin, Germany)

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

3:40

4pPA9. Ultrasonic scattering behavior in austenitic NiTi with varying grain size and precipitate characteristics. Olivia J. Cook (Eng. Sci. and Mech., Penn State Univ., 212 Earth Eng. Sci., University Park, PA 16801, ojc3@psu.edu), Foster K. Feni, Mique A. Gonzales, Reginald F. Hamilton, and Andrea P. Arguelles (Eng. Sci. and Mech., Penn State Univ., State College, PA)

Shape memory alloys like NiTi exhibit high strain recovery due to a reversible martensitic transformation when cooled or stressed, enabling actuation and biomedical applications. The characteristics of the stress-induced martensitic transformation in Ni-rich NiTi depend on the heat treatment. For example, aging induces the precipitation of $Ni_{:4}Ti_{:3}$ and lowers the critical stress needed for martensitic transformation. However, comprehensive characterization of these nanoscale precipitates is challenging as only small areas can be inspected efficiently. Ultrasonic immersion testing may assist in developing linkages between structure and transformation behavior as ultrasonic scattering is sensitive to the microstructure and easily mapped on the bulk scale. To this end, the ability of scattering behavior to discern between different precipitation characteristics must be probed to establish the sensitivity of ultrasonic parameters to changes in precipitation. This work explores the dependence of wave speed, attenuation, and backscattered energy on varying annealing and aging heat treatments of Ni-rich NiTi. The annealing treatments result in a gradient of grain sizes, while the aging treatments result in nucleation and growth of precipitate phases with increasing hold time. Variations in the ultrasonic parameters elucidate the sensitivity to changes in the microstructure, which are more quantitatively explored through existing analytical scattering models.

3:55

4pPA10. Ultrasonic evaluation of additively manufactured components with embedded structures. Harshith Kumar Adepu (Mech. Eng., Purdue Univ., 500 Central Dr., B36, West Lafayette, IN 47906, adepu@purdue.edu), Meher Mirza, Jacey Birkenmeyer, and Luz D. Sotelo (Mech. Eng., Purdue Univ., West Lafayette, IN)

In recent years, the use of ultrasonic nondestructive evaluation (NDE) for *in situ* and *ex situ* assessment of Additive Manufacturing (AM) parts has garnered interest due to its proficiency in defect detection and material characterization. This study focuses on evaluating the ability to resolve internal structures created by AM and hybrid AM using ultrasonic NDE as well as assessing the impact of microstructure heterogeneity. For this purpose, first the parameter space for laser powder bed fusion (LPBF) of SS316L is mapped in a Lumex Avance 25 system. Based on this parameter mapping, optimal process parameters are identified and sets of process parameters are selected to vary the thermal history of the samples, while minimizing the influence of porosity. Two groups of samples are printed with these parameter sets: a solid group of samples, and a group with internal structures, such that each internal structure sample has an equivalent solid sample. Measurements of ultrasonic velocity, attenuation, and backscatter amplitude are collected and compared against the intended geometry to assess the impact of non-optimal processing on the ability to resolve internal structures with ultrasound. The viability and limitations of using ultrasound to assess internal features created with AM are discussed.

4:10

4pPA11. Quantitative ultrasonic characterization of Ti-6Al-4V components with surface roughness. Sydney N. Assalita (Eng. Sci. and Mech., Penn State Univ., 212 Earth and Eng. Sci. Bldg., University Park, PA 16802, sna5228@psu.edu), Olivia J. Cook, and Andrea P. Arguelles (Eng. Sci. and Mech., Penn State Univ., State College, PA)

Ultrasonic immersion testing enables nondestructive characterization of material microstructure through quantitative linkages between wave propagation parameters and features of interest. However, accurate measurements of metrics such as wave speed and attenuation generally require smooth test samples to ensure the reflected wave packets are not distorted. Thus, rough surfaces are mechanically polished before ultrasonic testing, which compromises the nondestructive nature of the measurement, distorting the sample geometry and imparting local damage. This challenge is particularly salient with the advent of manufacturing processes, such as metal additive manufacturing, that result in rough samples that must be characterized for heterogeneities in their as-manufactured state. This work examines the influence of surface roughness on longitudinal wave speed and attenuation in wrought and additively manufactured Ti-6Al-4V specimens. Samples are measured in an ultrasonic immersion setup before and after mechanical polishing using varying transducer frequencies and focal profiles. The resulting data informs the error generated in ultrasonic measurements as a function of roughness parameters, thereby informing the sample preparation needed for desired precision levels. Lastly, new measurement protocols are explored to correct for the presence of surface irregularities in a point-by-point fashion. This work was supported by the Robert W. Young Award for Undergraduate Student Research.

4:25

4pPA12. The impact of the summer undergraduate research or internship experience in acoustics program on my knowledge and passion for material science and acoustics. Aja Leatherwood (case western reserve Univ., 10900 Euclid Ave., Cleveland, OH 44106, axl862@case.edu), Haley N. Jones, and Andrea P. Arguelles (Mater. Sci. and Eng., Penn State Univ., State College, PA)

Throughout an enriching 10-week experience with the Acoustical Society of America's Summer Undergraduate Research Experience in Acoustics program (SUREIA), I had the privilege of contributing to the innovative research at Penn State's Arguelles Lab, focusing on ultrasound as a tool for characterizing material properties and defects. Despite my background in communication science, I immersed myself in material science through the lab. I collaborated with graduate students to learn immersion ultrasound testing on bulk ceramic materials and processed data using MATLAB. At the end of the opportunity, I presented my findings at Penn State University and the SURIEA Program Open House. Beyond the academic pursuits, the true richness of this experience lies in the meaningful connections cultivated within my SUREIA cohort and lab members. These connections have not only enhanced my understanding of material science but have also provided me with a network of like-minded individuals whom I can collaborate with in the future. Additionally, the Arguelles Lab has provided me with invaluable mentorship and guidance, enabling me to enhance my research skills and broaden my knowledge in the unfamiliar yet fascinating field of material science.

4:40

4pPA13. Abstract withdrawn.

4:55

4pPA14. Topological acoustic sensing using the geometric phase. Keith Runge (New Frontiers of Sound (NewFoS) Ctr., Univ. of Arizona, 1235 E. James E. Rogers Way, Dept. of Mater. Sci. and Eng. University of Arizona, Tucson, AZ 85721, krunge@arizona.edu) and Pierre A. Deymier (New Frontiers of Sound (NewFoS) Ctr., Univ. of Arizona, Tucson, AZ)

We introduce a method, topological acoustic sensing, which exploits changes in the geometric phase of acoustic waves to sense defects in some structure or environment. This method is illustrated in two cases of perturbations taking the form of (1) a mass defect located on an array of coupled acoustic waveguides, and (2) a small subwavelength object on a flat surface submerged under water. We represent the state of the acoustic field in the unperturbed and perturbed cases as multidimensional vectors. The change in geometric phase is obtained by calculating the angle between those vectors. This angle represents a rotation of the state vector of the wave due to scattering by the perturbation. By exploiting sharp topological features spanned by the acoustic field multidimensional state vector, we show that this geometric phase sensing modality can have higher sensitivity than magnitude-based sensing approaches.

5:10

4pPA15. Long-term acoustic emission monitoring of a new alkali-activated material for sealing structures in nuclear waste repositories. Anna M. Sklodowska (8.2 Non-destructive Testing Methods for Civil Eng., Bundesanstalt für Materialforschung und -prüfung (BAM), Unter den Eichen 87, Berlin 12205, Germany, anna.sklodowska@bam.de), Vera Lay (8.2 Non-destructive Testing Methods for Civil Eng., Bundesanstalt für Materialforschung und -prüfung (BAM), Berlin, Germany), Franziska Baensch (Deutsches Institut für Normung (DIN), Berlin, Germany), Ernst Niederleithinger (8.2 Non-destructive Testing Methods for Civil Eng., Bundesanstalt für Materialforschung und -prüfung (BAM), Berlin, Germany), and Hans-Carsten Kühne (7.4 Technol. of Construction Mater., Bundesanstalt für Materialforschung und -prüfung (BAM), Berlin, Germany)

The crucial part of nuclear waste storage is the construction of sealing structures made of reliable, well-understood, and safe materials. Within the SealWasteSafe project, we compared the performance of an innovative alkali-activated material (AAM) and standard salt concrete (SC), as potential materials for sealing structures for nuclear waste repositories. Two 340-liter-cubic specimens were studied for up to ~250 days by a multisensory monitoring setup. Specifically, the long-term acoustic emission monitoring aimed to analyze the development of microstructural changes within materials. The monitoring analysis showed fewer acoustic emission events in AAM compared to SC in the first 61 days. After approximately two months of monitoring, the number of AE events in AAM significantly exceeded the number of events in SC. The analysis showed, however, that the increased AE activity was mainly caused by the surface effects of the AAM material and not by the formation of cracks within the material. This contribution presents the use of acoustic emission analysis, both in the time and frequency domains, for monitoring and characterization of materials with potential use as engineering barriers for nuclear waste repositories.

Session 4pPP

Psychological and Physiological Acoustics: Psychological and Physiological Acoustics Best Student Poster Award Session

Z. Ellen Peng, Cochair
Boys Town National Research Hospital, 555 North 30th Street, Omaha, NE 68131

Daniel R. Guest, Cochair
Department of Biomedical Engineering, University of Rochester, 601 Elmwood Ave, MC 5-6483, Rochester, NY 14620

All posters will be on display from 1:00 p.m. to 5:00 p.m. Authors of odd-numbered abstracts will be at their posters from 1:00 p.m. to 3:00 p.m. and authors of even-numbered abstracts will be at their posters from 3:00 p.m. to 5:00 p.m.

Contributed Papers

4pPP1. An exploratory investigation of acoustic features underlying arousal and valence perception of vocalizations from non-speaking individuals. Simon M. Radhakrishnan (Elec. Eng. & Comput. Sci., Massachusetts Inst. of Technol., 77 Massachusetts Ave., Cambridge, MA 02139, rdkn@mit.edu), Amanda M. O'Brien (Speech and Hearing Bioscience and Technol., Harvard Univ., Cambridge, MA), Thomas Quatieri (MIT Lincoln Lab., Massachusetts Inst. of Technol., Lexington, MA), and Kristina T. Johnson (Elec. & Comput. Eng., Northeastern Univ., Boston, MA)

Emotion perception of vocalizations, especially from individuals with no or few spoken words, remains an underexplored topic in acoustical research. The aim of this exploratory study was to identify acoustic features within non-speech vocalizations correlated with perceived arousal and valence. 364 vocalizations were selected from the open-access ReCANVo dataset, comprising non-speech communicative sounds by non-speaking individuals with autism and neurodevelopmental disorders. 108 listeners independently rated each vocalization for arousal and valence on a 5-point Likert scale (78624 total ratings). We then used random forest and elastic net regression techniques to determine correlations between acoustic-based features and perceived affect. Using 4 published (e.g., GeMAPs) and 2 custom feature sets (dim 12, 28) encompassing duration, intensity, pitch, formant, and spectral centroid features, we evaluated each model using R-squared and mean-squared error. Our 12-feature custom elastic net model achieved the highest performance for predicting arousal (R-sq = 0.763, MSE = 0.075), with intensity features weighted highest, comparable to other sets. Our 28-feature random forest model, which included vocal quality features, significantly outperformed other models in valence prediction (R-sq = 0.491; MSE = 0.139, $p < 0.001$). While further validation on a held-out test set is essential, these exploratory results expand our understanding of affective acoustic features of non-speech sounds from individuals with neurodevelopmental disorders.

4pPP2. Effects of musical training experience and sentence difficulty on noisy speech recognition in second language learners. Shuhang Chen (Appl. Psych. Programme, Dept. of Life Sci., BNU-HKBU United Int. College, q030016008@mail.uic.edu.cn, Zhuhai, Guangdong 519000, China, q030016008@mail.uic.edu.cn), Siheng Li (Appl. Psych. Programme, Dept. of Life Sci., BNU-HKBU United Int. College, Guangzhou, China), and Yu Li (Appl. Psych. Programme, Dept. of Life Sci., BNU-HKBU United Int. College, Zhuhai, Guangdong, China)

Correctly recognizing second-language speech in a noisy environment is a difficult task for second-language learners. Early works have indicated the beneficial effects of musical training in native language acquisition and development. However, it remains unknown how musical training experiences influence second-language speech-in-noise (SIN) recognition in second-language learners. To examine this question, we recruited 45 right-handed young adults who speak English as their second language and showed different musical training experiences over the past ten years, and employed an English SIN test in which two factors were manipulated, signal-to-noise ratio level (SNR; quiet, + 5 dB, −5 dB) and sentence difficulty (easy, hard). The percentage of words correctly identified from sentences was used in the data analyses. The results revealed a significant interaction between musical training experience, SNR level, and sentence difficulty. Further analyses revealed that compared with the short musical training group, the long musical training group exhibited larger differences between easy and hard sentences in the quiet condition. Overall, these results provide new evidence for the benefits of music training experiences for SIN recognition in the context of second language learning. [Work supported by the Humanities and Social Sciences Foundation of Ministry of Education of China 20YJCZH079.]

4pPP3. Adults show own-age advantage in inter-talker similarity ratings. Yu Li (Dept. of Life Sci., BNU-HKBU United Int. College, 2000 Jintong Rd., Tangjiawan, Zhuhai, Guangdong 519087, China, yuli@uic.edu.cn), Tongyu Qiu, Junze Li (Dept. of Life Sci., BNU-HKBU United Int. College, Zhuhai, Guangdong, China), Sabrina Yanan Jiang (Macau Univ. of Sci. and Technol., Macao, China), and Linjun Zhang (Peking Univ., Beijing, China)

Own-age advantage in talker identification has been reported in adults (Cooper *et al.*, 2020; Creel & Jimenez, 2012). Whether the own-age advantage can also be observed in an inter-talker similarity rating task is still an open question. In the current study, we first reversed sentences (completely unintelligible but have most sound structures preserved) recorded from 8 adults and 8 children. Then we asked adults to rate the inter-talker similarity of pairs of these reversed sentences by using a visual analog scale (0-10 with 0 denoting "totally different" and 10 "totally same; see Fleming *et al.*, 2014). For each participant, we calculated the average inter-talker similarity of pairs of reversed sentences separately for child and adult talkers. A lower inter-talker similarity reflects higher sensitivity to the differences in voices between talkers. The results showed that the inter-talker similarity was significantly higher for child talkers than adult talkers, suggesting that adults are more sensitive to the differences in voices between adult talkers. The finding demonstrates the own-age advantage in adults in a similarity rating task. The current study extended our existing knowledge of talker processing. [Work supported by the Humanities and Social Sciences Foundation of Ministry of Education of China 20YJCZH079.]

4pPP4. Neurophysiological biomarkers of speech-in-competition performance under different conditions of attentional load. Erick Correa-Medina (Lab. of Cognit. and Clinical Neurophysiology, National Inst. of Neurology and Neurosurgery, Circuito Ciudad Universitaria Avenida, C.U., Mexico City 04510, Mexico, neurick.psi@gmail.com), Rodolfo Solís-Vivanco (Lab. of Cognit. and Clinical Neurophysiology, National Inst. of Neurology and Neurosurgery, Mexico, Mexico), Tess Koerner, Frederick J. Gallun (Oregon Hearing Res. Ctr., Oregon Health and Sci. Univ., Portland, OR), Aaron Seitz (Ctr. for Cognit. and Brain Health, Northeastern Univ., Riverside, CA), and Sebastian Lelo de Larrea-Mancera (Ctr. for Cognit. and Brain Health, Northeastern Univ., Boston, MA)

A body of research has identified a complex integration of perceptual and cognitive processes that can modulate comprehension of speech-in-competition (SIC). However, it is not well understood how the neurophysiological correlates associated with SIC performance vary under different conditions of attentional load. Here, we analyzed the neurophysiological correlates of a SIC task using variations in masker type (spatialization, gender, language), target-to-masker ratios (TMRs), and presence of a simultaneous working memory (WM) task to manipulate attentional load during SIC performance. We used a counter-balanced block design to record EEG in each condition in 15 undergraduate young adults without hearing difficulties. SIC was assessed using the spatial release from masking task in the PART digital application. Behavioral results indicated significant differences in accuracy depending on TMR across types of maskers. We did not observe behavioral differences in SIC performance associated with the inclusion of a parallel WM task. Quantitative EEG showed a series of potential EEG biomarkers that can complement behavioral measurement in discriminating different attentional load conditions of SIC performance. These post hoc observations are discussed in relation to the extant literature and will be further investigated in future work.

4pPP5. Enhanced adult voice recognition ability in children. Tongyu Qiu (Life Sci., BNU-HKBU United Int. College, Zhuhai, Guangdong, China, 2642082558@qq.com) and Yu Li (Life Sci., BNU-HKBU United Int. College, Zhuhai, Guangdong, China)

Language acquisition during infancy and childhood is much influenced by various factors including social interactions with others. Previous works have indicated that children likely benefit more from interactions with people showing advanced language skills such as adults rather than peers who show less or inferior language skills. These benefits could be reflected by better recognition of adult voices containing rich linguistic information and conveying advanced language skills. We therefore hypothesized that children recognize adult voices better compared to child (peer) voices. To test this hypothesis, we used a voice recognition task to assess children's ability to recognize adult voices and child voices in a group of 8–9 years old children. The results showed children exhibited superior recognition for adult compared to child voices, which is in line with our hypothesis. The finding demonstrates the importance of interacting with adults who show advanced language skills in language acquisition from the perspective of voice recognition. We discussed the result by introducing the own-age advantage in voice recognition observed in adults (Cooper *et al.*, 2020) and provided implications for better facilitating child language acquisition. [Work supported by Guangdong Basic and Applied Basic Research Foundation 2022A1515110748 and UICstartup R72021207.]

4pPP6. Hearing of warning sounds under the settings of railway running noise considering age-related hearing loss. Kei Hoshino (Ergonomics Lab., Railway Tech. Res. Inst., hoshino.kei.10@rtri.or.jp, k.hoshino@ruri.waseda.jp, Tokyo 1858540, Japan, k.hoshino@ruri.waseda.jp), Ayako Suzuki (Ergonomics Lab., Railway Tech. Res. Inst., Tokyo, Japan), and Yasuhiro Oikawa (Intermedia Art and Sci., Waseda Univ., Tokyo, Japan)

Railroad drivers may become drowsy because of their irregular work schedules, such as late nights and early mornings. Therefore, in our research, we developed a system that detects driver drowsiness using images based on facial expressions and issues a warning sound to assist drivers during driving. However, it is necessary to function warning sounds even in loud running noise areas or various running noise qualities to make the system more practical. Moreover, it is necessary to consider the effect of a driver's age-related hearing loss on the setting of the warning sound volume in noisy environments, even for those who meet the hearing criteria as drivers. This is because the number of elderly drivers in their 60 s will increase in the future. Therefore, we investigated the appropriate volume of the warning sound under various running noise types and levels. Then, we compared it between groups of people in their early 20 s and 60 s regarding their hearing ability and the way they hear warning sounds under running noise. The results provide new insight into the relation between hearing ability and the ability to hear warning sounds under driving noise.

4pPP7. Localizing sounds revolving at very high velocities: An auditory wagon-wheel effect. Noa Kemp (Dept. of Physiol./School of Information Studies, McGill Univ./Ctr. for Interdisciplinary Res. in Music Media and Technol., 3460 Mc Tavish St., Montreal, QC H3A 0E6, Canada, noa.kemp@mail.mcgill.ca), Ulysse Lefeuvre, Cynthia Tarlao, and Catherine Guastavino (School of Information Studies, McGill Univ./Ctr. for Interdisciplinary Res. in Music Media and Technol., Montreal, QC, Canada)

Localizing sound sources as they move around us is a critical function of the auditory system. Yet most research focuses on static sound sources or sources moving at slow velocities. The present work explores circular trajectories at very high velocities well above the velocity at which we lose the sense of direction ($\sim$2.5 rot/s with white noise). As the number of rotations per second approaches the fundamental frequency of the spinning sound, a sense of direction re-emerges. This creates what has been described informally as the auditory equivalent to the wagon-wheel effect: the sound appears to move in one direction when the velocity is below the fundamental frequency, and it appears to move in the opposite direction when the velocity is above the fundamental frequency. We report on two experiments testing this effect with a 200-Hz complex sound using adaptive VBAP spatialization on a 16-loudspeaker array. Experiment 1 ($N = 15$) confirmed that participants perceived opposite directions when the velocity was below or above the fundamental frequency. Experiment 2 ($N = 7$) explores the relationship between this effect at very high velocities and the ability to track sound at slower velocities. Preliminary findings suggest some overlap between localization processes at slow and high velocities.

4pPP8. Effect of subsequent context on the real-time interpretation of ambiguous target words in spectrally degraded speech. Anna R. Tinnemore (Neurosci. and Cognit. Sci. Program, Univ. of Maryland, 0100 LeFrak Hall, College Park, MD 20742, annat@umd.edu), Sandra Gordon-Salant, and Matthew J. Goupell (Hearing and Speech Sci., Univ. of Maryland, College Park, MD)

Subsequent context in a sentence can be used to revise the interpretation of a word that was not clearly heard, a frequent occurrence in degraded speech. One strategy for avoiding the cognitive cost associated with revising an initial interpretation is to delay commitment to an interpretation until the end of the sentence, sometimes called a "wait-and-see" strategy. Listening with a significant hearing loss (i.e., constantly hearing degraded speech), may prompt a listener to use this strategy even in relatively easy listening conditions. This study used an eye-tracker to record gaze patterns during a phoneme classification task of ambiguous target words in two levels of background noise ($+5$ and -5 dB SNR). Participants either used cochlear implants (CIs) or were presented 8-channel noise vocoded speech. The target words were at the beginning of sentences; the subsequent context cued one target word interpretation (based on phoneme classification) or the other. We hypothesized that listeners who use CIs will demonstrate a delaying strategy at both levels of background noise, while listeners who have acoustic hearing will only demonstrate a delaying strategy at the highest level of background noise. These findings have implications for rehabilitative strategies to improve communication outcomes in adults with CIs.

4pPP9. Profile analysis in the inferior colliculus: physiology and modeling studies. Swapna Agarwalla (Biomedical Eng. Dept., Univ. of Rochester, University of Rochester, Rochester, NY 14642, swapwiz16@gmail.com), Daniel R. Guest, and Laurel H. Carney (Biomedical Eng., Univ. of Rochester, Rochester, NY)

Profile-analysis is a strategy for assessing sensitivity to differences in spectral shape [Green, Oxford University Press, 1988]. Despite extensive behavioral data, the neuronal mechanisms underlying profile-analysis remain elusive. To bridge this gap, extracellular recordings were made in awake rabbit inferior colliculus (IC) and simulated using a computation model. The standard stimulus was a log-spaced, n-component, zero-phase, equal-amplitude complex tone, presented diotically. The central component was incremented in the target stimulus. The discharge rate as a function of characteristic frequency (CF) was inferred by shifting the stimulus spectrum past the CF of each neuron. When the increment frequency was near CF, IC neurons had rates that decreased as the increment increased, contradicting a simple energy-based code. In some cases, the differences between rates for target and standard were largest for stimuli with component spacing that yielded the lowest thresholds in listeners. Responses of IC neurons that were excited by amplitude-modulated stimuli were consistent with model predictions for profile-analysis stimuli [Maxwell *et al.*, *JASA* **147**, 3523 (2020), Guest *et al.*, b*ioRxiv* (2023)]; however, the models did not explain responses of other types of IC neurons. Potential factors contributing to this discrepancy, such as off-CF inhibition and chirp selectivity, will be further explored. Support: NIHR01-DC010813

4pPP10. Neural mechanisms of spatial auditory attention with magnified interaural level difference cues. Benjamin N. Richardson (Neurosci. Inst., Carnegie Mellon Univ., 5702 Darlington Rd., Apartment 3, Pittsburgh, PA 15217, bnrichar@andrew.cmu.edu), Jana Kainerstorfer (Neurosci. Inst., Carnegie Mellon Univ., Pittsburgh, PA), Barbara Shinn-Cunningham (Carnegie Mellon Univ., Pittsburgh, PA), and Christopher A. Brown (Commun. Sci. and Disord., Univ. of Pittsburgh, Pittsburgh, PA)

Bilateral cochlear implant users struggle in spatial release from masking (SRM) tasks, likely due to restricted access to interaural time difference (ITD) cues. Instead, they must rely on interaural level difference (ILD) cues; however, our previous behavioral experiments suggest that magnification of ILDs can facilitate SRM. Here, we probed the neural mechanisms underlying the benefit of magnified ILDs. We tested 18 normal-hearing subjects in an anechoic chamber. Listeners heard target and masker sequences of object and color words from opposite (left and right) quarterfields and were asked to detect color words in the target stream. Both streams were spatialized using either 50 μS ITD (ITD50), 500 μS ITD (ITD500), a broadband 10 dB ILD (ILD10), or the largest naturally occurring frequency-specific ILD (70 degrees; ILD70n). We recorded task-elicited hemodynamic responses in dorsolateral prefrontal cortex (DLPFC) using functional Near-Infrared Spectroscopy. Subjects performed best in the ITD500 and ILD10 conditions. Hemodynamic response magnitudes were smaller for ITD50 than for all other conditions, consistent with frontal activity increasing when perceptual segregation is possible and spatial attention can be deployed successfully. These data show that magnified ILD cues enhance SRM and that the benefit ILDs confer arises because listeners can engage cognitive attentional processes.

4pPP11. Performance on a spatially selective auditory attention listening task for sources in the horizontal or coronal planes. Grace E. Otto (Neurosci. Graduate Program, Western Univ., London, ON, Canada) and Ewan A. Macpherson (National Ctr. for Audiol., Western Univ., 1201 Western Rd., EC 2262, London, ON N6G 1H1, Canada, ewan.macpherson@nca.uwo.ca)

Listeners can use spatially selective auditory attention (SSAA) to focus on one talker in a complex acoustic scene. SSAA has been primarily investigated with targets and distractors arrayed in the frontal horizontal plane. In this study we compared normally hearing human listeners' performance on static and dynamic SSAA tasks in a frontal target/distractor configuration to performance with sources arrayed in either the rear horizontal plane, the overhead coronal plane, or the "underhead" coronal plane, all of which provide similar binaural difference cues with which to differentiate target and distractor locations. To achieve the coronal plane configurations, the listener's head was tipped forward to align the normally vertical axis with the horizontal plane. In the SSAA task, listeners attempted to report a 4-digit sequence of spoken digits from the target location while ignoring two simultaneous equal-intensity sequences spoken by the same talker presented from flanking loudspeakers separated by $\pm 22.5^\circ$ from the target. Listeners either held their head still (static conditions) with the front (horizontal configurations) or top (coronal configurations) of the head oriented toward 0 azimuth, or were oscillated passively at $\sim$0.14 Hz with an amplitude of $\pm$45 degrees about the vertical axis (dynamic conditions).

4pPP12. Exploring the asymmetry in perception and production of mandarin tones: effects of high variability phonetic training with visual animation on native Thai speakers. Yilin Xiang (School of Foreign Studies, Xi'an Jiaotong Univ., No. 28 Xianning Rd. (W), Xi'an, Shaanxi, 710049, P. R. China, xiangyl@stu.xjtu.edu.cn), Bing Cheng (School of Foreign Studies, Xi'an Jiaotong Univ., Xi'an, Shaanxi, China), and Xiaojuan Zhang (Xi'an Jiaotong Univ., Xi'an, Shaanxi, China)

This study investigated the impact of a web-based high variability phonetic training (HVPT) program integrated with pitch contour visualization on Mandarin tone acquisition by native Thai speakers. Through a pre-test and post-test design, involving tasks of word identification and recordings, we examined the progress in both perception and production of four Mandarin tones in 20 participants. Our findings reveal a notable asymmetry in progress: While production accuracy improved significantly across most tones, as evidenced by higher ratings from Mandarin native speakers, perceptual accuracy did not show a parallel enhancement, except for Tone 1. Further analysis confirmed significant variability in training effectiveness across different tones. These findings underscore the pivotal role of perception-oriented training in enhancing tonal production for native speakers of tonal languages. Moreover, the differential improvements across four tones highlight the need for tailored training approaches in both perception and production. Key words: High variability phonetic training (HVPT), pitch contour visualization, Mandarin tone acquisition, perception and production asymmetry.

4pPP13. Auditory brightness in room reverberation. Chad Bullard (Indiana Univ., 2631 East Discovery Parkway, Bloomington, IN 47408, charbull@iu.edu) and Jennifer Lentz (Indiana Univ., Bloomington, IN)

This work examines the impact of room reverberation on the auditory perceptual brightness of sounds. We hypothesized that room size would impact the perception of brightness, with smaller rooms (less reverberant) leading to sounds being perceived as more bright than larger rooms. We generated single-note instrument sounds from five different instruments using Logic Pro which were reverberated in three different-sized simulated rooms. Reverberant stimuli were presented in sequential room-size pairs (small-medium, small-large, medium-large) in a randomized order. Participants (15 musicians) were asked to rate the brightness of the second sound as compared with the first using a Likert-style scale, and the stimuli were blocked by instrument such that in each block a participant would only hear room-size comparisons of one instrument. Results were analyzed using an ordered-probit regression of the brightness ratings against the spectral centroid of the stimuli. We found no clear effect of room size. However, brightness ratings were weakly correlated with the average spectral centroid as shown in previous literature, and a stronger correlation was found with the steady-state portion of the sound. This suggests room size has little effect on brightness. Future work should examine the effect of different segments (e.g., attack, steady-state) on perceived brightness.

4pPP14. Human perception of impact sounds suggests auditory intuitive physics. Vinayak Agarwal (Mech. Eng., MIT, 77 Massachusetts Ave., 46-4065, Cambridge, MA 02139, vinayaka@mit.edu), James Traer (Psychol. and Brain Sci., Univ. of Iowa, Iowa City, IA), and Joshua H. McDermott (MIT, Cambridge, MA)

Upon hearing objects collide, humans can estimate many of the underlying physical attributes, such as the objects' material and mass. Although the physics of sound generation are well established, the inverse problem that

listeners must solve – of inferring physical parameters from sound – remains poorly understood. In this work, we show that humans leverage an understanding of acoustical physics to constrain their perceptual inferences, allowing them to disambiguate multiple object properties from a single impact sound. We derived a linear generative model of impact sounds, combining theoretical acoustics with empirically measured statistics of object resonances. We used an analysis-by-synthesis algorithm to infer mode parameters from recorded object impulse responses. We then fit distributions to these parameters, from which object impulse responses could be sampled. Perceptual experiments demonstrated that humans could judge material and mass from sound alone, even when both of the underlying objects varied. However, performance with synthetic sounds was impaired if the simulated physical regularities were altered to be unnatural. The results suggest that listeners use internal physical models to separate the acoustic contributions of the objects that interact to create sound.

4pPP15. Investigating the dynamic relationship between mandarin tone perception and production: A study on native thai speakers through perceptual training. Xujia Li (English Dept. & Lang. and Cognit. Neurosci. Lab, Xi'an Jiaotong Univ., No. 28 Xianning Rd. (W), Xi'an, Shaanxi 710049, China, lixujia@stu.xjtu.edu.cn), Kangzhi Liao, and Bing Cheng (English Dept. & Lang. and Cognit. Neurosci. Lab, Xi'an Jiaotong Univ., Xi'an, Shaanxi, China)

This study aims to explore the correlation between perception and production of four Mandarin tones among Thai learners of Chinese before and after one-week intensive perceptual training. Existing research has provided inconsistent findings regarding this correlation, and limited investigations have examined changes during the learning process. The experiment involved twenty Thai speakers who completed a self-developed perceptual test program comprising 120 natural speech stimuli (30 monosyllabic words for each tone). The participants' recordings of these 120 words were presented to four native Chinese speakers in random order, who were required to identify the tone for each word. Before training, a strong and positive correlation between Mandarin tone perception and production was found ($r = 0.89$, $p < 0.001$). However, this correlation significantly weakened after the training, indicating a complex relationship between perception and production in speech learning for second language learners. Further analysis revealed that the perceptual training led to imbalanced development in perception and production for the four tones. Specifically, Tone 2 and Tone 3 did not exhibit significant correlation after training, as production accuracy improved much faster than perception. These findings also have implications for instructional approaches in teaching Mandarin tones. Keywords: Mandarin tone, speech learning, perception-production links

4pPP16. Abstract withdrawn.

4pPP17. Contributions of auditory processing and cognition to the development of frequency discrimination performance during adolescence. Jordin T. Benedict (Speech Pathol. and Audiol., Kent State Univ., 1325 Theatre Dr., Kent, OH 44240, jbened11@kent.edu), Serena A. Sereki, Bruna S. Mussoi, and Julia J. Huyck (Speech Pathol. and Audiol., Kent State Univ., Kent, OH)

Performance on auditory perceptual tasks develops throughout adolescence, likely due to the development of auditory and cognitive regions of the brain. We investigated the contributions of auditory processing and cognition to frequency discrimination. Frequency discrimination thresholds were measured in three conditions using a series of 3AFC tasks. For two conditions, each stimulus was two 15-ms tone pips separated by 100 ms. These stimuli were used both with a 3-down, 1-up procedure (threshold = 79.4%) with the method of constant stimuli (threshold = 66.7%). The third condition used a 130-ms stimulus and a 3-down, 1-up procedure. The auditory electrophysiological acoustic change complex was measured passively using an 800-ms tone that changed from 1000 Hz to a lower frequency. Cognitive testing was also administered. Preliminary results suggest thresholds correlate between conditions and performance improves with longer stimuli. Adolescents (10-17 years) performed worse than young adults (18-23) on only the adaptive-tracking conditions. Cognition contributed more to these conditions than the condition using the method of constant stimuli, suggesting that age differences on adaptive conditions are caused by cognitive development. Adolescents had larger acoustic change complex responses to the frequency contrasts. Behavioral data did not correlate with the electrophysiological data. [Funded by NIDCD].

4pPP18. Musical timbre with varied amplitude envelope improves efficacy in auditory alarms. Andres E. Elizondo Lopez (Psych., Neurosci. & Behaviour, McMaster Univ., 1280 Main St. West, Hamilton, ON L8S 4M2, Canada, elizonda@mcmaster.ca), Joseph J. Schlesinger (Vanderbilt Univ., Nashville, TN), and Michael Schutz (Psych., Neurosci. & Behaviour, McMaster Univ., Hamilton, ON, Canada)

Current alarm standards used in safety critical environments (e.g., medical alarms used in hospitals) suffer from a myriad of complications with detectability, annoyance, and alarm fatigue affecting the wellbeing of patients and staff. To a large extent, these are based on the same simplistic, temporally invariant tones. Here we explore how insights from the acoustic properties of the musical triangle can aid in detection, reducing overall levels hence reducing annoyance ratings. Two tones are used, (1) a standard tone similar to those used in current medical devices, and (2) a tone synthesized based on the spectral-temporal structure of a concert triangle. We conducted a detection experiment where participants indicated if the auditory stimulus is heard when presented a range of signal-to-noise (SNR) ratios and a two-alternative force choice task to measure annoyance ratings. Although reductions in SNRs reduced detectability for the standard tone, similar deductions had no meaningful effect on detectability of tones modeled off the musical triangle. Crucially, we identified a number of triangle inspired tones which are both less annoying and more detectable than standard tones. This suggests that these more complex sounds can reduce annoyance without harming detection, offering useful insight to medical device sound design.

4pPP19. Characterizing the primary resonator in the cochlea. Wei-Ching Lin (Dept. of Mech. Eng., Univ. of Rochester, Mech. Eng., Rochester, NY 14627, wlin28@UR.Rochester.edu), Anes Macic, and Jong-Hoon Nam (Mech. Eng., Univ. of Rochester, Rochester, NY)

The cochlear traveling waves are explained by a bank of independent resonators coupled longitudinally by lymphatic fluids. Many cochlear models require at least two resonators to account for observed responses. To investigate the resonators in the cochlea, we used high-resolution optical coherence tomography to measure 2-D vibration patterns of the organ of Corti in acutely excised cochleae from young Mongolian gerbils. The excised tissues were acoustically stimulated. The transverse and radial vibrations of the basilar membrane (BM) and the tectorial membrane (TM) were obtained over their radial span. The BM vibrated from the primary to a higher mode transversely as the stimulating frequency increased. The higher-order mode appeared near the best frequency (BF) of the measured location. Meanwhile, the TM showed no sign of a mode transition up to 1 octave above the BF in radial or transverse vibrating patterns. Within the physiological frequency range, the BM exhibits the characteristic behavior of a resonator. In contrast, the TM does not. Our results suggest that the TM as the second resonator is not the universal mechanism across the entire cochlea.

4pPP20. Acoustic indicators of voice quality in the context of social support. Luisa E. Hernández Melo (Integrated Program in Neurosci., McGill Univ., 3495 University St., Montreal, QC H3A2A8, Canada, luisa.hernandezmelo@mail.mcgill.ca) and Marc D. Pell (School of Commun. Sci. and Disord., McGill Univ., Montreal, QC, Canada)

Is social support communicated throught the subtle, yet powerful, acoustic variations in speech? This study attempted to answer this question by testing whether acoustic parameters vary when expressing social support. Participants underwent an experiment in which they watched video testimonials of a woman describing either a neutral subject or a sensitive, emotionally-charged experience. After this, participants provided voice messages to the person appearing in the testimony. Employing the openSMILE toolkit, we extracted from these speech responses the Geneva Minimalistic Acoustic Parameter Set (GeMAPS), a set of emotion-related acoustic features. Our investigation reveals an acoustic profile characteristic of supportive speech, distinguished by changes in the Alpha ratio, spectral slope, and the Hammarberg index — parameters representing the high-frequency content and spectral balance. These acoustic differences not only help to differentiate supportive utterances but also characterize its voice quality, thereby enhancing the emotional richness of this affective stance. Our research findings have potential applications in therapeutic and communication settings and open avenues for further exploration in speech science.

4pPP21. The contributions of acoustic and non-acoustic factors to spatial release from masking. Morgan Barkhouse (Speech-Lang. Pathol. and Audiol., Towson Univ., Towson University, Towson, MD 21252, mbarkh1@students.towson.edu), Sadie O'Neill, Chhayakanta Patro, and Nirmal Kumar Srinivasan (Speech-Lang. Pathol. and Audiol., Towson Univ., Towson, MD)

The ability to identify speech at worse signal-to-noise ratios when the maskers are spatially separated compared to when they are colocated is called spatial release from masking (SRM) and is thought to be a combination of monaural and binaural advantages arising from spatially separating the target from the maskers. Also, a host of other cognitive skills including attention, working memory, and executive function that support listeners' ability to segregate, track and attend to a "target" signal while tuning out other unwanted signals contributes to understanding speech better in complex environments. Previous research with young normal hearing listeners from our lab indicated that SRM was predicted by an individual's ability to use binaural cues (ITD thresholds), switch attention between two streams of information (trail making test) and ability to suppress irrelevant information (flanker task). Here, we present data from older listeners with varying hearing abilities on acoustic (envITD sensitivity) and non-acoustic tasks measuring attention (auditory and visual single and dual task), processing (trail making task), executive control (flanker task) and speech in noise tests (spatial release from masking using coordinate Response Measure sentences). The relationship between these acoustic and non-acoustic factors to speech understanding in complex listening environments will be discussed.

4pPP22. Spatial release from masking with simulated hybrid cochlear implant speech. Bailey Borkowski (Speech Lang. Pathol. and Audiol., Towson Univ., 8000 York Rd., Towson, MD 21252, bborkowski@students.towson.edu), Morgan Barkhouse (Speech Lang. Pathol. and Audiol., Towson Univ., Towson, MD), Chhayakanta Patro (Speech Lang. Pathol. and Audiol., Towson Univ., Towson, MD), and Nirmal Kumar Srinivasan (Speech Lang. Pathol. and Audiol., Towson Univ., Towson, MD)

Spatial release from masking (SRM) is the improvement in speech intelligibility when the masking signals are spatially separated from the target signal. Cochlear implant (CI) users utilizing electric-only (E-Only) stimulation had poorer speech recognition than CI users utilizing electric-acoustic stimulation (EAS). Previous research investigating SRM on hybrid CI users big spatial separations ($\pm 90^\circ$) between the target and the maskers which were unrealistic in conversational settings. Here, we present SRM data using natural speech, simulated CI speech, and simulated EAS speech from young, normal hearing listeners for smaller, yet realistic, spatial separations between the target and the maskers. An eight-channel noise-excited vocoder was used to simulate cochlear implant processing and low-frequency filtering was used to simulate residual low-frequency hearing. Five spatial configurations were used: (colocated (target and the two maskers presented from 0° azimuth) and one of four spatially separated conditions (target at 0°, symmetrical maskers at $\pm 5^\circ$, $\pm 10^\circ$, $\pm 15^\circ$, or $\pm 30^\circ$). Initial analysis of the data revealed that the listeners had significantly poorer speech identification thresholds for the simulated CI speech when compared to EAS speech. The individual variability and the relationship between the speech identification thresholds in the three conditions will be discussed.

4pPP23. Spatial release from masking and pupillometry. Rebecka Livingstone (Speech-Lang. Pathol. and Audiol., Towson Univ., Speech-Lang. Pathol. and Audiol., 8000 York Rd., Towson, MD 21252, rliving1@students.towson.edu), Chhayakanta Patro, and Nirmal Kumar Srinivasan (Speech-Lang. Pathol. and Audiol., Towson Univ., Towson, MD)

Speech perception in complex listening environments is driven by both the auditory factors and the cognitive abilities. There are various ways in which cognitive processing while perceiving speech could be measured. In the present study, we applied pupillometry to assess cognitive processing load resulting due to various demands imposed by the presented speech. PupilCore eye glasses were used to capture pupil dilation while doing two tasks: 1. Speech recognition task when the target and the maskers were either colocated or spatially separated, and 2. a localization task where the listeners had to discriminate whether the two sounds originated form the same location or different locations. Initial data analysis revealed different patterns of pupil dilations for the collocated and spatially separated conditions. For the collocated condition, the listeneing effort had a u-shaped pattern as the target-to-masker increased whereas the listening effort linearly increased as the target-to-masker ration increased for the spatially separated condition. Also, the pupil dilation decreased as the localization task became easier. The intricate relationship between listening effort and listening environment will be discussed in detail.

4pPP24. Relationship between attention and spatial processing abilities. Sadie O'Neill (Speech-Lang. Pathol. and Audiol., Towson Univ., 8000 York Rd., Towson, MD 21252, soneil7@students.towson.edu), Morgan Barkhouse, Chhayakanta Patro, and Nirmal Kumar Srinivasan (Speech-Lang. Pathol. and Audiol., Towson Univ., Towson, MD)

Reduction in an individual's hearing ability and their working memory (WM) capacity are hypothesized to be two of the major contributors to the decline in listener performance seen in complex listening environments. Generally, the improvement in speech intelligibility that may occur when a target is spatially separated from competing talkers is quantified as spatial release from masking (SRM). The goal of this study was to estimate working memory capacity based on a divided attention version of the classic spatial release from masking task (the temporal overlap task) and the classic abbreviated reading span task (aRST). Temporal overlap threshold was estimated by adaptively varying the maximum amount of temporal overlap of the signals where a listener could still correctly identify the speech source presented directly ahead of the listener when the speech material was either colocated or spatially separated. Initial analyses of the data revealed a strong relationship between SRM and temporal overlap thresholds, suggesting that SRM is driven by the listeners' ability to modulate their attentional mechanisms. The relationship between SRM, temporal overlap thresholds, and reading span measures will also be discussed.

4pPP25. Comparison of spectral resolution and spectro-temporal modulation sensitivity well above and near the hearing threshold. Ryan Hildebrandt (Commun. Sci. and Disord., Univ. of South Florida, 4202 E Fowler Ave. PCD 1017, Tampa, FL 33620, rhildebrandt@usf.edu), Erol J. Ozmeral (Commun. Sci. and Disord., Univ. of South Florida, Tampa, FL), Stefan Klockgether, Julia Rehmann (R&D, Sonova AG, Stäfa, Zürich, Switzerland), Matthias Keller (R&D, Sonova AG, Staefa, Switzerland), and David A. Eddins (Commun. Sci. & Disord., Univ. of Central Florida, Tampa, FL)

Spectral resolution and spectro-temporal modulation (STM) sensitivity both show a correlation with speech reception thresholds in noise. Both are also sensitive to the difficulties resulting from sensorineural hearing loss; however, the relationship between the two is not fully understood. The

present study investigated the potential relationship between spectral resolution and STM sensitivity and the impact of presentation level in young, normal hearing listeners. Four noise carriers were tested with the two broader bandwidths and two 1-octave wide bandwidths: 200–6400 Hz, 1250–5000 Hz, 1250–2500 Hz, & 2500–5000 Hz. Presentation levels were 70 dB SPL and 20 dB SL relative to the individual's pure-tone frequency-shaped noise carrier threshold. Spectral resolution was tested using a phase reversal paradigm, also known as a spectral ripple discrimination task, with thresholds measured in cycles per octave. Spectral, temporal, and STM sensitivity were each tested using a depth detection task, with thresholds measured in dB. On a test-by-test case, results were aligned with past studies that included a single test. The results show that both sensitivities are distinct at a low presentation level close to individual hearing thresholds, and a mixed relationship between both measures indicate that spectral resolution may only be one contributor to STM sensitivity.

4pPP26. Post-auricular orientation of auditory attention in sound field versus virtual sound space. Melina Markotjohn (Audiol., Dalhousie Univ., 816-1715 Lower Water St., Halifax, NS B3J0J4, Canada, ml314650@dal.ca) and Steven Aiken (Audiol., Dalhousie Univ., Halifax, NS, Canada)

The post-auricular muscle in many species' changes orientation of external ears to improve hearing for biologically relevant sounds. The muscle exists in humans but cannot similarly change direction of their ears. Objectives focused on measuring activity of muscle during speech-in-noise task where orientations of speaker and noise were controlled experimentally to determine how signal-to-noise varies as function of presentation mode and azimuth (target speech and noise co-localized, 45°, or spatially separated, 135° and 45°, respectively). It was hypothesized that activity would be elicited in same proportion of subjects when evoked via earphones compared to speakers; there would be no significant differences in magnitude between conditions; maximum engagement would be observed with speech 135°, noise 45°. Activity was recorded with electrodes affixed around ears, outer canthi and neck, while listeners completed a spatialized listening test (locations of speaker/noise controlled experimentally). There was significant main effect of channel; significant interaction between presentation mode and channel; no significant differences between presentation modes for other muscles; no significant effect of azimuth. Engagement in virtual sound-space suggests that muscle activation occurs consequently of spatially directed attention, even when changes in pinna orientation are unlikely to have effect on sound heard.

4pPP27. Exploring the influence of bilingual experience on speech-in-competition measures. Katia Padilla-Bustos (Lab. of Cognit. and Clinical Neurophysiology, National Inst. of Neurology and Neurosurgery, Av. Insurgentes Sur 3877, La Fama, Tlalpan, Ciudad de México, CDMX, Mexico City 14269, Mexico, katia208padilla@gmail.com), Frederick J. Gallun (Oregon Hearing Res. Ctr., Oregon Health and Sci. Univ., Portland, OR), Aaron R. Seitz (Dept. of Psych. and Ctr. for Cognit. and Brain Health, Northeastern Univ., Boston, MA), Rodolfo Solís-Vivanco (Lab. of Cognit. and Clinical Neurophysiology, National Inst. of Neurology and Neurosurgery, Mexico, Mexico), and Esteban Sebastian Lelo de Larrea-Mancera (Dept. of Psych. and Ctr. for Cognit. and Brain Health, Northeastern Univ., Boston, MA)

Speech-in-Competition (SIC) tasks mimic real-world environments by including background noise and multi-speaker scenarios. However, few SIC tasks have been translated into Spanish and validated. Moreover, the influence of bilingual experience on SIC measures is not well understood. This study aimed to investigate how bilingual experience influences performance on speech-in-competition tasks conducted in Spanish versus English. We tested fifty-six Mexican undergraduate students whose native language was Spanish who self-rated English dominance on a 10-point scale ($M = 5.8$, $SD = 2.3$). Participants performed better on the Spanish version of a spatial release from masking compared to the English version but had similar performance in a digits-in-noise test. The relationship between performances and bilingual experience was tested by creating a linguistic profile based on five dimensions of the second language. Statistically significant medium-size correlations were found between performance and the linguistic dimensions of status, proficiency, and history, but not with demand of use, or stability. This indication that the influence of bilingual experience on SIC measures can be substantial suggests that SIC tasks may be inappropriate for individuals whose native language is not English. This also suggests that other linguistically diverse populations may need specialized SIC measures as well.

4pPP28. Cochlear hair bundle dynamics: modeling calcium effects and row-wise interactions. Varun Goyal (Mech. Eng., Univ. of Michigan, Ann Arbor, 3632, G.G.B. Labs., 2350 Hayward St., Ann Arbor, MI 48109, varungo@umich.edu) and Karl Grosh (Mechancial Eng., Univ. of Michigan, Ann Arbor, MI)

External vibrations cause the eardrum to oscillate, resulting in the excitation of the sensory structures of the cochlea. In this talk, we focus on the hair bundles (HBs) of the outer hair cells situated inside the cochlea. These bundles, protruding from the cells, convert mechanical motion into electrical currents. Experimental observations reveal that the calcium concentration inside the stereocilia (hair-like structures that comprise the HB) influences current adaptation. This regulation impacts shifts of the curve, sensitivity, and active range of bundles. We aim to mechanistically understand the intracellular calcium effects by adjusting the adaptation complex in our HB model. We modified model parameters to match experimental data on current, bundle displacement, and shift trends at different calcium concentrations. This involved increasing the adaptation stiffness, stall force, stereocilia pivot, and gating stiffness. A stiffer adaptation spring reduces ion-channel reclosure, affecting the steady-state response. A higher stall force lessens the effective force on the adaptation complex, replicating experimental observations. Unlike the other properties, pivot, and gating stiffness likely do not depend on calcium concentration. Therefore, we will conduct error minimization analyses to identify adaptation complex properties while maintaining constant pivot and gating stiffness across calcium concentrations. Work supported by NIH grant NIH-NIDCD-R01 04084.

4pPP29. The frequency and level dependence of across-frequency binaural interference for interaural time differences. Allison Choi (Hearing and Speech Sci., Univ. of Maryland-College Park, 0119 Lefrak Hall, College Park, MD, achoi123@umd.edu), Anhelina Bilokon (Hearing and Speech Sci., Univ. of Maryland-College Park, College Park, MD), and Matthew J. Goupell (Hearing and Speech Sci., Univ. of Maryland-College Park, College Park, MD)

Binaural hearing enables listeners to localize sound in the horizontal plane, even when dealing with intricate speech-like signals characterized by distributed energy across a broad range of frequencies and varying sound pressure levels. Low-frequency interaural time differences (ITDs) are the most important cue for human horizontal-plane sound localization. Across-frequency binaural interference tasks (where changes in ITD discrimination thresholds of a target signal are measured in the presence of a remote interferer signal) can be used to measure how a listener weighs different frequency regions for ITD. While it is known that low-frequency (0.5 kHz) targets experience less binaural interference from higher-frequency (4 kHz) interferers than vice versa, the impact of the relative level of the two signals has remained mostly unexplored. ITD discrimination thresholds were measured in normal-hearing listeners with target/interferer intensities of 35, 55, and 75 dB-A and frequencies of 0.5, 4, and 8 kHz. Besides the expected frequency effects, data to date show binaural interference increases with increasing interferer level. These results help better understand how sound level variation and more realistic complex sounds like speech are localized.

4pPP30. Finding the ratio of perceived duration between tones with flat and percussive amplitude envelopes. Connor Wessel (Psych., Neurosci., and Behaviour, McMaster Univ., 187 Oak Ave., Hamilton, ON L8L5M9, Canada, wesselc@mcmaster.ca), Cindy Zhang (Faculty of Health Sci., McMaster Univ., Hamilton, ON, Canada), and Michael Schutz (School of the Arts, McMaster Univ., Hamilton, ON, Canada)

The extensive literature on duration assessment generally uses tones with clear onsets and offsets. However, simplistic sounds can fail to evoke the same processes used when listening to sounds with time varying amplitude envelopes (Schutz & Gillard, 2020). To facilitate future research on the

duration assessment of time varying tones, here we explore the ratio at which constant amplitude 'flat' and varying amplitude 'percussive' tones are perceived as the same duration. We will use an adaptive staircase procedure, in which flat and percussive tones are presented in pairs and participants state which tone sounded longer in duration. Each response changes the duration difference on subsequent trials and continues until responses converge around a specific point, with convergence defined as four consecutive reversals in response direction. One instance of these trials constitutes a single staircase; we will then calculate the millisecond point of subjective equality between flat and percussive tones by finding the average point of convergence between multiple, interleaved staircases. Beyond providing guidance on stimulus durations for studies comparing amplitude envelope, we aim to shed light on the process of duration assessment in sounds with time-varying amplitude envelopes.

THURSDAY AFTERNOON, 16 MAY 2024 ROOM 201, 1:00 P.M. TO 3:05 P.M.

Session 4pSA

Structural Acoustics and Vibration, Education in Acoustics and Physical Acoustics: Structural Acoustics and Vibrations Tutorial

Anthony L. Bonomo, Cochair
Naval Surface Warfare Center, Carderock Division, 9500 MacArthur Blvd, West Bethesda, MD 20817

Peter Kerrian, Cochair
ATA Engineering Inc., 13290 Evening Creek Dr. S, San Diego, CA 92128

Stephanie Konarski, Cochair
Johns Hopkins University Applied Physics Laboratory, 11100 Johns Hopkins Road, Laurel, MD 20723

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

Chair's Introduction—1:00

Invited Paper

1:05

4pSA1. How to model vibration damping in complex materials and structures. James G. McDaniel (Mech. Eng., Boston Univ., Boston University, Dept. of Mech. Eng., Boston, MA 02215, jgm@bu.edu)

Damping is perhaps the most challenging aspect of any model where damping is relevant. For example, an accurate damping model is often important when modeling vibration of metamaterials, vehicles, and machinery. Mass density is easily and accurately calculated from simple measurements of weight and volume. Elastic moduli are easily and accurately calculated from simple measurements of stress and strain. Large databases of mass density and elastic moduli are readily available and accurate for most materials. But damping is elusive. Many models have been proposed in many different research communities. Published numerical values for the parameters are in short supply. But the choice of damping model may dramatically affect the predicted response. The damping model is likely to make the difference between a useful result and a useless result. This presentation seeks to collect and summarize knowledge related to damping to help us model damping more accurately and more efficiently. The three most common models of damping are reviewed. These are material damping, viscous damping, and Coulomb damping. For each, an analysis is presented that begins with the fundamental assumptions and ends with the inclusion of the damping mechanism in finite element models. [Work supported by ONR under Grant N00014-22-1-2785.]

Session 4pSC

Speech Communication: Speech Production Poster Session

Kelly Berkson, Chair
Linguistics, Indiana University, Bloomington, IN

All posters will be on display from 1:15 p.m. to 5:15 p.m. Authors of odd-numbered abstracts will be at their posters from 1:15 p.m. to 3:15 p.m. and authors of even-numbered abstracts will be at their posters from 3:15 p.m. to 5:15 p.m.

Contributed Papers

4pSC1. Location of constriction in velar sounds in French. Md Jahurul Islam (Linguist., The Univ. of BC, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, jahurul.islam741@gmail.com), Victor Wong, Dayeon Choi, and Bryan Gick (Linguist., The Univ. of BC, Vancouver, BC, Canada)

This study examines the constriction location (CL) of the velar stop [k] in French. While previous studies have investigated how vocalic contexts influence the CL of velar stops [Liker & Gibbon (2008) *Clin. Ling. & Phon. 22*(2); Tabain (2000) *JPhon 28*(2)], few investigated high-level changes in CL. Building on our prior work [Islam & Gick (2023) *JASA* 154], which showed an unexpected palato-velar articulation of [k] followed by [a] compared with [w] followed by [a], we tested whether this [k]-palatalizing is universal or specific to [a]. Using a French MRI speech corpus [Isaieva *et al.*, *Scientific Data* 8], we measured constriction in [k] before [i], [ɛ], [o], [u], and [a]. MRI video frames were manually traced to mark the upper surface of the tongue and the lower surface of the hard palate, resulting in two contours. Using Euclidean distance, the location of the constriction was identified as the narrowest point and distance, respectively, between the two contours. A Python script that measured Euclidean distance from traced MRI frames calculated these distances from the MRI frames. Results indicate a frontal shift in [k] across all contexts, indicating a general fronting articulation of [k] in French speech.

4pSC2. Gender-related variation in /s/ and /ʃ/ in child-directed speech. Eugene Wong (Univ. of Minnesota, 164 Pillsbury Dr. SE, University of Minnesota, Minneapolis, MN 55455, wong0703@umn.edu) and Benjamin Munson (Univ. of Minnesota, Minneapolis, MN)

Previous research showed that 3-year-old children assigned male at birth begin to produce /s/ differently from children assigned female at birth (Munson, Koeppe & Lackas, 2022). This mirrors the differences in /s/ found between cisgender men and women in American English (e.g., Jongman, Wayland & Wong, 2000; Munson, McDonald, DeBoe, & White, 2006). The current study investigates whether children are exposed to the gender-marking /s/ variants through child-directed speech, a hypothesis suggested by Foulkes, Docherty, and Watt (2005). We collected speech samples from 36 mothers of children aged 2 to 3 years through a story-reading task. Mothers read a male-themed story and a female-themed story to their children. We also measured mothers' attitude towards their children's gendered behavior. Spectral characteristics of /s/ and /ʃ/ were measured. As there was no evidence that /ʃ/ marks gender, the acoustic characteristics of /ʃ/ is measured to control for the potential influence of anatomical variation on fricative acoustics. Analysis is ongoing, and will examine whether the acoustic characteristics of /s/ in male- and female-themed stories resemble the male- and female-typed /s/ variants found in previous studies. The results of this study will inform models of the development of gendered speech in children.

4pSC3. Acoustic cues in perception of reduced speech. Song Yi Kim (Dept. of Linguist., The Univ. of Arizona, 3201 E Fort Lowell Rd., Tucson, AZ 85716, songyikim@arizona.edu) and Natasha Warner (Dept. of Linguist., The Univ. of Arizona, Tucson, AZ)

In conversational speech, sounds undergo reduction and deletion, which challenges listeners to decode reduced forms using various cues. Previous research argues for the predominance of acoustic cues in a study of listeners' perception of reduced speech such as "he's" vs. "he was", which can both be realized [ĕz]. Analyzing data from a previous perception experiment (Warner *et al.*, Brain Sci., 2022), this study explores what types of acoustic cues listeners use to process reduced speech. Ambiguous [ĕz] with low second formant (F2) or long duration might be perceived as past due to an additional consonant (/w/) and potential contractions (low F2 as a cue to /w/). Low F2 is expected to result in smaller Bark F2-F1 and larger Bark F3-F2. Current study tests whether the stimuli show acoustic differences in the predicted direction and whether the formant and duration measures correlate with listeners' identification of tense. It was found that there are no significant acoustic differences between past and present forms due to reduction, but that listeners significantly associated smaller Bark F2-F1 with past tense in singular verbs and longer duration with past tense in plural verbs. This confirms listeners' use of the predicted acoustic cues in perceiving reduced speech.

4pSC4. Bidirectional C-to-V coarticulation across syllable and word boundaries. Scarlet Wan Yee Li (Dept. of Linguist., Univ. of Ottawa, Hamelin Hall, Rm. 401, 70 Laurier Ave. East, Ottawa, ON K1N 6N5, Canada, wli240@uottawa.ca) and Suzy Ahn (Dept. of Linguist., Univ. of Ottawa, Ottawa, ON, Canada)

Extensive work has explored the anticipatory or carryover coarticulatory effects of consonants on F2 of adjacent vowels using locus equation. However, studies examining bidirectional effects remain scarce. This study examines how voicing and place of articulation (PoA) of consonants affect bidirectional C-to-V coarticulation across syllable and word boundaries. Recordings of /$C_{;1}V_{;1}.C_{;2}V_{;2}$/, /$C_{;1}V_{;1}\#C_{;2}V_{;2}$/ and /$C_{;1}V_{;1}C_{;2}\#V_{;2}$t/ sequences from native speakers of Canadian English were analyzed. F0, F1, and F2 were measured at the onset and offset of the target vowels (/i ɑ/) to compare the coarticulatory effects caused by the adjacent consonants with different voicing and PoA (/p b t d g k/). Preliminary results ($N = 6$) indicate that coarticulatory effects vary based on voicing and PoA of consonants. Voiceless consonants exhibited a greater effect on f0 bidirectionally, but smaller effects on F1 and F2. Regarding PoA, only the effect of F2 in anticipatory contexts was dependent on different PoA. The findings also suggest that bidirectional coarticulatory effects are found across both syllable and word boundaries. Within and across these boundaries, larger anticipatory effects were found on f0, while the carryover effect on F1 was more robust. Concerning F2, differences between coarticulatory effects were only evident in voiceless and velar contexts.

4pSC5. Phonetics characteristics of seoul korean tense stops. Harim Kwon (English Lang. and Lit., Seoul National Univ., 4400 University Dr., 3E4, Fairfax, VA 22030, harimkwon@snu.ac.kr) and Suzy Ahn (Dept. of Linguist., Univ. of Ottawa, Ottawa, ON, Canada)

This study investigates Seoul Korean tense stops, asking whether underlying intervocalic tense stops in /at*a ap*a ak*a/ share phonetic characteristics with those derived from a sequence of two lax stops by applying the post-obstruent tensing rule /at.ta ap.pa ak.ka/ → [at*a ap*a ak*a]. To examine whether the tense stops show articulatory fortition compared to lax stops, and if so, whether the underlying and derived tense stops differ from each other, tongue configuration and closure duration of tense stops in [at*a ap*a ak*a] that are underlyingly either /at*a ap*a ak*a/ or /at.ta ap.pa ak.ka/ are compared with those of lax stops in [ata apa aka]. Ultrasound tongue imaging data from six native speakers of Seoul Korean reveal indistinguishable tongue configurations among underlying tense stops, tense stops derived from underlying lax sequences, and lax stops. Both derived and underlying tense stops have longer closure duration than lax stops, but the two surface tense stops did not significantly differ in their closure duration. Taking the articulatory and acoustic evidence together, underlying and derived tense stops in Seoul Korean are phonetically indistinguishable from each other although both have longer closure durations than singleton lax stops. The findings further suggest that Seoul Korean may not use tongue configuration as an articulatory strategy for fortition for either underlying or derived tense stops.

4pSC6. Abstract withdrawn.

4pSC7. Dialect effects in hesitations in older children's spontaneous speech. Ewa Jacewicz (Speech and Hearing Sci., The Ohio State Univ., 1070 Carmack Rd., 110 Pressey Hall, Columbus, OH 43210, jacewicz.1@osu.edu), Emma Cronin (Speech and Hearing Sci., The Ohio State Univ., Columbus, OH), Quinn Baumgartner (Speech and Hearing Sci., The Ohio State Univ., Columbus, OH), and Robert A. Fox (Speech and Hearing Sci., The Ohio State Univ., Columbus, OH)

Natural speech is interspersed with pauses, word repetitions, phrase repairs, and other disfluencies. These hesitations, reflecting speakers' performance delays including word fillers, self-corrections, and filled or silent pauses are typical traits of spontaneous speech. Acoustic characteristics of hesitations have been studied primarily in adults, providing evidence that at least some hesitation types exhibit speaker-inherent strategies that are language specific. It is unknown whether older children, whose abilities to engage in conversation approach those of adults, utilize hesitations that reflect the adult model prevalent in a given speech community. Here, we examine spontaneous conversations of American English-speaking 8–12 years-old children growing up in the South (Western North Carolina) and in the Midwest (Central Ohio). We seek to determine whether hesitation types in their speech are shaped by features of their regional dialect. Acoustic analyzes include breathing pattern, phrase length, articulation rate, silent and filled pauses, filler vocalizations, and repair strategies. Preliminary findings indicate that dialect does have an effect on the nature and frequency of hesitations. The work is currently ongoing, and the results will be discussed.

4pSC8. Where does the gamma go?: Acoustics of the non-labialized voiced velar approximant in Tlingit. Amanda Cardoso (Linguist., Univ. of BC, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, amanda.cardoso@ubc.ca), Simone Brown, Omar Lahlou, Ella Paulin, and James A. Crippen (Linguist., McGill Univ., Montreal, QC, Canada)

The acoustic properties of the voiced velar approximant [ɰ] are poorly characterized, partly from understudy of relevant languages and partly from uncertainty about the sound itself. We investigate this sound in Tlingit, a critically endangered indigenous language of northwestern North America, where [ɰ] contrasts with labialized [w] and palatal [j] in some dialects but not in others where it was lost through sound change (reductive primary split with merge). We use audio recordings of spoken narratives spanning much of the 20th century from speakers with and without [ɰ] to characterize this sound both dialectally and diachronically. We measure formants (f1–f3) to identify place information, amplitude for oral aperture, and harmonic-to-noise ratio for manner. Existing descriptions and phonological patterns predict that [ɰ] is an approximant and not a fricative so we compare against known approximants for differences in turbulence and spectral properties. We also compare against labialized sounds including contrastive [w] for absence of lowered f3 to exclude lip rounding in [ɰ]. We argue that it is the intersection of these acoustic cues that distinguishes [ɰ] from similar sounds, but that these cues are unstable so that dialectal sound change arises from misperception in perturbations of their production.

4pSC9. Signaling phrasal prosody through multimodal coordination. Yoonjeong Lee (Linguist., Univ. of Michigan, 611 Tappan St., 421 Lorch Hall, Ann Arbor, MI 48109-1220, yoonjeol@umich.edu), Alison McGrath, Vikas Tatineni (Linguist., Univ. of Michigan, Ann Arbor, MI), and Jelena Krivokapić (Linguist., Univ. of Michigan, Ann Arbor, MI)

This study examines the relationship between prosodic structure and non-referential co-speech gestures in Seoul Korean, a language that lacks lexical stress and uses phrasing to mark prominence. Acknowledging this crucial role of phrasing in Seoul Korean and considering the importance of prominence in orchestrating speech and co-speech gestures across diverse languages, we predict inter-articulator coordination through prosodic boundaries in Seoul Korean. Eight Seoul Korean speakers read a children's story while their speech, manual, and head gestures were recorded using electromagnetic articulography and camcorders. Results uncover systematic inter-articulator coordination patterns, deviating from those observed in languages that use pitch accents to mark prominence. Manual gestures serve as global phrase markers, synchronizing with both phrase-edge segment and tone gestures, demonstrating substantial stability. The manual gesture onset aligns precisely with the phrase-initial tone gesture onset and constriction gesture target. Similarly, the manual gesture target aligns with both the phrase-final boundary tone target and constriction gesture target. Head gestures, serving as local phrase-edge markers, precisely align with phrase-edge syllables, displaying onset-to-onset and max-to-max inter-articulator coordination patterns. Findings underscore the impact of linguistic structure on coordinating speech and co-speech gestures, elucidating how language-specific prosodic structure is conveyed through multimodal coordination.

4pSC10. Unflappable? Buckeyes say "whatever" to English flapping rules. Sean A. Fulop (Linguist., California State Univ. Fresno, 5245 N Backer Ave. Linguist PB92, Fresno, CA 93740-0001, sfulop@csufresno.edu), Mark P. Ryan (Albany, OR), and Hannah J. Scott (Comput. Sci., Oregon State Univ., Corvallis, OR)

American English dialects generally use a 'flap/tap' consonant in some places where /t/ or /d/ is both indicated in the spelling and used in some other dialects. What environments allow a flap to occur instead of /t/ or /d/? This question has inspired considerable debate in the literature, but a consensus has settled on a lack of stress on the syllable following as being the most critical factor in the occurrence of a flap word-internally. This study investigates the occurrence of flaps in the Buckeye Corpus of American English (40 speakers from Columbus, OH), detailing the frequency of flapping in various phonetic environments. One significant finding is that all speakers produced some flaps word-internally before syllables bearing primary or secondary stress (as in "whatever").

4pSC11. Dynamic auditory-acoustic properties differentiate word-final fricatives in Gitksan. Una Y. Chow (Linguist., Univ. of BC, Totem Field Studios, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, una.chow@ubc.ca) and Molly Babel (Linguist., Univ. of BC, Vancouver, BC, Canada)

Gitksan is an endangered, understudied Tsimshian language spoken in northwestern British Columbia. This study aims to provide an acoustic-phonetic description of Gitksan fricatives. Isolated words containing word-final fricatives were recorded with three Gitksan first-language speakers (2 male, 1 female; age 65+). Postvocalic fricative tokens ([s] = 159, [ɬ] = 123, [ç] = 180, [X^w] = 150, [χ] = 267) were analyzed. Dynamic, spectral measurements of peak frequency in equivalent rectangular bandwidths (peakERB; Reidy, 2016) were estimated from 17 20-*ms* windows centered at equidistant points between the onset and offset of the fricative. The resulting

4p THU. PM

trajectories (windows 3–15) show that the peakERB decreases as the place of articulation shifts further back in the vocal tract (alveolar [s]∼25–31, palatal [ç]∼20–23, uvular [χ]∼17–19; exception: labiovelar [X^w]∼15–16, likely lowered by labialization). Also, the lateral alveolar [ɬ] shows a lower peakERB (∼22–24) than the central alveolar [s]. A random forest model (Breiman, 2001) was trained on 586 tokens in predicting the fricatives with peakERB measures of the 17 windows. The trained model was tested on 293 tokens; it identified [s] most accurately and [ɬ] least accurately. Additionally, the model's relative-importance ranking of the 17 peakERB predictors suggests that the middle third of the peakERB trajectory is robust in differentiating word-final fricatives in Gitksan.

4pSC12. Classification patterns in conversational English fricatives: Between- and within- speaker analyses. Viktor Kharlamov (Florida Atlantic Univ., 777 Glades Rd., CU-97, Ste 280, Boca Raton, FL 33431, vkharlamov@fau.edu), Daniel Brenner (Alameda, CA), and Benjamin V. Tucker (Northern Arizona Univ., Flagstaff, AZ)

This study examines between- and within- speaker patterns in random forest classification models for fricatives in conversational English. Prior investigations on the categorization of fricatives have mostly focused on group-level analyses and careful speech. We use a corpus of sociolinguistic interview speech from Western Canadian English, which represents a more casual speech style, and we compare group-level results to the findings of classification models on individual speakers' productions. The models use 23 different spectral, temporal, and amplitudinal measures to predict phonetic labels of the fricatives. Our results reveal that certain top-ranked predictors at the group level (e.g., spectral peak frequency, segment duration) are also important for fricative classification in the individual models. Other measures (e.g., midpoint kurtosis, RMS) show substantial variability in their relative prominence and play greater roles in individual- versus group-level modeling. We discuss the implications of these findings for our understanding of individual variation in connected speech and what the findings may imply about models of production and perception.

4pSC13. Spectral properties of Quebec French sibilants. Massimo Lipari (Dept. of Linguist., McGill Univ., 1085, Ave. du Docteur-Penfield, Montréal, QC H3A 1A7, Canada, massimo.lipari@mail.mcgill.ca) and Morgan Sonderegger (Dept. of Linguist., McGill Univ., Montreal, QC, Canada)

Most acoustic work on sibilants has focused on English /s/ and /ʃ/, examining spectra of the middle portion of these consonants. Voiced sibilants, which are rarer cross-linguistically (Ohala, 1983), as well as sibilants in other languages, have received less attention; in particular, there has been no previous work on Quebec French. Previous cross-linguistic studies have suggested important differences in sibilant acoustics exist between languages (Gordon *et al*., 2002), including in the dynamics of spectral properties over the course of the segment (Reidy, 2016). This study adds to the literature on the typology of sibilant acoustics by examining the spectral dynamics of sibilants in Quebec French (/s, ʃ, z, ʒ/). ∼28 k word-initial, prevocalic tokens from more than 100 speakers are extracted from a large corpus of parliamentary speech (Milne, 2014). For each token, multitaper spectra (Reidy, 2013) over a 20 ms window are calculated at 17 equidistant points. A variety of acoustic measures, including segment duration and spectral moments, are reported. Preliminary results examining static measures reveal that the anterior sibilants have higher spectral speak (∼5700 Hz vs ∼4600 Hz for posterior ones), as expected; unlike in English, however, /s/ and /ʃ/ have similar average duration.

4pSC14. Abstract withdrawn.

4pSC15. Regional differences in the production of tones in standard mandarin. Sishi Fei (School of Foreign Lang. and Cultures, Nanjing Normal Univ., School of Foreign Lang. and Cultures, Nanjing Normal University, Ninghai Rd., 122, Nanjing, 210097, China, Nanjing, Jiangsu 210097, China, sishi_fei@163.com)

This study investigates the intricate patterns of tonal variations in monosyllabic words across distinct regional accents of Standard Mandarin, with a focus on Shanghai, Guangzhou, and Beijing. Due to the differing tonal systems of the local Chinese dialects in the three cities, speakers from these areas may exhibit distinct patterns in their production of tones in Standard Mandarin, influenced by their native Chinese dialects. Therefore, to capture the nuances of tonal differences, the study extracts the time-normalized f0 values of four tones (i.e., level, rising, dipping, and falling tones) from monosyllabic words in a Mandarin Chinese speech corpus. The overall f0 contours of the four tones within and across the three regional accents are modeled using growth curve analysis. We expect to find that regional accents of Standard Mandarin exhibit significant tonal variations such that the tone shapes of speakers from Shanghai and Guangzhou may significantly deviate from those of Beijing speakers. Specifically, there are variations in contour shape and temporal dynamics, particularly in the quadratic term for dipping tones and the slopes of rising and falling tones. These findings can provide valuable insights into the intricate characteristics of regional tonal variations in Standard Mandarin.

4pSC16. Phonetic adaptation in conversation: The case of Cantonese tone merging. Ivan Fong (Dept. of Linguist., Simon Fraser Univ., Burnaby, BC, Canada, ivan_fong@sfu.ca), Fenqi Wang, Kira Chan, Tyne Johnson-Dhillon, Jadeyn Trasolini (Dept. of Linguist., Simon Fraser Univ., Burnaby, BC, Canada), Dawn Behne (Dept. of Psych., Norwegian Univ. of Sci. and Technol., Trondheim, Norway), Allard Jongman (Dept. of Linguist., The Univ. of Kansas, Lawrence, KS), Joan Sereno, and Yue Wang (Dept. of Linguist., Simon Fraser Univ., Burnaby, BC, Canada)

Phonetic adaptation occurs when one interlocutor adjusts their speech to converge to or diverge from that of their conversation partner to enhance intelligibility. While most research investigates segmental adaptations, our study focuses on suprasegmentals, specifically Cantonese tone merging. Some Cantonese speakers ("mergers") are found to merge certain lexical tones (e.g., mid-level Tone3 and low-level Tone6), which may cause confusions when interacting with non-merger speakers. Previous research has shown that a merger may unmerge a level tone pair (Tone3/Tone6) when shadowing a non-merger. However, still unclear is whether such changes result from automatic acoustic mimicking or reflect goal-oriented adaptations for intelligibility benefits. This study uses an unscripted conversation task involving a merger and a non-merger playing a video game, where productions of merged tones may cause confusions, thus motivating goal-oriented adaptations. Initial acoustic analyses focus on average F0 and F0 taken at 10 points along the contour in target Tone3 and Tone6 productions by mergers. Differences in these values for Tone3 versus Tone6 provide evidence that a merger is unmerging the tone pair. Preliminary results show increasing unmerging trends as the task progresses, suggesting progressive alignment toward a non-merger's productions for intelligibility gains.

4pSC17. Phonetics relating stance-taking in tonal languages-examples from Mandarin Interjections. Wu Siyu (College of Foreign Lang., Nanjing Univ. of Aeronautics and Astronautics, Jiangjun Avenue, Jiangning District, Nanjing, Jiangsu 211100, China, siyuwu@nuaa.edu.cn)

Previous studies on phonetics and stance-taking mainly focused on non-tonal languages. In this study, we found that tone variation could affect the perception of stance information. The study includes a production experiment and a perception experiment which is aimed to evaluate the stance polarity and stance strength. When the tone was nearer to its citation form as its duration got longer, the participants were more likely to identify the stance information as positive or negative. This gives us insight that there may be a express-inference process during the perception of stance-taking.

4pSC18. Examining the time-varying measures of F0 in sarcasm: Wigglines, spaciousness, and contour clustering. Csilla Tatar (Linguist., Univ. of Michigan, 611 Tappan St., Ann Arbor, MI 48109, cstatar@umich.edu), Jonathan R. Brennan, Jelena Krivokapić, and Ezra Keshet (Linguist., Univ. of Michigan, Ann Arbor, MI)

The present project examines the phonetic correlates, specifically those related to F0, of sarcastic speech. In a production study, American English speakers (N = 12) produced identically worded utterance pairs presented in contexts conducive to sarcasm and sincerity. Measures of F0 variability are contrasted; these are wiggliness and spaciousness [Wehrle, Cangemi,

Krüger, & Grice (2018) Proceedings of AISV], F0 range and F0 mean SD. Raw values were entered into by-speaker logistic regression models. Wiggliness and spaciousness together were found to be comparable to F0 mean SD and F0 range in distinguishing sarcasm and sincerity for eight of the speakers. Intonational characteristics were further examined via by-speaker F0 contour clustering [Kaland, 2021]. These by-speaker analyses showed that many speakers produce contours characteristic of sarcasm or sincerity, but that these contours differ by speaker. Further exploration of a subset of nine speakers' data showed that wiggliness and spaciousness alone can capture some of the differences between sincere and sarcastic contour clusters for some speakers. Speaker strategies vary in terms of F0, and sarcastic speech is characterized by reduced wiggliness and spaciousness for some of the speakers.

4pSC19. Abstract withdrawn.

4pSC20. Influence of tone, vowel, and consonant constraints on lexical selection in Cantonese. Minghao Zheng (Dept. of Linguist., Univ. of Florida, 4131 Turlington Hall, Gainesville, FL 32611, minghao.zheng@ufl.edu)

This study explores the impact of tones, vowels, and consonants on lexical selection in Hong Kong Cantonese, employing a word reconstruction paradigm. While prior research in European languages suggested greater mutability in vowels than consonants, the applicability of this hypothesis to tonal languages remained untested. Building on Wiener and Turnbull's (2016) Mandarin Chinese experiment, this study involved ten native speakers in a word reconstruction task with vowel, consonant, and tone substitution conditions, and a free choice condition which allowed participants to change either a vowel, consonant, or tone. Results revealed a preference for altering tone over vowels or consonants when transforming fake words into real ones, indicating that in Cantonese, tone carries the lowest information load and constrains lexical access least tightly. Moreover, participants exhibited greater accuracy in tone substitution compared to consonants and vowels. Contrary to the universal intrinsic vowel mutability hypothesis, this study suggests that, in tonal languages like Cantonese, vowels are less mutable than tones or consonants, carry the highest information load, and constrain lexical access most tightly. Vowels are more mutable only in the absence of lexically contrastive suprasegmental information. These findings contribute to understanding lexical selection in tonal languages, challenging prevailing notions regarding vowel mutability.

4pSC21. Investigating post-focus compression in code-switching: A comparative study of Twain Mandarin and English. Grace Kuo (Foreign Lang. and Literatures, National Taiwan Univ., 1 Section 4, Roosevelt Rd., Taipei, Taipei City 106, Taiwan, graciakuo@ntu.edu.tw)

On-focus pitch range expansion is a documented phenomenon in both English and Mandarin Chinese. However, the presence of post-focus compression (PFC) contrasts with its absence in Taiwan Mandarin, a regional variant of Mandarin Chinese. This discrepancy becomes particularly intriguing in the context of code-switching, which provides a unique lens to observe the interplay between these two languages. This study probes the manifestation and cognitive processing of PFC in code-switching constructs, employing a bilingual paradigm between English and Taiwan Mandarin. Through a carefully designed production experiment, this research addresses two questions: (a) Is PFC evidence in the code-switching utterances of simultaneous and early bilinguals? (b) Is there an observable correlation between the emergence of PRC in linguistic output and the language proficiency of late bilinguals, potentially indicating a transfer effect from Taiwan Mandarin to English? Anticipated results may substantiate the presence of linguistic transfer in the code-switching sentences of bilingual individuals of diverse English proficiencies. Furthermore, the implications of these findings may extend to pedagogical strategies, offering insights into the deployment of code-switching within multilingual educational settings.

4pSC22. The [s – > ʃ] sound change in /str-/ contexts: how temporal changes in the acoustic spectrum affect fricative percept. Christine H. Shadle (Yale Child Study Ctr., Yale Univ., 300 George St Ste. 900, New Haven, CT 06511, shadle@haskins.yale.edu), Laura L. Koenig, and Weirong Chen (Yale Child Study Ctr., Yale Univ., New Haven, CT)

Studies have documented a sound change in some English dialects whereby /s/ in the context /strV/ surfaces as [ʃtrV]. This can be interpreted as /s/ rounding for the rhotic being re-analyzed as [ʃ]. We recently measured $F_{;M}$, the frequency of the main peak in mid-fricative spectra, in seven adults producing words with /s/ and /ʃ/ across phonetic contexts and compared the acoustics to perceptual rankings on an [ʃ] to [s] scale. One speaker exhibited a perceived sound change; two, partial sound changes. It was shown that $F_{;M}$ was correlated with the perceptual rankings for /strV/ words. However, in all speakers, lowered $F_{;M}$ for /s/ in labial contexts did not affect percepts. The time course of lip rounding and tongue-tip retraction in /strV/ could help differentiate coarticulation versus a phonemic change. Interestingly, however, X-Ray Microbeam data [Iskarous, K. *et al*. (2011) Articulatory-acoustic kinematics of /s/, J. Acoust. Soc. Am. 129, 944-954] showed that /s/ constriction location was constant over time except in /str-/ context. This study will assess the amount and timing of $F_{;M}$ lowering, its perceptual consequences and likely articulatory causes, to gain more insight into the [strV] – > [ʃtrV] sound change.

4pSC23. Protective face masks affect clearly produced diphthongs by L1 and L2 English talkers. Baorian Nuchged (Dept. of Linguist., The Univ. of Texas at Austin, Atton Hall, 305 E 23rd St Ste. 4.304, Austin, TX 78712, baorian@utexas.edu) and Rajka Smiljanic (Dept. of Linguist., The Univ. of Texas at Austin, Austin, TX)

The study investigated how the use of protective face masks and language experience shape the production of listener-oriented clear speech. One L1 and one L2 English talker read sentences in a clear and conversational speaking style with and without a surgical mask. Formant trajectories between the onset and offset of two diphthongs, /aɪ/ and /eɪ/, were analyzed using Euclidean distance in the F1–F2 vowel space. The results showed that the distance between the onset /a/ and the offset /ɪ/ was larger when speech was produced without the mask and when speaking clearly. These modifications were larger for L1 talker compared to the L2 talker. The Euclidian distance for /eɪ/was only affected by speaking style. The results suggest that talkers produced hyperarticulated diphthongs characterized by larger formant movements in clear speech. The presence of a mask limited jaw movement for the diphthong containing the low vowel. Additionally, the L1 talker made larger articulatory modifications in response to the presence of a mask and listener-oriented clear speech compared to the talker with less extensive experience with the target language. These production patterns may be related to lower word recognition in noise for masked, conversational, and L2 speech found in previous work.

4pSC24. Register shifts in whistling: Investigating the influence on tongue shaping and gender-related variances. Laryssa A. Lancaster (Univ. of BC, 7480 Belair Dr., Richmond, BC V7A1B6, Canada, laryssalancaster@gmail.com), Dayeon Choi, Justine Cavaco, Chelsea Lisiecki, Jahurul Islam, and Bryan Gick (Univ. of BC, Vancouver, BC, Canada)

The study examines whistling register changes and the influence of tongue shape. In previous work, [Kaburagi *et al*., 2018] used magnetic resonance imaging (MRI), finding the degree and type of restriction made by the tongue directly impacted whistle F0. [Belyk et.al., 2019] found similar results using real-time magnetic resonance imaging (rtMRI), however, did not specifically observe register shifts. This study aimed to validate whistle register shifts and identify their variations among individuals regarding tongue position and gender. We hypothesize that participants with broader whistling ranges will exhibit register shifts based on tongue shape. Measures based on ultrasound imaging of male and female participants recorded while completing whistling exercises will be included in the analysis. This methodology enables tongue movements to be visualized, offering a view of the interplay between whistling registers and tongue configurations. ImageJ analysis results of tongue positions from ultrasound imaging will be presented with relevance to whether individuals produce a shift in their vocal register while whistling, and whether these conclusions depend on gender, as in sung register. Implications will be discussed regarding potential similarities between register changes during whistling and tongue shapes associated with speech.

Session 4pSP

Signal Processing in Acoustics: Signal Processing in Acoustics Potpourri II

Raymond Plasse, Cochair
Nashville, TN

Trevor Jerome, Cochair
Naval Surface Warfare Center, Carderock Division, 9500 MacArthur Blvd, BLDG 3 #329, West Bethesda, MD 20817

Chair's Introduction—1:00

Contributed Papers

1:05

4pSP1. Range and depth estimates from array invariant and model inversion, compared with machine learning approach. Paul Hursky (Appl. Ocean Sci. LLC, 4825 Fairport Way, San Diego, CA 92130, paul.hursky@gmail.com)

The array invariant is derived from the waveguide invariant, and can be used to predict the shape of multipath arrival patterns on a 2D plane with axes of time and angle of arrival. The shape observed on a vertical line array is an ellipse that is a function of the waveguide invariant and the range. Once the range is obtained, ray tracing can be used to invert the multipath pattern for a depth estimate. We will compare results of using the physics-based range and depth estimator with results obtained by training a neural network on synthetic data produced by a ray tracer. We will produce two sets of exemplars to train our estimator. In one, we will provide the range as an input, and train the network to produce a depth. In the second, we will train the network to produce both range and depth. The method derived from the array invariant is expected to be brittle with respect to range-dependent bathymetry. We will compare how well the machine learning approach deals with such variability, compared to the array invariant approach.

1:20

4pSP2. Inversion of partially spanning array data for normal mode parameters using seabed impedance constraint. Paul Hursky (Appl. Ocean Sci. LLC, 4825 Fairport Way, San Diego, CA 92130, paul.hursky@gmail.com)

Previous work with fully or nearly spanning vertical line arrays has shown that normal mode shapes can be identified as the eigenvectors of the covariance matrix averaged over lengthy source tracks that served to de-correlate the modes. Recent work with partially spanning arrays has used compressed sensing or super-resolution methods like MUSIC to estimate mode wavenumbers. These methods assume the sound speed profile is available, but not the seabed properties. We will present another alternative for inverting for normal modes under these conditions, including a partially spanning vertical line array, in which sets of candidate modes are restricted to modes that satisfy the same impedance boundary condition at the seabed. Each distinct impedance condition defines a physically consistent set of modes. Thus, our inversion is sparse in impedance conditions. We demonstrate how the normal modes resulting from this process are used to form matched field processing replicas that recover source tracks in range and depth.

1:35

4pSP3. Real-time, sample-by-sample estimation of multiple signal waveforms from acoustic array data using B-splines. Garth Frazier (NCPA, Univ. of MS, NCPA, University of MS, P.O. Box 1848, Oxford, MS 38677, frazier@olemiss.edu)

This work presents a real-time signal processing algorithm that estimates *time-domain* waveforms of multiple plane wave signals on a sample-by-sample basis from data measured by an acoustic array. Moreover, the algorithm provides sample-by-sample estimates of the direction-of-arrival (DOA) of the waveforms. In this case sample-by-sample means that as each sample of data is measured by the array estimates of the waveforms and their directions-of-arrival are updated. While the basic idea can be based on almost any finite-dimensional approximation to a function space, this algorithm makes use of B-spline basis for the signal waveforms. Other features of the approach include no restrictions on signal waveform shape, automatic estimation of the number of signals present, automatic step-size adjustment for correction of signal waveform and direction-of-arrival parameters to achieve good performance while guaranteeing stability, and the use of circular (spherical) statistical models for 2-D (3-D) DOA estimates. The approach contrasts with blind source separation (BSS) methods that are based on non-Gaussian statistical assumptions and that do not assume a known array geometry nor a propagation model. This method is closely related to algorithms based on block-of-data by block-of-data processing but does not require stitching results from block processing to create time-domain waveform estimates.

1:50–2:05 Break

2:05

4pSP4. Auditory-inspired adaptive frequency tracking. Vijay Peddinti (NUWC Div. Newport, Howell St., Newport, RI 02841, vijaykumar.peddinti.civ@us.navy.mil)

One of the best frequency trackers to date is the human (mammalian) auditory system, which has evolved through millions of years of resolving classification problems. It is a versatile, elegant and powerful sound processing unit. It excels in detecting, estimating, and classifying multiple targets simultaneously even in noisy environments. Hence, mimicking even some features of the auditory system could be beneficial in developing superior frequency tracking and classification algorithms. An auditory inspired adaptive synchrony capture filterbank (SCFB) signal processing architecture for tracking signal frequency components was proposed in a related paper [JASA-2013]. The SCFB architecture consists of a fixed array of traditional, passive linear, gammatone filters in cascade with a bank of three adaptively tunable bandpass filters that form a frequency-discriminator-loop (FDL).

The SCFB exhibits many desirable properties for processing speech, music, and other complex sounds. In recent work [Dec2021], the algorithm was modified using adaptive tuning parameters, and a generalized way to determine/suppress voiced and unvoiced (silent) regions. This modified algorithm estimates frequencies with higher accuracy even in the presence of closely spaced input tones. Preliminary analysis with synthetic, human speech and humpback/whale-call signals demonstrates that the revised algorithm performs well. This talk will focus on follow-on work.

2:20

4pSP5. Multichannel signal transmission and reception using compact multichannel underwater communication devices: A unified theory and experimental results from different underwater media. Ali Abdi (Elec. Comput. Eng. Dept., New Jersey Inst. of Technol., 323 Dr. Martin Luther King Jr Blvd, Newark, NJ, ali.abdi@njit.edu), Erjian Zhang, and Rami Rashid (Elec. Comput. Eng. Dept., New Jersey Inst. of Technol., Newark, NJ)

Simultaneous transmission of multiple signals over the same bandwidth allows for optimal utilization of the limited bandwidth in underwater systems and networks. To avoid using multiple spatially separated single-channel projectors, one compact multichannel device can be used instead [E. Zhang, R. Rashid, A. Abdi, “Particle velocity underwater data communication: Physics, channels, system and experiments,” IEEE J. Oceanic Eng. (2023); E. Zhang, R. Rashid, A. Abdi, “Underwater communication experiments for transmitting multiple data streams using a vector acoustic MIMO system: OFDM and FSK modulations,” Proc. Oceans (2023)]. Simultaneous reception of multiple signals using one compact multichannel device is also feasible and improves the communication performance, without using several spatially separated single-channel hydrophones [R. Rashid, E. Zhang, A. Abdi, “On the performance of a new wireless communication compact multichannel underwater receiver using a sphere vector sensor,” IEEE Transactions on Vehicular Technology (2023); R. Rashid, E. Zhang, A. Abdi, “Underwater acoustic signal acquisition and sensing using a ring vector sensor communication receiver: Theory and experiments,” Sensors (2023)]. In this paper, we present various experimental results using a number of compact multichannel underwater devices in different setups, and demonstrate their usefulness for underwater communication. [Work supported in part by NSF, Grant IIP-1500123].

2:35

4pSP6. Dual-repetitive pilot aided parameter estimation for underwater orthogonal frequency division multiplexing acoustic communications in the presence of strong composite noise. Zhiqiang Liu (US Naval Res. Lab., 4555 Overlook Ave. Washington, DC 20375, zhiqiang@ieee.org)

In this work, a novel design of signaling parameters and pilot placement is proposed for supporting robust underwater orthogonal frequency division multiplexing acoustic communications in the presence of strong composite noise. The proposed design allows for high-resolution estimation of various channel parameters that are essential to successful symbol recovery. Receiver processing algorithms are developed by exploring special properties of the received signal that hold valid despite complicated channel and noise effects. These algorithms are efficient since they reuse the same set of pilot symbols for multiple tasks of estimation that often require separate sets of pilots. They are also extremely effective because they largely avoid the effects of noise even when the noise is composite. The performance of these algorithms is evaluated via both extensive simulations and an at-sea experiment.

Session 4pUW

Underwater Acoustics: Underwater Acoustic Modeling

Benjamin Cray, Cochair
NUWC, 1176 Howell Street, Newport, RI 02841-1708

Jessica Desrochers, Cochair
Ocean Engineering, The University of Rhode Island, 48 Villa Avenue, Warwick, RI 02886

Contributed Papers

1:00

4pUW1. Acoustic propagation modeling using sound velocity profiles estimated from high resolution temperature data. Brian Amaral (Ocean State Sensing, 21 Admiral Kalbfus Rd., Newport, RI 02840, Oceanstatesensing@gmail.com), Jennifer L. Amaral (Marine Acoust., Inc., Middletown, RI), Antone B. Eliasen (Ocean State Sensing, Mclean, VA), and Russell Shomberg (Marine Acoust., Inc., Newport, RI)

An accurate sound velocity profile (SVP) is an important input for underwater acoustic propagation modeling. The SVP is largely driven by the temperature of the water column, especially at shallower depths. Measuring the SVP in high spatial and temporal resolution is challenging with traditional instrumentation, however techniques exist to produce high resolution temperature measurements that can be used to derive estimates of the SVP. Fiber optic distributed temperature sensing (DTS) offers an improvement of several orders of magnitude over traditional singular point measurement devices and has the ability to measure temperature over large range and depth scales in an efficient manner. A DTS system was towed for 172 nm off the coast of New England over several days. The measured temperature data was used to estimate the SVP across the range and depth and will be compared to traditional environmental data sets, such as HYCOM. A comparison of modeled acoustic propagation using different resolution SVPs will be presented.

1:15

4pUW2. Understanding model fidelity for training synthetic aperture sonar image classifiers. Thomas E. Blanford (Univ. of New Hampshire, University of New Hampshire, Durham, NH 03824, thomas.blanford@unh.edu), David Williams (The Penn State Univ., La Spezia, SP, Italy), J. Daniel Park (The Penn State Univ., State College, PA), Brian Reinhardt (The Penn State Univ., University Park, PA), Shawn Johnson, and Daniel C. Brown (The Penn State Univ., State College, PA)

Convolutional neural networks (CNNs) are increasingly employed for classification tasks in automated target recognition (ATR) algorithms for synthetic aperture sonar (SAS) images. Training ATR algorithms, however, requires many unique observations of the targets of interest. Data available for training is often limited, and acquiring additional training data through experimentation is prohibitively expensive. Increasingly, training with simulated data is being considered as an alternative, but the fidelity required of the models that generate this data is not yet known. SAS imagery typically contains significant complexity from countless physical mechanisms. Simulating complex scenes requires multiple models to account for these different effects, but some effects may not require accurate modeling for the purposes of training an ATR algorithm. This presentation will describe a study to investigate the fidelity of models required so that simulated data may be used interchangeably with experimental data for training CNNs. Using in-air experimentation, a high-fidelity data set was developed with multiple degrees of complexity. A high-frequency sonar signal model was then used to generate complementary simulated data. This approach allows for specific physical features in the data to be individually isolated, enabling detailed exploration of the relationships between model fidelity and CNN architecture.

1:30

4pUW3. Effects of seabed corrugation and stratification on 3D acoustic propagation along the New England Shelf Break. Brendan J. DeCourcy (Woods Hole Oceanographic Inst., 86 Water St., Falmouth, MA 02543, bdecourcy@whoi.edu), Ying-Tsong Lin (Scripps Inst. of Oceanogr., La Jolla, CA), and Jason Chaytor (U.S. Geological Survey, Woods Hole, MA)

Acoustic signals transmitted and recorded during the New England Shelf Break Acoustics Experiment in 2021 (NESBA) display evidence of complex scattering effects which were not captured in *in situ* modeling efforts. In particular, although the acoustic parabolic equation models used can predict later arrivals of a single transmission, initial arrivals show mismatch between model predictions and observation. In this presentation, estimates of sub-bottom sediment structure are supplemented by a high-resolution model of the bathymetry to simulate 3D sound propagation across a realistic seabed. The influence of the seabed corrugation and stratification on acoustic arrival times is examined in the context of the NESBA environment. [This research is supported by the Office of Naval Research].

1:45

4pUW4. Acoustic arrival patterns in the Beaufort duct using normal mode and ray trace acoustic prediction models. Jessica Desrochers (Ocean Eng., The Univ. of Rhode Island, 48 Villa Ave., Warwick, RI 02886, jessicabfothergill@gmail.com), Lora Van Uffelen (Ocean Eng., Univ. of Rhode Island, Narragansett, RI), and Alexander Muniz (NUWC, Newport, RI)

The stratification of the Beaufort Sea has experienced significant changes over the last few decades resulting in a subsurface duct between 100- and 300-meters depths, known as the Beaufort Duct. This duct allows for long-range acoustic transmissions due to little interaction with the seafloor or sea surface. Acoustic arrival predictions for broadband acoustic sources centered around 250 Hz, such as those deployed in the Beaufort Sea in 2016–2017 show a peak acoustic arrival prior to the final cutoff centered on the sound speed minimum in the duct. This reverse dispersion feature in the acoustic time front can be connected back to the unique ducting features in the sound-speed profile. This relationship is explored using normal mode modeling and geometric optics. Modal speed predictions and ray path lengths and travel times are used to interpret the acoustic arrival patterns, particularly the dispersion feature present in the acoustic time front.

2:00

4pUW5. Temporal evolution of sound pulse in shallow water under conditions of horizontal refraction. Vertical modes and space-time horizontal rays. Boris Katsnelson (Marine Geosciences, Univ. of Haifa, 199 Adda Khouchy Ave. Haifa 3498838, Israel, bkatsnels@univ.haifa.ac.il) and Alexander Kaplun (Marine Geosciences, Univ. of Haifa, Haifa, Israel)

As is known, to describe horizontal refraction in a shallow water waveguide, the "vertical modes and horizontal rays" approach (Burridge & Weinberg, 1974) is used, within which the two-dimensional eikonal equation for the modal amplitude includes the refractive index determined by the eigenvalues depending on the horizontal coordinates (x,y) and frequency (f). In other words, we have an effective two-dimensional inhomogeneous dispersive medium, to describe the propagation of signals in which, the authors propose to use the so-called space-time rays (Rytov, 1940, Connor & Felsen, 1974, Hillion, 1993) applied for the horizontal plane (STHR). The set of STHRs in a given situation is constructed for each mode. Within the framework of this approach, it is possible to obtain the evolution of the pulses of each mode from the solution of the nonstationary eikonal equation in the horizontal plane or from the corresponding Hamilton-Jacobi equations, obtained using non-local integral wave equation. Examples of constructing STHR in some typical situations (in particular in a wedge) are considered, some specific effects during signal propagation are considered, in particular, the so-called the pulse front tilt, previously discovered in laser optics (Bor & Rasz, 1985). Work was supported by ISF, grant 946/20.

2:15

4pUW6. The effects of bathymetry on estimating the depth of a source in range-dependent environments with a single hydrophone. Ivars P. Kirsteins (NUWCDIVNPT, 1176 Howell St., Newport, RI 02813, i.kirsteins@gmail.com)

The effects of varying or range-dependent bathymetry, e.g., such as a sloping bottom, on the depth resolution characteristics of single hydrophone matched field processing (MFP), e.g., width of depth main-lobe as a function of source and receiver nominal positions, is not well understood. A major challenge is that in the case of range-dependent environments there are generally no analytical representations for the pressure field, thus requiring numerical methods for their evaluation. To provide some theoretical insights into how the shape of the bathymetry and locations of the source and receiver influences the MFP depth ambiguity function, we study the case of an ideal wedge with perfectly reflective pressure-release boundaries using the analytical solutions of Buckingham [1]. We derive a simple approximation for the depth ambiguity function main lobe width and compare it against the theoretical result for an ideal range-independent environment [2] and then numerically for a penetrable wedge and more realistic shelf-like environments. M.K. Buckingham, "Acoustic propagation in a wedge-shaped ocean with perfectly reflecting boundaries," NRL report 8793, March 1984. M. Cheney and I.P. Kirsteins, "Resolution of matched field processing for a single hydrophone in a rigid waveguide," J. Acoust Soc. Am. 151, pp. 3186–3197, 2022.

2:30

4pUW7. Deconvolution of the waveguide impulse response for source localization at low frequency. Florent Le Courtois (DGA TN, Ave. de la Tour Royale, Toulon 83000, France, florent.lecourtois@gmail.com), Bazile Kinda (DGA TN, Brest, France), and Myriam Lajaunie (Shom, Brest, France)

Matched-mode processing (MMP) methods have been widely investigated for source localization and geoacoustic inversion. For horizontal line arrays (HLA), it aims at retrieving the horizontal wavenumbers and thus at inferring source range and bearing. In this work, the matched-mode is performed by applying the deconvolution of the waveguide response in a narrowband context in order to improve the localization accuracy. Deconvolution usually refers to a sparse framework when the number of sources is smaller than the number of sensors of the HLA. The Orthogonal Matching Pursuit algorithm is then used in this work. First results from numerical simulations for Pekeris waveguide highlight better localization accuracy than MMP. The deconvolution shows as well stronger robustness to mismatch in the sound speed values of the seabed. Results from measurement campaign, involving large HLA, low to ultra-low frequency and water depth up to 1500 m are investigated as well.

2:45–3:00 Break

3:00

4pUW8. A study of model selection techniques for modeling synthetic aperture sonar backscatter statistics. Derek R. Olson (Oceanogr., Naval Postgrad. School, 833 Dyer Rd., 313b Spanagel Hall, Monterey, CA 93943, olson.derek.r@gmail.com) and Marc Geilhufe (Norwegian Defence Res. Establishment, Kjeller, Norway)

High resolution acoustic imaging of the seafloor, such as with synthetic aperture sonar, can reveal complex environments due to the ability to resolve scatterers of different types. The statistical distribution of the backscattered field is appropriately modeled using a mixture distribution, which consists of a sum of a K pdf for each scatterer type, weighted by the relative frequency with which they occur. In this type of modeling, the number of distributions must be selected before parameters can be fit, and as the number of components increases, so does the danger of overfitting the data. Several methods of model selection are explored here. Two methods, the Bayesian information criterion and the Akaike information criterion are based on point parameter estimates, and a penalty due to the number of model parameters. The other two, the deviance information criterion and the Watanabe-Akaike information criterion, are based on Monte-Carlo sampling of the model parameter space. These four model selection criteria are compared to each other, and to the authors manual selection of the number of components.

3:15

4pUW9. Analysis of seamount propagation through the first convergence zone. Johnathan J. Todd (Penn State, 201 Appl. Sci. Bldg., University Park, PA 16802, jjt6090@psu.edu), William Hodgkiss (Scripps Inst. of Oceanogr., San Diego, CA), Daniel C. Brown (Penn State, State College, PA), and Chad M. Smith (Penn State, State College, PA)

During the recent Office of Naval Research (ONR) New England Seamounts Acoustics (NESMA) Pilot experiment, continuous measurements were made of underwater acoustic propagation and scattering from close range through the first deep-water acoustic convergence zone (CZ) using a stationary mid-frequency source and the ONR Three Octave Research Array (THORA) at Penn State. The source was positioned at a shallow depth over the plateau of one of the Atlantis II seamounts; the THORA was towed from 1 km range to a maximum of 85 km from the source. High-level goals of this work are to: 1) assess the influence of seamount scattering on mid-frequency transmission loss (TL) within the shadow zone, and 2) characterize how seamount interaction and the CZ is influenced by the shallow sound speed profile (SSP) in this region. The data was beamformed and analyzed using narrowband and broadband techniques, while ray-based and parabolic equation (PE) based models were used to develop an understanding of the influence of the seamount bathymetry, roughness, and the SSP. This talk will discuss a frequency dependent decrease in TL (acoustic enhancement) within the shadow zone due to seamount interaction and the limited influence of the seamount within the first CZ.

3:30

4pUW10. A model-experiment validation of near-range phase center breakdown. Jonah Warner (The Penn State Univ., 805 West Aaron Dr., Apt A1, State College, PA 16803, jbw5756@psu.edu), Thomas E. Blanford (Univ. of New Hampshire, Durham, NH), and Daniel C. Brown (Penn State Univ., State College, PA)

Spatial coherence of a scattered field has helped to improve sensor navigation techniques and field imagery. Under the phase center approximation (PCA), a bistatic sensor geometry is approximated as a monostatic phase center. In the far field, the approximation is valid, and signals with common phase centers are highly coherent. The approximation fails when a non-trivial spatiotemporal delay appears in the sensor's near-field, and signals that share phase centers are less coherent. A model by Brown and Blanford shows this near-field degradation of PCA, using comparisons between the van Cittert-Zernike Theorem and a point-based scattering model. Experimental validation of the model was performed using an in-air sonar mounted

on a linear actuator. The sonar collected returns from a shallow bed filled with plastic beads, which approximates rough interfaces typical of underwater environments. Multiple trials were conducted using a variety of transmitters, receiver placements, and ping intervals. These geometries were simulated using the point-based scattering model and the two sets of data were compared in order to validate the model. Results from this model-data comparison, along with future applications for sonar imaging and navigation, will be discussed.

Invited Paper

3:45

4pUW11. Hybrid model for acoustic and vibration predictions based on vessel induced acoustic vibration: A review. Solomon O. Ologe (Mech., Univ. Polytechnic of Catalonia, Calle de Colom, 11, Tarrassa Campus, Barcelona 08222, Spain, ologe.solomon@upc.edu)

This research examines hybrid models for acoustic and vibration predictions based on vessel-induced acoustic vibration. Acoustic and vibration are caused by complex machineries and marine plants, affecting crew members and aquatic life. To protect marine species, urgent action is needed. Previous studies have developed models for estimating and predicting acoustic and vibration levels, but these models have limitations. Hybridization, combining numerical and computational methods, and simulations of vibro-acoustic behavior could improve results. Hybridization is encouraged for greater efficiency.

Contributed Papers

4:00

4pUW12. Analysis of bubble curtain effectiveness in the coastal Virginia offshore wind turbine installation. Gavin Dies (Ocean Eng., Univ. of Rhode Island, Narragansett, RI), Ying-Tsong Lin (Woods Hole Oceanographic Inst., 266 Woods Hole Rd., Woods Hole, MA 02543, ytlin@whoi.edu), Gopu R. Potty, James H. Miller (Ocean Eng., Univ. of Rhode Island, Narragansett, RI), Jade Case (Ocean Eng., Univ. of Rhode Island, Winter Park, FL), Jennifer L. Amaral (Marine Acoust., Inc, Middletown, RI), and Anwar A. Khan (HDR Inc, Fort Lauderdale, FL)

The Coastal Virginia Offshore Wind (CVOW) consists of two turbines roughly 40 km off the coast of Virginia Beach, Virginia. Water- and seabed-borne acoustic signals from impact pile driving at CVOW were recorded during the installation of these turbines in May 2020. In-water pressure signals were measured using two vertical line arrays (VLAs) at ranges of 3 km and 7.5 km. The water depth at the wind turbines and both VLAs was about 26 m. During installation, one of the piles utilized a double bubble curtain during construction while the other pile did not. Bubble curtains are used to reduce the acoustic impacts of pile driving by creating a barrier of bubbles around the source. We found that the bubble curtains were most effective at longer ranges and at frequencies above 200 Hz. Normal modes are an efficient representation of the acoustic field in this relatively range-independent shallow water environment. The amplitudes of acoustic modes at several frequencies were used to analyze the effectiveness of bubble curtains. Energy propagating in the seabed was also measured by an array of geophones and was found to be unattenuated by the bubble screens.

4:15

4pUW13. An evaluation of cross-modal acoustic intensity modulation. Benjamin Cray (NUWC, 1176 Howell St., Newport, RI 02841-1708, benjamin.cray@navy.mil)

Range to a submerged source, in a shallow-water waveguide, is determined using a single vector sensor receiver. Low order cross-modal variations in acoustic intensity (all four components, vertical and radial reactive and active) estimate target range. The algorithm relies on intensity modulations, generated from the superposition of a limited number of low-order trapped modes. Performance was evaluated with Acoustic Toolkit's SCOOTER[1,2] software (a Fast Field Program based on a contour integral representation of the range-independent waveguide's acoustic pressure). SCOOTER, which allows for sediment attenuation, multiple seafloor layers, and various sound speed profiles, generates the corresponding Depth-Dependent Green's Function. Comparisons of SCOOTER predictions to measured data, collected in Monterey Bay (California), and within the New England Mud Patch (Rhode Island), will be presented. Maggi, A. L., Duncan, A. J., "Acoustic Toolbox User interface and Post processor, AcTUP V2.2L", Centre for Marine Science & Technology, Curtin University of Technology, Perth, AU. Alec Duncan, "A Consistent, User Friendly Interface for Running a Variety of Underwater Acoustic Propagation Codes", Proceedings of ACOUSTICS 2006, 20–22 November 2006, Christchurch, New Zealand