Session 1aAA

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

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

Invited Papers

8:05

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

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

8:25

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

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

8:45

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

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

9:05

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

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

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

9:25

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

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

Contributed Papers

9:45

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

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

10:00–10:15 Break

10:15

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

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

10:30

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

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

10:45

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

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

11:00

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

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

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

11:15

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

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

11:30

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

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

11:45

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

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

Session 1aBAa

Biomedical Acoustics and Physical Acoustics: Sonobiopsy for Noninvasive Molecular Diagnosis

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

Invited Papers

8:00

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

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

8:30

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

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

9:00

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

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

9:30

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

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

10:00–10:15 Break

10:15

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

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

10:45

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

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

11:15

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

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

Contributed Paper

1a MON. AM

11:45

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

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

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

Session 1aBAb

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

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

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

Chair's Introduction—8:55

Invited Papers

9:00

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

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

9:20

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

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

9:40

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

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

10:00

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

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

10:20–10:35 Break

Contributed Papers

10:35

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

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

10:50

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

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

11:05

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

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

11:20

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

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

11:35

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

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

Session 1aCA

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

Shung H. Sung, Cochair
SHS Consulting, LLC, 4178 Drexel Dr., Troy, MI 48098

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

Invited Papers

9:30

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

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

9:50

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

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

10:10

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

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

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

10:30–10:45 Break

10:45

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

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

11:05

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

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

11:25

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

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

Contributed Paper

11:45

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

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

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

Session 1aID

Interdisciplinary and Student Council: Introduction to Technical Committees

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

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

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

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

Invited Papers

8:00

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

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

8:20

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

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

8:40

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

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

9:00

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

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

9:20

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

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

9:40

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

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

10:00–10:15 Break

10:15

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

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

10:35

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

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

10:55

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

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

11:15

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

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

11:35

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

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

Session 1aPA

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

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

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

Chair's Introduction—8:00

Invited Papers

8:05

1aPA1. Unilateral transmission in bilinearly coupled systems. Ali Kogani (Concordia Univ., Montreal, Montreal, QC, Canada) and Behrooz Yousefzadeh (Concordia Univ., Montreal, 1455 De Maisonneuve Blvd. W., Rm. EV-4.139, Montreal, QC H3G 1M8, Canada, behrooz.yousefzadeh@concordia.ca)

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

8:25

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

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

8:45

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

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

9:05

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

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

9:25

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

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

9:45–10:00 Break

10:00

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

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

Contributed Papers

10:20

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

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

10:35

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

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

10:50–11:50
Panel Discussion

Session 1aPP

Psychological and Physiological Acoustics: Psychological and Physiological Acoustics Poster Session

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

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

Contributed Papers

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

Even with the widespread adoption of newborn hearing screening and Early Hearing Detection and Intervention systems, infants who are hard-of-hearing (IHH) remain at increased risk for poor or delayed development of auditory and speech perception skills. Speech perception is positively correlated with better auditory development on functional auditory skills and a variety of language outcomes. However, clinical utilization of speech perception tests has not been broadly accepted. One clinically viable tool to assess speech perception during infancy is our electroencephalography (EEG) procedure, which has been validated as an objective measure of speech discrimination. However, we have not examined the development of infant speech perception among IHH and infants with normal hearing (INH). Perceptual attunement—or "narrowing"—is a model of perceptual learning that posits that perceptual abilities are shaped by environmental experiences over the first year of life. Here, we describe our adapted EEG methods to examine the development of infant speech perception to study periods of perceptual attunement between IHH and INH. Preliminary findings confirm that at 3 months of age, infants demonstrate neural encoding of native and non-native speech sounds, and at 6 months of age results suggest a decrease in non-native speech sound encoding.

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

1aPP20. Abstract withdrawn.

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

1aPP29. Abstract withdrawn.

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

Session 1aSA

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

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

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

Chair's Introduction—10:00

Invited Papers

10:05

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

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

10:30

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

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

10:55

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

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

11:20

1aSA4. Abstract withdrawn.

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

Session 1aSP

Signal Processing in Acoustics: Signal Processing in Acoustics Poster Potpourri

Trevor Jerome, Chair

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

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

Contributed Papers

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

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

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

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

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

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

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

Session 1aUW

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

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

Zoi-Heleni Michalopoulou, Cochair
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Tracianne B. Neilsen, Cochair
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Haiqiang Niu, Cochair
Inst. of Acoust., Chinese Acad. of Sci., No. 21 North 4th Ring Road, Beijing 100190, China

Chair's Introduction—8:00

Invited Papers

8:05

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

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

8:25

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

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

8:45

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

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

9:05

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

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

Contributed Papers

9:25

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

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

9:40–9:55 Break

9:55

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

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

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

10:10

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

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

Invited Paper

10:25

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

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

Contributed Papers

10:45

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

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

11:00

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

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

Invited Paper

11:15

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

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

Contributed Paper

11:35

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

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

Session 1pAA

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

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

Invited Papers

1:00

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

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

1:20

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

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

1:40

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

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

2:00

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

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

2:20–2:35 Break

Contributed Papers

2:35

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

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

2:50

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

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

3:05

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

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

3:20

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

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

3:35

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

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

3:50

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

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

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

Session 1pBAa

Biomedical Acoustics: General Topics in Biomedical Acoustics: Imaging

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

Contributed Papers

1:00

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

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

1:15

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

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

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

1:30

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

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

1:45

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

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

2:00

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

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

2:15

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

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

2:30–2:45 Break

2:45

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

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

3:00

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

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

3:15

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

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

3:30

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

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

3:45

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

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

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

4:00

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

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

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

Session 1pBAb

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

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

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

Invited Papers

1:00

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

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

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

1:20

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

There is a long-existing tradeoff between the imaging resolution and the penetration depth in imaging systems caused by the diffraction limit. We developed a "blind label" approach to tackle this problem, which significantly improves the practicality of acoustic subwavelength imaging in biomedical ultrasound imaging, non-destructive testing, and other acoustic sensing and communication applications. The "blind labels" in our system refer to randomly distributed acoustic scatterers with deep-subwavelength sizes whose exact locations and trajectories are not necessary information in image reconstruction. Our imaging framework is composed of two parts: (1) spatial mixing: a physical process that converts the originally evanescent components in the scattered waves from the object to propagating components that can reach the far-field detector and (2) computational reconstruction. In this talk, we will mainly report our quantitative investigation of the system parameters' impact on the performance of the blind-label subwavelength imaging system, providing guidance to future system setups in various applications.

1:40

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

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

2:00

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

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

2:20–2:35 Break

Contributed Papers

2:35

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

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

2:50

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

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

3:05

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

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

3:20

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

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

3:35

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

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

3:50–4:10
Panel Discussion

Session 1pCA

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

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

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

Invited Papers

1:00

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

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

1:20

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

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

Contributed Paper

1:40

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

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

Session 1pEA

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

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Chair's Introduction—1:00

Invited Papers

1:05

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

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

1:25

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

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

1:45

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

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

2:05

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

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

2:25–2:40 Break

Contributed Papers

2:40

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

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

2:55

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

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

3:10

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

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

3:25

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

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

3:40

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

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

3:55

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

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

Session 1pMU

Musical Acoustics: General Topics in Musical Acoustics I

Andrew A. Piacsek, Chair
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Contributed Papers

1:30

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

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

1:45

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

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

2:00

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

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

2:15

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

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

2:30

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

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

2:45–3:00 Break

3:00

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

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

3:15

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

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

3:30

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

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

3:45

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

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

Session 1pNSa

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

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Contributed Papers

1:00

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

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

1:15

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

The rapid launch cadence of SpaceX's Falcon 9 rocket provides the opportunity to study the relative consistency of far-field noise propagation. Acoustical measurements of several Falcon 9 launches have been made on and near Vandenberg Space Force Base at a far-field location 8.4–14 km from the pad. This paper compares collocated measurements from different Falcon 9 missions to begin to understand data variability as a function of launch and environmental conditions at far-field locations. The 8.4 km location has been measured over 10 times, whereas other locations span subsets of launches. This comparative analysis includes time-varying levels, spectra, and waveform statistics, such as the pressure derivative skewness. Time periods of particular interest are liftoff, peak noise, and late into the launch when the vehicle is significantly downrange. [Work supported by USACE.]

1:30

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

The rapid launch cadence of SpaceX's Falcon 9 rocket provides the opportunity to study the relative consistency of noise radiation. Acoustical measurements of several Falcon 9 launches have been made on and near Vandenberg Space Force Base within 1 km of the launch pad. This paper compares collocated mid-field measurements from different Falcon 9 missions to begin to understand data variability as a function of launch and environmental conditions. One location 395 m from the pad has been measured over 5 times. This comparative analysis includes time-varying levels, spectra, and waveform statistics such as the pressure derivative skewness. Time periods of particular interest are liftoff, peak noise, and late into the launch when the vehicle is significantly downrange. [Work supported by USACE.]

1:45

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

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

2:00

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

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

2:15

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

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

2:30–2:45 Break

2:45

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

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

3:00

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

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

3:15

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

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

3:30

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

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

3:45

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

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

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

Session 1pNSb

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

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

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

Chair's Introduction—1:00

Invited Papers

1:05

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

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

1:25

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

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

1:45

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

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

2:05

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

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

2:25

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

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

2:45–3:00 Break

Contributed Papers

3:00

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

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

3:15

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

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

3:30

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

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

Invited Papers

3:45

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

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

4:05

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

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

4:25

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

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

4:45

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

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

Contributed Paper

5:05

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

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

Session 1pPAa

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

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

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

Chair's Introduction—1:00

Invited Papers

1:05

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

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

1:30

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

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

1:55

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

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

2:20

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

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

2:35–2:50 Break

2:50

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

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

3:05

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

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

3:20

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

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

3:35

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

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

Session 1pPAb

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

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

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

Invited Papers

1:00

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

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

1:20

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

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

1:40

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

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

2:00

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

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

2:20

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

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

2:40

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

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

3:00–3:15 Break

Contributed Papers

3:15

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

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

3:30

1pPAb8. Abstract withdrawn.

3:45

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

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

4:00

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

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

4:15

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

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

4:30

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

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

4:45

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

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

5:00

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

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

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

5:15

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

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

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

Session 1pPP

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

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

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

Chair's Introduction—1:00

Invited Papers

1:05

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

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

1:30

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

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

1:55

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

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

2:20–2:35 Break

2:35

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

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

3:00

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

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

3:25

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

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

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

Session 1pSA

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

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

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

Chair's Introduction—1:00

Invited Papers

1:05

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

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

1:25

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

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

1:45

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

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

2:05

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

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

Contributed Papers

2:25

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

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

2:40

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

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

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

2:55

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

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

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

Session 1pSC

Speech Communication: Speech Perception Poster Session I

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

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

Contributed Papers

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

1pSC10. Listening effort for L2 accents: Adults' adaptation to Mandarin accented english. Ruth Altmiller (Psychol. and Brain Sci., Washington Univ. in St. Louis, 1 Brookings Dr., Campus Box 1125, St. Louis, MO 63130, ruth.altmiller@wustl.edu), Mel Mallard, and Kristin J. Van Engen (Psychol. and Brain Sci., Washington Univ. in St. Louis, St. Louis, MO)

Using task-evoked pupillary response (TEPR) and dual-task procedures, prior research has shown that adults recruit additional cognitive resources when processing fully intelligible L2-accented speech compared to speech in their own L1 accent (McLaughlin and Van Engen, 2020; Brown *et al*., 2020). In this study, we investigated the relationship between three prosodic measures (relative word duration variance, pitch stability, and pitch range) and changes in listening effort for L1 and L2 speakers over the course of an experiment. Using growth curve analysis, we found significant interactions between each of the three measures and trial number, indicating that prosodic factors influence speech adaptation. However, these interactions were in opposite directions for the L1 and L2 speakers. Thus, adult monolingual English listeners rely on prosodic information to adapt to a particular speaker's voice, but use that information differently based on speaker identity.

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

1pSC21. Exploring auditory recognition: Native Japanese listeners' perception of frequent and infrequent English words in noise with varied phonological neighborhood densities. Takeshi Nozawa (Ctr. for Lang. Education and Res., Ritsumeikan Univ., 1-1-1 Nojihigashi, Kusatsu, Shiga 5258577, Japan, t-nozawa@ec.ritsumei.ac.jp) and Ratree Wayland (Linguist., Univ. of Florida, Gainesville, FL)

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

1pSC32. Abstract withdrawn.

1pSC33. Abstract withdrawn.

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

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

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

This study explores the detrimental effects of classroom noise and teacher dysphonia on children's speech understanding and teacher voice

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

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

Session 1pSP

Signal Processing in Acoustics: Signal Processing in Acoustics Potpourri I

Trevor Jerome, Cochair
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Manton J. Guers, Cochair
Acoustics, Penn State Univ., P.O. Box 30, State College, PA 16804

Chair's Introduction—1:00

Contributed Papers

1:05

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

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

1:20

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

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

1:35

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

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

1:50

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

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

2:05

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

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

2:20

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

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

2:35

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

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

Session 1pUW

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

Alexander S. Douglass, Cochair
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Invited Papers

1:00

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

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

1:20

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

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

1:40

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

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

1:55

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

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

2:10

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

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

2:25

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

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

Invited Paper

2:40

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

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

Contributed Paper

3:00

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

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

3:15–3:30 Break

Invited Paper

3:30

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

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

3:50

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

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

4:05

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

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

4:20

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

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

4:35

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

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

4:50

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

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

5:05

1pUW15. Effect of oceanographic fluctuations on geo-acoustic inversions and source localization using ships of opportunity. Christian D. Escobar-Amado (Elec. and Comput. Eng., Univ. of Delaware, 139 The Green, Newark, DE 19716, escobarc@udel.edu) and Mohsen Badiey (Elec. and Comput. Eng., Univ. of Delaware, Newark, DE)

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

5:20

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

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