Session 2aAAa

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

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

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

Invited Papers

7:55

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

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

8:15

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

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

8:35

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

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

8:55

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

It's easy for laboratories to make sound absorption measurement for baffles and blades treatment options. The difficult part is to make measurements and report results in a method that can be useable across a wide variety of users. As example, the architectural acoustic community, the specifying community, and marketing teams, all need to communicate effectively to a diverse clientele each

with different needs and understanding. This paper will explore the methodology of one company's latest investment to provide applicable baffles and blades data.

9:15

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

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

9:35

2aAAa6. An in-depth look at empirical estimates of assembly performance. Michael Raley (PAC Int., 2000 4th Ave., Canby, OR 97013, mraley@pac-intl.com)

Acousticians often look for acoustical test data for wall and floor/ceiling assemblies that match the assemblies on their projects. However, with the endless possible permutations of assemblies, it can be difficult to find testing for an assembly that is an exact match. As a consultant, I frequently created engineering evaluations that documented my predicted performance of an assembly. As a manufacturer, I now have a say in what assemblies are tested, but I still can't test all the possible permutations. This presentation is an overview of my current approach to assembly estimates, including the development of test series that are specifically intended to produce the data necessary to create good estimates.

9:55

2aAAa7. Trampolines and insulation and data, Oh My!. Joseph Keefe (Ostergaard Acoust. Assoc., 1460 US Hwy. 9 North, STE 209, Woodbridge, NJ 07095, jkeefe@acousticalconsultant.com)

This presentation will discuss sound and vibration data gathered for two separate and unrelated acoustical consultations. The first is a consultation for an adventure park facility (trampolines, arcade games, ball pits, etc.) planned on the upper level of an existing mall; acoustical concerns included vibration from jumping on trampolines and noise from amplified music to adjacent tenant spaces. The second is a comprehensive set of sound level reduction measurements in accordance with ASTM E336 / ASTM E2235, as well as ASTC calculations in accordance with ASTM E413; these results facilitated an investigation of the effect of complete versus partial cavity insulation within partitions.

10:15–10:30 Break

10:30

2aAAa8. A comparative study of modeled and measured room responses. Nicolaus T. Dulworth (Threshold Acoust., 141 W Jackson Blvd Ste 2080, Chicago, IL 60604, ndulworth@thresholdacoustics.com), Shane J. Kanter (Threshold Acoust., Chicago, IL), and Carl P. Giegold (Threshold Acoust. LLC, Evanston, IL)

At the heart of Brown University's Lindemann Performing Arts Center is a shape-shifting hall capable of supporting a wide range of performance types. Acoustic simulation played a crucial role in optimizing each of the hall's five primary configurations during design. Leading up to the opening events 2023 and 2024, the lengthy commissioning and tuning process provided an invaluable opportunity to measure subtle and drastic changes in the hall's form and materiality for comparison to the predictive simulation. Drawing from the Lindemann set, the paper compares modeled and measured data from the PAC to that from other projects designed using similar tools. The paper identifies points of uncertainty during design and delves into commonalities and differences between ray-based simulation and measurement in performance spaces of varying form. This paper will address the inherent complexities of approximating acoustic behavior with side by side comparison of measurement and simulated data, we will share insights that inform our future modeling process.

10:50

2aAAa9. Measurements using three methods. K. Anthony Hoover (McKay Conant Hoover, 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362, thoover@mchinc.com) and Henry Ashburn (McKay Conant Hoover, Westlake Village, CA)

Room responses of various spaces were measured using three methods in each: interrupted pink noise using full-range loudspeakers, impulse using balloon pops, and sine sweep using dodec loudspeaker array. Our primary interest was to compare the results from each, as well as to better understand their relative advantages and disadvantages. Results will be shown and discussed.

11:10

2aAAa10. Comparison of reverberation times and average absorption coefficients in classrooms, based on *in situ* measurements versus calculated from material databases as during design. Sanjay Kumar (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, Omaha, NE) and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, PKI 100C, 1110 S. 67th St., Omaha, NE 68182-0816, lwang4@unl.edu)

Reverberation time (RT) is an acoustic descriptor used widely by acousticians for characterizing indoor acoustic environments, as specified in standards such as ANSI S12.60 “Acoustical Performance Criteria, Design Requirements, and Guidelines for Schools”. It is often calculated in the design phase for simple spaces by applying equations like Sabine’s or Eyring’s, requiring information on room dimensions and the surface areas and acoustic absorption coefficients across frequency of room materials. In this project, measured RTs from a recent survey of over 200 K-12 classrooms are compared against the RTs predicted across speech frequency octave bands using Sabine’s and Eyring’s equations. Agreement between predicted and measured values varies based on the octave band and material selection. The working group currently revising ANSI S12.60 is considering providing guidance on the desired average absorption coefficient in classrooms, rather than reverberation time, so this study also compares the average absorption coefficients for the classrooms. The coefficients are back-calculated from the measured RTs using Sabine’s and Eyring’s equations, and compared against those calculated from surface dimensions and commonly available material absorption coefficients.

Contributed Papers

11:30

2aAAa11. Exploring variations to ASTM E336 directional sound source positioning. Pier-Gui Lalonde (Integral DX Eng. Ltd., 907 Admiral Ave., Ottawa, ON K1Z 6L6, Canada, pier-gui@integraldxengineering.ca) and Gregory E. Clunis (Integral DX Eng. Ltd., Ottawa, ON, Canada)

This paper explores variations to the procedures presented in ASTM E336 *Standard Test Method for Measurement of Airborne Sound Attenuation between Rooms in Buildings*. Per the Test Standard, directional sound sources must be aimed into corners away from the test partition. For source rooms horizontally adjacent to occupied spaces on all sides, multiple loudspeaker positions are needed to measure noise isolation in all directions. Measurements made in 4 office spaces are presented, each made in two ways: firstly in compliance with the Standard; and secondly, with sound sources orientated and operated differently, to explore whether using a single configuration of directional loudspeakers can produce similar results. Variations in the results are discussed, and suggestions are provided for future work.

11:45

2aAAa12. Comparing measured sound strength to theoretical sound strength for rooms of varying size. Joshua Dunham (Stantec, 720 Third Ave., Seattle, WA 98104, Joshua.Dunham@stantec.com)

ISO Standard 23591 shows how the sound pressure level in a room can be estimated with the following two pieces of information: the sound power of sources in that room, and the sound strength G of that room as defined in ISO 3382-1. In an earlier study, the theory connecting room volume V, reverberation time T, and sound strength G was summarized, and a comparison of measured G to this theory was explored for a wide range of rooms. That test setup used Odeon software. In the present study, a different test setup is used implementing Dirac software to expand G testing to additional rooms of various sizes for evaluation under the same mathematical model. After a description of the equipment using Dirac software is discussed, a brief review of the additional rooms in which sound strength was measured is provided. Both measured and theoretical sound strength results are presented for these rooms. The difference between measured G and theoretical G using the mathematical model is analyzed statistically both for the Dirac test setup and compared to the results from the previous study.

Session 2aAAb

Architectural Acoustics: Student Design Competition (Poster Session)

Robin Glosemeyer Petrone, Chair
Threshold Acoustics, 141 W Jackson Blvd, Suite 2080, Chicago, IL 60604

This competition is intended to encourage students in the disciplines of architecture, engineering, physics, and other curriculums that involve building design and/or acoustics to express their knowledge of architectural acoustics and noise control in the design of a facility in which acoustical considerations are of significent importance.

Design Scenario: A college with a very strong music and vocal program intends to construct a new 1,200-seat performance hall primarily for opera. Although the main purpose of the hall is to support their opera program, the hall will also be used for speaking engagements by the school's president and other invited speakers.

Entries to the 2024 competition will be posted in this session for viewing.

TUESDAY MORNING, 14 MAY 2024 ROOM 204, 8:00 A.M. TO 11:10 A.M.

Session 2aAB

Animal Bioacoustics, Computational Acoustics, Acoustical Oceanography, Underwater Acoustics, and Signal Processing in Acoustics: Data to Information, Navigating the Application of Acoustic Data for Conservation I

Megan F. McKenna, Cochair
Cooperative Institute for Research in Environmental Sciences, University of Colorado Boulder, 442 Gibson Ave., Pacific Grove, CA 93950

Carrie Wall, Cochair
University of Colorado, 325 Broadway, Boulder, CO 80305

Chair's Introduction—8:00

Invited Papers

8:05

2aAB1. A primer on terrestrial passive acoustic monitoring data management: Best practices and guidelines. Dena J. Clink (K. Lisa Yang Ctr. for Conservation Bioacoustics, Cornell Univ., 159 Sapsucker Woods Rd., Ithaca, NY 14850, djc426@cornell.edu), Chris Pelkie, Russell A. Charif, and Holger Klinck (K. Lisa Yang Ctr. for Conservation Bioacoustics, Cornell Lab of Ornithology, Cornell Univ., Ithaca, NY)

Recent advances in recording technology and data storage capabilities have revolutionized how we monitor vocal animals and their habitats. Recently, there has been an explosion of interest in the use of terrestrial passive acoustic monitoring (PAM) which relies on autonomous acoustic recording units (ARU), but terrestrial PAM studies are limited by a lack of standardized protocols for data collection, analysis, and archiving. Here, we draw on collective experiences of recording, analyzing, and archiving, hundreds of years of PAM data to provide guidelines and recommendations for PAM practitioners. First, we highlight some of the major pitfalls in field data collection, including inappropriate recording settings for the research question and incomplete metadata. We then provide recommendations for data management, including suggestions for backing up, compressing, and archiving the data. Our goal is to provide the community with standardized working guidelines that will help facilitate comparative, large-scale terrestrial studies using PAM.

8:25

2aAB2. Using acoustic monitoring to evaluate the co-benefits of urban restoration. Rachel T. Buxton (Inst. of Environ. and Interdisciplinary Sci., Carleton Univ., 1125, Colonel By Dr., Ottawa, ON K1S5B6, Canada, Rachel.buxton@carleton.ca), Christopher Dennison (Dept. of Biology, Carleton Univ., Ottawa, ON, Canada), Katherine Brown, and Amber L. Pearson (CS Mott Dept. of Public Health, Michigan State Univ., Flint, MI)

Restoring urban greenspaces is an increasingly common nature-based solution aiming to bolster biodiversity while benefiting human wellbeing. However, there is limited understanding of the co-benefits, including potential trade-offs and synergies between outcomes for nature and people. Although passive acoustic recording may be a valuable outcome monitoring tool, appropriate analytical techniques are unknown. We explore how different acoustic analyses capture patterns in species diversity and harmful noise pollution using 2021–2023 acoustic data from Detroit, Michigan. Data were collected as part of a longitudinal panel study aimed at quantifying the effects of park restoration on physical activity, stress, and cardio-metabolic health. We used AudioMoth recording devices at 100 sites: 10 in the neighborhood around each of 5 restored parks, where native plants were seeded, and 10 around each of 5 unmaintained control parks, matched to restored parks based on specified conditions. Recordings were analyzed by manually identifying bird species and noise in a subset of spectrograms, using an artificial neural network (BirdNET), and using acoustic indices. We found that using acoustic indices in a model could accurately predict bird species diversity and richness, but not community composition. We compared outputs from each type of analysis with mental health indicators of study participants.

8:45

2aAB3. Navigating the bioacoustic landscape: Standardization and interoperability of acoustic data products. Marie A. Roch (Dept. of Comput. Sci., San Diego State Univ., 5500 Campanile Dr., Dept. of Comput. Sci., San Diego State University, San Diego, CA 92182-7720, marie.roch@sdsu.edu), Simone Baumann-Pickering (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA), Douglas Gillespie (Sea Mammal Res. Unit, Univ. of St Andrews, St Andrews, Scotland, United Kingdom), Jasper Kanes (Sci. Services, Ocean Networks Canada, Victoria, BC, Canada), Katherine Kim (Greeneridge Sci., Inc., Santa Barbara, CA), Holger Klinck (K. Lisa Yang Ctr. for Conservation Bioacoustics, Cornell Lab of Ornithology, Cornell Univ., Ithaca, NY), Xavier Mouy (Passive Acoust. Res. Group, NOAA, Miami, FL), and Ana Sirovic (Norwegian Univ. of Sci. and Technol. (NTNU), Trondheim, Norway)

The importance of bioacoustics for understanding and protecting the natural world has grown significantly over the last few decades. Our capacity to monitor the acoustical landscape has increased with lower cost instrumentation, improved communication, infrastructure, and storage, as well as the development of reliable machine learning methods for analysis. Understanding long-term trends in animal populations as well as the impacts of anthropogenic noise requires the ability to retain and interpret records over decadal time scales, record in detail what type of analysis has been performed and understand when multiple analyses may be combined and when they should not be. This is not possible without a well-defined set of terms and meanings. In this talk, we will provide an overview of the work of Acoustical Society of America's working group S3-SC1-WG7 that has been charged with developing an American National Standards Institute standard for bioacoustics information derived from acoustic recordings. The proposed standard details what information should be recorded for bioacoustics recording and analysis effort, ranging from specification of data to be noted about instrumentation to information gleaned from recordings such as noise levels, animal calls, and acoustic source locations.

9:05

2aAB4. Five years of data collection, student theses, and disseminating information soundscape studies in the Gulf of Tribugá, Colombia. Kerri D. Seger (Appl. Ocean Sci., 2127 1/2 Stewart St., Santa Monica, CA 90404, kerri.seger.d@gmail.com), Christina E. Perazio (Environment and Sustainability, Univ. at Buffalo, Buffalo, NY), Nohelia Farias Curtidor (Fundación Macuáticos Colombia, Bogotá, Colombia), L. V. Huertas-Amaya (Ports, Humpbacks, y Sound In Colombia (PHySIC), Bogotá, Colombia), Maria P. Rey-Baquero (Fundación Macuáticos Colombia, Bogota, Colombia), Daniel Norena (Ports, Humpbacks, y Sound In Colombia (PHySIC), Bogota, Bogotá DC, Colombia), Mar Palanca (MadreAgua Colombia, Valencia, Spain), Astarte Brown (Ecology & Evolutionary Biology, Univ. of California, Santa Cruz, Santa Cruz, CA), Valentina Ramirez Caycedo (Ports, Humpbacks, y Sound In Colombia (PHySIC), Bogotá, Colombia), and Natalia Botero-Acosta (Fundación Macuáticos Colombia, Bogotá, Colombia)

In 2018, the Ports, Humpbacks, y Sound In Colombia (PHySIC) Project began to record soundscapes in the Gulf of Tribugá, Northern Colombian Pacific, for the first time. This was of interest to local conservation groups who were against building a megaport in the native mangrove habitat that provides livelihoods for the people of Chocó. Five years later, no port has been built, documentaries about the ecosystem have won awards, before-after/control-impact studies have been done on fish and humpback whale acoustic behavior, delphinid call catalogs have been assembled, propagation modeling has mapped acoustic connectivity, and a new vocalization category for humpback whales has been discovered. The work has been executed through internships, and undergraduate and masters theses at various universities. This project has a variety of data-to-information challenges: storing data for access across various countries, navigating data analysis and dissemination in two languages, transferring knowledge as students graduate, and tailoring results for general conservation groups and policymaking audiences. To celebrate the colleagueship and academic successes of the first five years of PHySIC, this talk will describe the list of discoveries and studies done in the Gulf of Tribugá, which was named a UNESCO World Heritage site in the summer of 2023.

9:25

2aAB5. From sounds to science on public lands: Using emerging tools in terrestrial bioacoustics to understand national park soundscapes. Cathleen Balantic (National Park Service, 1201 Oakridge Dr. Ste. 100, Fort Collins, CO 80525, cathleen_balantic@nps.gov)

The National Park Service manages over 400 units across the United States, each with unique resource protection needs. Many parks have natural resource conservation priorities that include the protection of natural soundscapes and the sound-producing species who contribute to them. It is challenging, however, to translate terabytes of raw audio into applied information for the diverse research and management questions posed by individual parks. Recent advances in terrestrial bioacoustics are making this process more tractable, particularly as BirdNET has emerged as an extensible off-the-shelf tool equipped to detect diverse assemblages of sound-producing species across varied park ecosystems. We illustrate the applied utility of using and adapting this emerging tool in combination with open code and visualizations to answer multi-taxa questions about occurrence, phenology, and wildlife responses to management. This work demonstrates both the iterative nature and challenges inherent to keeping a federal agency astride the leading edge in a rapidly progressing, data-heavy field, amid a backdrop of global change.

9:40–9:55 Break

9:55

2aAB6. Methods for managing soundscape quality: Evidence from Denali National Park and Preserve. Lauren A. Ferguson (Recreation Management and Policy, Univ. of New Hampshire, 4 Library Way, Hewitt Hall, Durham, NH 03824, lauren.ferguson@unh.edu), Peter Newman (Recreation, Parks and Tourism Management, The Penn State Univ., State College, PA), Megan F. McKenna (Cooperative Inst. for Res. in Environ. Sci., Univ. of Colorado Boulder, Pacific Grove, CA), Davyd H. Betchkal (Natural Sounds and Night Skies Div., National Park Service, Denali Park, AK), Zach Miller (Visitor Use Management, National Park Service, Lakewood, CO), Rose Keller (Norwegian Inst. for Nature Res., Lilehammer, Norway), Kurt Fristrup (Natural Sounds and Night Skies Div., National Park Service, Fort Collins, CO), and Derrick Taff (Recreation, Parks and Tourism Management, The Penn State Univ., State College, PA)

The purpose of the United States national park system is to preserve resources unimpaired for the enjoyment of future generations. At Denali National Park (DENA), high levels of air tour traffic and air transportation present challenges to preserving wilderness character. Accordingly, the park attached high priority to measuring visitor impressions of soundscape quality. Visitors in four distinct settings rated the acceptability of five randomly chosen recordings of aircraft noise. We used a cumulative link mixed model to fit visitor's response to acoustic and nonacoustic factors. The negative effects of increased noise were moderated for visitors who were interested in taking an air tour. To be conservative, we examined visitors uninterested in an air tour and found the probability of rating aircraft noise as unacceptable at 54 dB LAeq, 30 s or higher was 26%. For context, noise above 55dB is incompatible with outdoor, rural activities. Visitor response predictions were joined to a spatial model of aircraft noise propagation to create a map depicting the acceptability of aircraft noise in DENA's frontcountry area. The map can be used to forecast the range of soundscape conditions park visitors are exposed to, inform hiking recommendations for visitors, and evaluate park management zones.

10:10

2aAB7. Investigation of duty cycles for measuring activity in passive acoustic bat monitoring. Aditya Krishna (Elec. and Comput. Eng., Univ. of Washington, 1013 NE 40TH St., Henderson Hall (APL) Rm. No. 459, Seattle, WA 98105, adkris@uw.edu) and Wu-Jung Lee (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Echolocating bats are important bioindicators that can be monitored effectively using passive acoustic monitoring (PAM) techniques. In PAM, duty-cycle-based temporal subsampling is often used to collect data at ON/OFF intervals to circumvent the limitations of recorder battery and storage capacity to enable long-term monitoring. However, potential bias introduced by temporal subsampling has not been systematically investigated for bat monitoring. Here, we use continuous audio recordings from the Union Bay Natural Area in Seattle in Summer 2022 to simulate the effects of temporal subsampling using different duty cycle parameters. We detected bat calls automatically using a deep learning model [Aodha *et al.*, 2022, BioRxiv] and calculated three metrics as proxy for bat activity: number of calls, Activity Index (AI), and Bout Time Percentage (BTP). We found that although the number of calls and AI are more readily computable using the detected calls, BTP is likely a more accurate measure that relies less on the performance of automated call detection methods. In addition, reduced sampling for only a portion of the night (e.g., 4 hrs) was generally inadequate for capturing bat activity. Our results suggest that considering species-specific acoustic characteristics is crucial for reducing sampling bias for PAM of bats.

10:25

2aAB8. The past, present and future of underwater passive acoustic monitoring. John Hildebrand (Scripps Inst. of Oceanogr., Univ. of California San Diego, Mail Code 0205, La Jolla, CA 92093, jhildebrand@ucsd.edu), Sean Wiggins, Simone Baumann-Pickering, Kait Frasier (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA), and Marie A. Roch (San Diego State Univ., San Diego, CA)

We review the history of underwater passive acoustic monitoring and predict the future trajectory. Over the past three decades, advances in digital data storage capacity and low power electronics have made it possible to collect autonomous long-term broadband passive acoustic monitoring data. Concomitant advances have taken place in data analysis and curation. Standardized spectra are an excellent first-step in analysis, calculated for all data using multiple frequency bands. These spectra allow *in-situ* instrument calibration based on an understanding of underwater ambient noise. Software for efficient manual scanning and signal discovery is used for verification and error estimation. Automatic detectors/classifiers are obtained from both supervised and unsupervised machine learning. Detections are aggregated into a database that allows the combination of multiple datasets and association with environmental or other data. Future work will increasing use of arrays of acoustics sensors, the integration of passive acoustics with other sensors, and machine learning that integrates these data streams to provide better understanding of anthropogenic, biological and physical processes in the ocean.

10:40

2aAB9. Connecting separate monitoring programs through the SoundCoop. Carrie Wall (Univ. of Colorado, 325 Broadway, Boulder, CO 80305, carrie.wall@noaa.gov), Leila Hatch (ONMS, NOAA, Scituate, MA), Sofie Van Parijs (NEFSC, NOAA, Woods Hole, MA), Rob Bochenek (Axiom Data Sci., Anchorage, AK), Catherine Berchok (AFSC, NOAA, Seattle, WA), Genevieve Davis (NEFSC, NOAA, Woods Hole, MA), Peter Dugan (NUWC, Middeltown, RI), Kait Frasier (Scripps Inst. of Oceanogr., La Jolla, CA), Samara Haver (Oregon State Univ., Newport, OR), Megan F. McKenna (Cooperative Inst. for Res. in Environ. Sci., Univ. of Colorado Boulder, Pacific Grove, CA), Jennifer Miksis-Olds (Univ. of New Hampshire, Durham, NH), Clea Parcerisas (Flanders Marine Institute/Ghent Univ., Orstend, Belgium), Dimitri Ponirakis (Cornell Univ., Ithaca, NY), Timothy Rowell (SEFSC, NOAA, Beaufort, NC), John P. Ryan (Res., MBARI, Moss Landing, CA), and Karolin Thomisch (Alfred Wegener Inst. for Polar and Marine Res., Bremer, Germany)

Passive acoustic monitoring (PAM) data collection has been growing exponentially, resulting in petabytes of data that document ocean soundscapes, how they change over time, and what animals use these ecosystems at varying timescales. Efficiently extracting this critical information and comparing it to other datasets in the context of ecosystem-based management is a Big Data challenge that traditional desktop processing methods cannot address. The curation, management, and dissemination of PAM datasets is another challenge in need of collaborative progress. To meet these exigencies, a multi-agency funded Sound Cooperative (SoundCoop) project is building community-focused, national cyberinfrastructure capability for PAM data to promote improved, scalable and sustainable accessibility and applications for management and science. Driven by partnerships and framed by four case studies, the SoundCoop has established guidance on the standardized processing of sound level metrics using free software toolkits and begun developing core cyberinfrastructure components that future PAM projects can leverage. U.S. and international scientists contributed PAM data collected across 10 long-term monitoring projects to operationalize the production of hybrid-millidecade spectra across a diversity of labs/instruments. Collectively, the contributed data demonstrate the value of standardized processing that enables the creation of comparable results from disparate monitoring efforts.

10:55

2aAB10. Understanding variability in coral reef soundscapes to enable tool development and early warnings of ecosystem change. T. Aran Mooney (Biology Dept., Woods Hole Oceanographic Inst., 266 Woods Hole Rd., Marine Res. Facility 227 (MS# 50), Woods Hole, MA 02543-1050, amooney@whoi.edu), Sierra Jarriel, Nathan Formel, Nadége Aoki (Biology Dept., Woods Hole Oceanographic Inst., Woods Hole, MA), Seth A. McCammon (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA), Amy Apprill (Marine Chemistry and Geochemistry, Woods Hole Oceanographic Inst., Woods Hole, MA), David Mann (Loggerhead Instruments, Sarasota, FL), and Yogesh Girdhar (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

Coral reefs harbor some of the highest biodiversity on Earth. Their rich soundscape is vital to inhabiting animals and can provide a means of tracking community health. Reefs are facing immense climate stressors and are declining rapidly. Passive acoustic monitoring can provide a powerful, scalable tool for stakeholders, but these data are not evaluated on actionably relevant timescales. Here we present results from a long-term (10-years) acoustic and ecosystem study of multiple coral reefs in the U.S. Virgin Islands. Acoustic measurements (including snap rates, sound levels in low-frequency fish, and high-frequency shrimp bands) were made in-tandem with traditional, diver-based benthic and fish visual surveys. These key baseline data over multiple spatial and temporal scales provide a means of examining how soundscape changes are driven by climate-related stressors, and an important testbed for developing new analyses and tools. Here, we show how physical changes (temperature, light, coral disease) can influence the cue rates of snapping shrimp and fish, and apply novel tools including real-time recorders and underwater robots listening for biodiversity. These initial steps were then implemented into a novel rapid acoustic assessment, a key step toward providing actionable information to stakeholders monitoring for habitat change and weighing resource management.

Session 2aAO

Acoustical Oceanography: Topics in Acoustical Oceanography I

John A. Colosi, Cochair
Oceanography, Naval Postgraduate School, Department of Oceanography, Naval Postgraduate School, Monterey, CA 93943

Julien Bonnel, Cochair
Woods Hole Oceanographic Institution, 266 Woods Hole Rd, MS# 11, Woods Hole, MA 02543-1050

Contributed Papers

8:30

2aAO1. Theoretical model for correlation analysis of wideband signal received by two vertical arrays and retrieval of modal dispersion curves in shallow water. Marina Yarina (Marine Geoscience, Univ. of Haifa, Abba Khoushy 199, Haifa 3498838, Israel, myarina@campus.haifa.ac.il), Boris Katsnelson (Marine Geosciences, Univ. of Haifa, Haifa, Israel), and Oleg A. Godin (Phys., Naval Postgrad. School, Monterey, CA)

Goal of this work is to carry out theoretical modeling extraction of characteristics of layered bottom (sound speeds $c_{;2}$ in layer, c in water, $c_{;b}$ in half-space and layer's thickness h using construction of correlation function (matrix) of wideband synthetic sound signal radiated by the moving source and imitating shipping noise. This work's motivation was experiment carried out by authors in shallow water. Combination of two vertical line arrays acts as a diffraction grating with 2N hydrophones that uses interference pattern in vertical and horizontal directions and gives more information than one-dimensional array or single receiver. Our waveguide model constitutes water layer above low-speed thin layer ($c_{;2} \ll c$, h is about a few meters) and half space ($c_{;b} > c$). In this waveguide there are two sorts of modes: in dependence on frequency and thickness of sediment's layer– modes with vertical dependence distributed in water layer and modes trapped in the near bottom layer. It is shown that correlation function constructed in coordinates (phase speed, frequency) allows us to retrieve dispersion curves and to estimate critical frequencies—determining creation of trapped modes, which in turn give parameters of layered bottom. [Work was supported by ISF, Grant No. 946/20.]

8:45

2aAO2. Kauai Beacon receptions and analysis with open-access hydrophones in the North Pacific Ocean. John Ragland (Elec. and Comput. Eng., Univ. of Washington, 185 W Stevens Way NE, Seattle, WA 98195, jhrag@uw.edu), Nicholas Durofchalk (Phys., Naval Post Graduate School, Monterey, CA), David Dall'Osto (Appl. Phys. Lab. at the Univ. of Washington, Seattle, WA), Kay L. Gemba (Phys., Naval Post Graduate School, Monterey, CA), and Shima Abadi (Univ. of Washington, Seattle, WA)

The Kauai Beacon began regularly transmitting a 75 Hz, maximum length sequence encoded, signal that has been received by hydrophones as far as 4000 km from the source. The received signal, which has travelled across the entire ocean basin, contains integrated environmental information about the full path between the transmitter and the receiver. This allows us to simultaneously understand how the environment effects long-range acoustic propagation, and to use the long-range acoustic propagation to estimate environmental variables such as path-averaged ocean temperature. In this talk, we present continued work to analyze the receptions of the Kauai Beacon using open-access ambient sound data recorded in the North Pacific Ocean. We present long-term trends of acoustic arrival times, the spread of the acoustic energy, and frequency shift of the signal due to mooring motion, or ocean state fluctuation. [Work supported by ONR.]

9:00

2aAO3. Acoustic tomography in a deep stratified lake; toward reliable estimates of large-scale currents by differential travel time. John C. Wells (Civil & Environ. Eng., Ritsumeikan Univ., Noji Higashi 1-1-1, Tricea I, Civil & Env'l Eng., Kusatsu, Shiga 525-8577, Japan, jwells@se.ritsumei.ac.jp) and Naokazu Taniguchi (Graduate School of Adv. Sci. and Eng., Hiroshima Univ., Higashi-Hiroshima, Japan)

Lakes worldwide are under increasing stress. For example climate change tends to strengthen stratification, promoting hypoxia and other nefarious effects. Targeting scientific investigation of such changes, and operational monitoring, we have been developing the lacustrine application of Coastal Acoustic Tomography ("CAT") at Lake Biwa. The largest lake in Japan, Biwa supplies water to 14 million, and has a mean depth in its main basin of 41 m. Since November 2016, we have used M-sequences (5 kHz carrier) to execute the first successful tests of CAT at multikilometer ranges in any lake, with multiweek deployment of 3 (5) stations in Nov 2017 (2018). In 2021, we confirmed reciprocal transmission in spring through early summer between two stations at 7 km. In autumn, we often observed two distinct groups of arrivals. Ray-tracing simulations, based on sound-speed distributions from realistic hydrodynamic simulations, revealed a warm "surface channel" and cold "deep channel" whose predicted arrival times corresponded well with the observations. In continuing work to be reported, we are targeting reliable measurement of raypath-averaged currents by differential travel time, as already achieved by the second author's team for the stronger tidal currents in the Seto Inland Sea at ranges of 3 km. Preliminary processing has revealed continuous evolution of phase difference between reciprocal receptions. By exploiting such phase differences we hope to achieve a useful level of accuracy in differential travel time despite the weak currents.

9:15

2aAO4. Trans-dimensional joint inversion of modal amplitude and arrival-time data for geoacoustic properties in layered muddy sediments. Yong-Min Jiang (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC V8W2Y2, Canada, minj@uvic.ca), Stan Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada), and Julien Bonnel (Woods Hole Oceanographic Inst., Woods Hole, MA)

Long-range sound fields in a shallow-water waveguide can be expressed as a summation of several dispersive propagating modes. Modal arrival times as a function of frequency, extracted via warping time-frequency analysis, have been used by many researchers to estimate sound speeds and densities for layered models of the seabed sediments. This paper explores joint inversion of modal amplitude spectra together with arrival times to also estimate sediment-layer attenuations. To consider the common case in underwater-acoustic applications where the source amplitude spectrum is unknown, we include this spectrum as unknown parameters in the inversion, formulated implicitly within the likelihood function. Trans-dimensional

Bayesian inversion is applied to estimate marginal probability profiles of the geoacoustic properties, providing quantitative uncertainties. The data considered here were collected on the New England Mud Patch during the 2017 Seabed Characterization Experiment. [Work supported by the Office of Naval Research.]

9:30

2aAO5. Acoustic propagation through a biological deep scattering layer at the New England shelf break. Natalie Kukshtel (Appl. Ocean Phys. & Eng., Woods Hole Oceanographic Inst., 98 Water St., Woods Hole, MA 02543, nkukshtel@whoi.edu), Andone C. Lavery (Appl. Ocean Phys. & Eng., Woods Hole Oceanographic Inst., Woods Hole, MA), and Ying-Tsong Lin (Scripps Inst. of Oceanogr., La Jolla, CA)

The New England Shelf Break is a dynamic ocean environment with strong spatial variability due to complex physical processes and interactions with warm core rings originating from the Gulf Stream. An acoustic autonomous underwater vehicle (AUV) was deployed in the New England Shelf Break Acoustics (NESBA) experiment in May 2021 to investigate environmental variability and its effect on sound propagation. The AUV system was comprised of a modified REMUS 600 vehicle with a hull-mounted 3.5 kHz transducer, a downward-facing EK80 echosounder, and a towed linear hydrophone array. Shipboard EK80 data measured during the AUV mission showed the presence of a biological deep scattering layer. Preliminary analysis of AUV source signals passing through the layer suggested significant attenuation compared to signals not passing through the layer. To investigate this further, a parabolic equation (PE) model with range-dependent attenuation patches in the water column was constructed to replicate the biological attenuation through the scattering layer. The model uses estimates of absorption and scattering derived from shipboard and AUV EK80 data. Through this study we aim to better understand the scattering effects of this deep biological layer on acoustic propagation. [Work supported by the Office of Naval Research.]

9:45–10:00 Break

10:00

2aAO6. Mid-frequency acoustic tracking of breaking waves. Ryan Saenger (Scripps Inst. of Oceanogr., Univ. of California San Diego, 9500 Gilman Dr., San Diego, CA 92093, rsaenger@ucsd.edu), Luc Lenain, William Kuperman, and William Hodgkiss (Scripps Inst. of Oceanogr., Univ. of California San Diego, San Diego, CA)

Large surface wave breaking events in deep water are acoustically detectable by beamforming at 5-6 kHz with a mid-frequency planar array located 130 m below the surface. Due to the array's modest 1 m horizontal aperture, wave breaking events cannot be conveniently tracked by beamforming alone. However, their trajectories can be estimated by splitting the array into left-right and upper-lower sub-array pairs, beamforming each sub-array toward the source, and computing the left-right and upper-lower beam cross-correlations. The cross-correlations can be used to estimate the relative time delay between sub-arrays of noise generated by breaking waves. The time delays map to a source location on the surface that can be tracked over time. Source tracks estimated from the sub-array cross-correlations match the trajectories of breaking waves that are visible in aerial images of the ocean surface above the array. [Work supported by the Office of Naval Research.]

10:15

2aAO7. Listening to the acidity of the ocean: Inversion of passive deep sea acoustic data for pH. Ernst Uzhansky (Oceanogr., Dalhousie Univ., Izhaq Greenboim Str., Apt. 12, (Koren Family), Haifa 3498793, Israel, ernstuzhansky@gmail.com) and David R. Barclay (Oceanogr., Dalhousie Univ., Halifax, NS, Canada)

Ocean acidification is an ongoing concern due to its impact on the marine ecosystem. The volume integrated pH of sea water can be determined from the depth-dependence of ambient sound, which depends on the acoustic absorption properties of seawater. For a wind-driven noise in the ocean over the band 1–10 kHz, two main contributions to sound attenuation are associated with the ionic relaxation of boric acid ($<$3 kHz), related to pH, and magnesium sulfate ($>$3 kHz), unrelated to pH. When local winds are strong ($>$10 m/s), the ambient noise is dominated by locally generated surface noise and has a depth-independent directionality and a weakly frequency and depth-dependent intensity, due to sound absorption. By measuring the attenuation of sound in a wide frequency band, it is possible to estimate pH by comparing the experimentally measured attenuation with an analytical theory of passive acoustic absorption spectroscopy. Measurements of the depth-dependent ambient sound field were carried out in the Philippine Sea, Mariana Trench, and Tonga Trench throughout 2009—2021. The wideband (5 Hz—30 kHz) acoustic data were recorded with untethered free-falling autonomous recording systems carrying two or four hydrophones.

10:30

2aAO8. Acoustic measurements of photosynthetically formed gas bubble distributions in Posidonia oceanica seagrass meadows. Anthony P. Lyons (Univ. of New Hampshire, Durham, NH, anthony.lyons@unh.edu), Angeliki Xenaxi, and Yan Pailhas (Ctr. for Maritime Res. and Experimentation, La Spezia, Italy)

The formation of gas bubbles on seagrass leaves commonly occurs during periods of high productivity in many aquatic ecosystems. Gas bubbles produced by seagrass meadows and subsequently released to the atmosphere can be a major component of primary production that is not quantified by measurements of dissolved oxygen. Acoustic propagation measurements have been explored as a possible tool for estimating photosynthetically relevant properties *in situ*. However, a lack of knowledge of bubble densities and size distributions has hindered acoustic attenuation as a quantitative measurement technique. Here we explore using measurements of acoustic backscattering from gas bubbles in and above Posidonia oceanica seagrass meadows as a more direct method for estimating bubble densities and size distributions. Acoustic backscattering data for this study was collected in September 2023 in Biodola Bay, Elba Island, Italy, using 1i–16 kHz linear frequency modulated pulses. Inversions of the broadband acoustic data yielded log-normal bubble size distributions with mean sizes between 0.4 and 0.5 mm in agreement with previously reported values. Estimated bubble densities yielded predicted attenuations in line with those found by propagation experiments. Results demonstrate the possible use of direct acoustic backscattering measurements as a supplement to dissolved oxygen measurements to better estimate rates of gas production which in turn can be used to better understand seagrass ecosystem properties, such as productivity and health.

10:45

2aAO9. Acoustic observations of individual bubble release events from melting glacier ice in an arctic fjord. Hayden A. Johnson (Scripps Inst. of Oceanogr., Univ. of California, San Diego, 8622 Kennel Way, La Jolla, CA 92037, h3johnso@ucsd.edu), Grant B. Deane (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA), Oskar Glowacki (Inst. of Geophys., Polish Acad. of Sci., La Jolla, CA), M. Dale Stokes (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA), Mandar Chitre, Hari Vishnu (Acoust. Res. Lab., National Univ. of Singapore, Singapore, Singapore), and Elizabeth Weidner (Scripps Inst. of Oceanogr., Univ. of California, San Diego, Durham, NH)

Melting below the waterline of marine-terminating glaciers and ice sheets has the potential to remove ice that is holding back land-based ice upstream, and increase the rate of global mean sea level rise. It is difficult to observe this melting directly, and the physical processes involved remain an active area of study. As glacier ice melts, the air bubbles trapped in the ice are released into the water. The air bubbles are often at greater pressure than the surrounding hydrostatic pressure in the water, causing them to be released explosively and generate sound through resulting monopole oscillations. Studying this sound at the level of individual bubbles provides an opportunity to learn more about the role of the bubbles in affecting heat flux across the ice-water boundary layer. It is also essential for proposed efforts to use acoustic emissions as a tool to quantify glacier-scale submarine melting. We will present and discuss observations made at the ice-water interface of floating pieces of glacier ice in an Arctic fjord. These observations included measurements of the water thermal structure, velocity, and salinity, as well as ablation of the ice face, made concurrently with acoustic measurements of bubble release pulses from a 2-element hydrophone array.

11:00

2aAO10. Measurements of the frequency dependence of geo-alpha in clayey silt at low frequencies. Orest Diachok (Poseidon Sound, 3272 Fox Mill Rd., Oakton, VA 22124, orestdia@aol.com) and Altan Turgut (Naval Res. Lab., Washington, DC)

Transmission loss (TL) measurements in the Santa Barbara Channel, together with co-located chirp measurements, show that geo-alpha (dB/λ) in clayey silt increases approximately linearly with frequency (f) at $f < 3$ kHz. TL was measured between a fixed source and fixed vertical array at 3.7 km at $0.3 < f < 5$ kHz. Geo-alpha was inverted from the TL data at 0.3, 1.1 and 1.9 kHz. Inversion calculations at 1.1 and 1.9 kHz considered both geo-alpha and bio-alpha due to layer of anchovies at 12 m. Inversion calculations at 0.3 kHz considered only the effect of geo-alpha; the effect bio-alpha on TL was small. Geo-alpha was also inferred from chirp sonar measurements at 3.2 and 6.2 kHz at the ends and middle of the TL track. Chirp sonar showed that the bottom consisted of a layer of unconsolidated sediments overlying a layer of sand. Co-located cores revealed that the unconsolidated layer consisted of clayey silt. Bio-alpha data were compared to Biot and VGS theories and the theory of attenuation in suspended sediments (Pierce *et al.*, 2017). The measured frequency dependence is consistent with Pierce *et al.*'s theory. This research was supported by The Office of Naval Research Ocean Acoustics Program.

11:15

2aAO11. Angular dependence of acoustic scattering statistics for rocky outcrops. Lindsey Darling (Oceanogr., Naval Postgrad. School, 1 University Circle, Monterey, CA 93943, lindsey.darling@nps.edu), Derek Olson (Oceanogr., Naval Postgrad. School, Monterey, CA), and Marc Geilhufe (Norwegian Defence Res. Establishment, Kjeller, Norway)

The use of automated detection algorithms in undersea remote sensing and target detection allows for decreased timelines and risk to human assets. However, understanding the performance of these systems requires a thorough understanding of the physical processes that affect the acoustic scattering statistics of the sonar image. Many studies have reported on the scattering statistics of the seafloor generally, and recent work has quantified the dependence of scintillation index on range for sandy seafloors. To date, no studies have examined the angular dependence of acoustic scattering statistics for rocky bottoms, which have more complex spatial texture. This paper analyzes synthetic aperture sonar (SAS) images of rocky outcrops imaged near Bergen Norway to quantify the grazing angle dependency of scattering statistics for rocky seafloors. Outcrops are categorized by texture and analyzed using scintillation index and relative scattering strength with respect to both true grazing angle and a flat seafloor assumption. Comparisons between results utilizing calibrated (beampattern removed) and non-calibrated images will be presented to display the complexity of the grazing angle dependence of scattering statistics for rocky seafloors. SAS imagery provided by the Norwegian Defence Research Establishment. [Funding provided by the Office of Naval Research]

TUESDAY MORNING, 14 MAY 2024 — ROOM 212, 7:55 A.M. TO 12:00 NOON

Session 2aBAa

Biomedical Acoustics, Physical Acoustics, Signal Processing in Acoustics, and Structural Acoustics and Vibration: Ultrasound Beamforming and its Applications I

Jian-yu Lu, Chair
Bioengineering, The University of Toledo, 2801 West Bancroft Street, Toledo, OH 43606

Chair's Introduction—7:55

Invited Papers

8:00

2aBAa1. Some thoughts on ultrasound futures. Kai E. Thomenius (Dept. of Radiology, Massachusetts General & Brigham, 74 Van Vranken Rd., Clifton Park, NY 12065, kaius@mac.com)

Development of ultrasound technology began in early 1950s and has maintained a fast pace. In this talk I will review some of this history with the goal of setting the stage for future development. I have opted to split this development chronology into four stages by the nature of the electronics with the culmination of software-based systems. This talk will consider near-term futures that are gained from technologies such as software beamformation. An aid in defining this future will be information gained from the programs of various technical conferences. It is assumed that much of the work presented at such meetings has gone through broader review processes such as reviews of grant proposals and manuscript submissions. A hopefully interesting recommendation that arises from this analysis is towards research for improved understanding of sound/tissue interactions based on received echo data (e.g., channel data) and how this understanding may be used to improve image formation and increase our understanding of pathologies. An aspect that may arise from this is the potential of being able to define an upper limit on performance of ultrasound scanners in a broad patient population.

8:20

2aBAa2. Ultrasparse ultrasonic synthetic aperture beamforming via passive sensing. Francesco Lanza di Scalea (Structural Eng., Univ. of California San Diego, 9500 Gilman Dr., MC 0085, La Jolla, CA 92093-0085, flanza@ucsd.edu), Chengyang Huang, and Ali Hosseinzadeh (Structural Eng., Univ. of California San Diego, La Jolla, CA)

This presentation will describe the implementation of a passive ultrasonic sensing technique to achieve ultrasparse-transmission Synthetic Aperture Focusing (SAF) with Full Matrix Capture (FMC) capabilities. SAF ultrasonic arrays with sparse transmissions have been employed in both medical and industrial NDT imaging to increase imaging speed and simplify multiplexer hardware by reducing the number of high-voltage output channels, at the expense of a reduced beamforming matrix. A technique based on passive reconstruction of the ultrasonic Impulse Response Function (IRF) between two receivers will be demonstrated to create a "virtual" FMC capture while keeping a minimum number of physical transmitters, hence an ultrasparse-transmit array with full synthetic focusing capabilities. Several key steps of this passive beamforming approach will be discussed from both a theoretical and an experimental standpoint. The technique will be demonstrated for the imaging of drilled holes in an aluminum block using a linear, 64-element transducer array and only 1–4 physical transmitters. While the results presented have direct application to NDT imaging of solids, many of the aspects can be potentially applied to ultrafast imaging in the medical field, as well as to seismic interferometry for the health monitoring of civil structures.

8:40

2aBAa3. Towards computational super-resolution ultrasonic array imaging of material defects. Homin Song (Gachon Univ., Gachon, Korea (the Republic of)) and Yongchao Yang (Michigan Technolog. Univ., 1400 Townsend Dr., Houghton, MI 49931, ycyang@mtu.edu)

The resolution of sensing systems is fundamentally governed by the diffraction limit, which indicates that the minimum resolvable feature size is in the order of the wavelength of a propagating wave. Imaging smaller features (e.g., hidden material defects) requires short wavelengths (or high frequencies), which could be dangerous and causes low signal-to-noise ratio due to high attenuation of the propagating wave and coherent noise due to material backscattering. Computational super-resolution sensing, aiming to recover the sub-wavelength object features (e.g., material defects) from measurements taken with insufficient wavelength, is widely pursued across many applications; however, it is an ill-posed inverse problem that remains a significant challenge. Here we present a multi-scale deep learning approach to enable super-resolution ultrasonic beamforming that computationally exceeds the diffraction limit and visualizes sub-wavelength material defects. We investigate and discuss the applicability of this approach for computational super-resolution ultrasonic array imaging of hidden sub-wavelength defects of metallic structures.

9:00

2aBAa4. Model-based deep learning for ultrasound beamforming. Tristan S. Stevens (Elec. Eng., Eindhoven Univ. of Technol., 's Gravesandestraat 7s, Eindhoven, Noord-Brabant 5612JM, the Netherlands, t.s.w.stevens@tue.nl), Ben Luijten, and Ruud J. Sloun (Elec. Eng., Eindhoven Univ. of Technol., Eindhoven, the Netherlands)

Recently, the ultrasound signal processing pipeline has shifted from hardware-based solutions to the digital domain, enabling more intricate reconstruction techniques. We highlight several beamforming methods from a statistical inference perspective, connecting deep learning techniques to established signal processing models grounded in fundamental principles. The ultrasound acquisition process can be modeled as a linear combination of scatterers $\mathbf{x}$ (i.e. tissue reflectivity) which amounts to the received channel data $\mathbf{y}$, captured by a forward model $\mathbf{y} = \mathbf{H}\mathbf{x} + \mathbf{n}$, with the measurement matrix $\mathbf{H}$. Solving this system relies heavily on priors to yield a unique and anatomically feasible solution. A naive linear estimator for $\mathbf{x}$ is $\mathbf{H}^{T}\mathbf{y}$, also known as the delay-and-sum beamformer, which crudely disregards all off-axis scattering as zero-mean white noise. Based on the structured-noise assumption for off-axis scattering in the minimum variance beamformer, ABLE implements fast statistical inference of the optimal linear beamformer through deep learning. In addition, we can infuse prior knowledge on the distribution of $\mathbf{x}$ resulting in the neural maximum-a-posteriori beamformer. All aforementioned methods assume no dependency between individual pixels. Ultimately, the use of generative models allows us to extend the methods with more informative spatial priors.

9:20

2aBAa5. Improved beamforming by accounting for multiple domain shifts and sources of degradation. Ying-Chun Pan (Biomedical Eng., Vanderbilt Univ., Nashville, TN), Siegfried Schlunk, Christopher Khan (Vanderbilt Univ., Nashville, TN), and Brett Byram (Vanderbilt Univ., 2301 Vanderbilt Pl, Nashville, TN 37235, brett.c.byram@vanderbilt.edu)

A number of mechanisms are known to degrade ultrasound images. To address these issues, many beamforming strategies have been proposed including a slew of recent approaches centered around deep learning techniques. Most deep learning methods target a single type of image degradation. Here, we will simultaneously address multiple sources of image degradation within the same beamformer. In particular, we will address phase aberration, reverberation and off-axis scatting within a single network. Within this context of training networks to address multiple sources of image degradation, we will highlight the problem of domain shift in ultrasound beamforming deep networks. Domain shift is the mismatch in the statistics of the training data and the data encountered in the intended use case that reduces the performance of the trained deep network. We will show that ultrasound beamformers struggle with domain shift on both the input side and the output side of the networks, and we will present strategies for addressing both domain shifts including the shift on the output side where there is no ground truth data to learn from.

9:40

2aBAa6. Transcranial photoacoustic computed tomography image reconstruction with consideration of uncertainties in skull properties. Hsuan-Kai Huang (Elec. and Comput. Eng., Univ. of Illinois, Urbana-Champaign, 208 North Wright St., Urbana, IL 61801, hkhuang3@illinois.edu), Joseph Kuo (Elec. and Comput. Eng., Univ. of Illinois, Urbana-Champaign, Urbana, IL), Umberto Villa (Oden Inst. for Computational Eng. and Sci., The Univ. of Texas at Austin, Austin, TX), Lihong V. Wang (Elec. Eng., California Inst. of Technol., Pasadena, CA), and Mark Anastasio (Bioengineering, Univ. of Illinois Urbana-Champaign, Urbana, IL)

Transcranial photoacoustic computed tomography (PACT) is an emerging human neuroimaging modality but encounters challenges in achieving accurate image reconstruction due to aberrations induced by the skull. To compensate for aberrations, model-based image reconstruction methods that are based on the elastic wave equation have been proposed. However, to provide effective compensation, such methods often require the elastic and acoustic properties of the skull to be precisely known, which is challenging in practice. Moreover, these methods impose significant computational demands. In response to these challenges, a novel image reconstruction framework that was based on a learning method was proposed. The method involves the use of a U-Net based learning model to map an efficient but approximate reconstructed image to a de-aberrated, high-quality estimation of the initial pressure distribution within the cortical region of the brain. Computer-simulation studies that involved a realistic 3D stochastic head phantom were performed. The results of our studies demonstrate that the learning-based method exhibited similar performance to a state-of-the-art model-based technique when the assumed skull properties were precise, and notably surpassed the performance of the model-based method when uncertainties in the skull parameters were accounted for.

10:00–10:20 Break

10:20

2aBAa7. Differentiable beamforming for adaptive ultrasound imaging. Dongwoon Hyun (Radiology, Stanford Univ., 3155 Porter Dr., Palo Alto, CA 94304, dongwoon.hyun@stanford.edu)

Differentiable beamforming (DB) is a powerful new approach to image reconstruction that optimizes the beamforming algorithm to the given imaging target via autodifferentiation. In DB, the beamformer is expressed as a function of some unknown parameters, e.g., the sound speed throughout the medium, the shape of a flexible transducer, or the position and orientation of a swept transducer. Gradient descent is then used to find the parameters that optimize the focusing quality of the beamformer. With the appropriate choice of focusing criterion, the optimal beamforming parameters coincide with the ground truth. We demonstrate DB in applications of ultrasound autofocusing and sound speed imaging, flexible array shape estimation for wearable applications, and sensorless freehand swept synthetic aperture imaging.

10:40

2aBAa8. Abstract withdrawn.

11:00

2aBAa9. Parallel- and micro-beamforming challenges in real-time, high-frame-rate, ultrasound imaging. Piero Tortoli (Information Eng., Università di Firenze, via Santa Marta 3, Firenze 50136, Italy, piero.tortoli@unifi.it), Lorenzo Castrignano, Claudio Giangrossi, Valentino Meacci, Enrico Boni, and Alessandro Ramalli (Information Eng., Università di Firenze, Florence, Italy)

Delay-and-sum beamforming for classic (line-by-line) ultrasound imaging is usually performed in real-time by FPGAs and GPUs. However, when a high-frame-rate (HFR) has to be achieved, beamforming becomes challenging. Here, the echoes following the transmission of multi-line focused beams, plane waves, or diverging waves, must be simultaneously beamformed along multiple view lines. Such parallel beamforming is feasible online when the scanner is endowed with high flexibility and processing power. This talk will show how the FPGAs of the ULA-OP 256, a hardware-based open scanner, were efficiently utilized to enable parallel beamforming at high speed. The talk will also discuss how the data transfer between the scanner boards impacts the frame rate, which actually achieved 4400 frames per second through a new communication topology. The talk will also examine the image quality deterioration emerging when HFR imaging sequences are implemented in probes equipped with a microbeamformer (uB) that was designed for focused beam transmission. Simulations show how the transmitted beamwidth and uB size impact the image contrast and resolution, taking into account that the uB ASIC cannot handle multiple sets of delays and apodization weights after each transmit event. Technological improvements needed in the next generation uBs will be finally discussed.

11:20

2aBAa10. Real-time 3D ultrasound imaging with a clip-on device attached to common 1D array transducers. Zhijie Dong (Elec. and Comput. Eng., Univ. of Illinois Urbana-Champaign, Urbana, IL), Shuangliang Li (Elec. and Comput. Eng., Texas A&M Univ., College Station, TX), Chengwu Huang (Radiology, Mayo Clinic, Rochester, MN), Matthew R. Lowerison, Dongliang Yan, Yike Wang (Elec. and Comput. Eng., Univ. of Illinois Urbana-Champaign, Urbana, IL), Shigao Chen (Radiology, Mayo Clinic, Rochester, MN), Jun Zou (Elec. and Comput. Eng., Texas A&M Univ., College Station, TX), and Pengfei Song (Elec. and Comput. Eng., Univ. of Illinois Urbana-Champaign, 405 N. Mathews Ave., Beckman Inst. 4041, Urbana, IL 61801, songp@illinois.edu)

Performing 3D ultrasound imaging at a real-time volume rate (e.g., >20 Hz) is a challenging task. While 2D array transducers remain the most practical approach for real-time 3D imaging, the large number of transducer elements (e.g., several thousand) that are necessary to cover an effective 3D field-of-view impose a fundamental constraint on imaging speed. Although solutions such as multiplexing and specialized transducers, including sparse arrays and row-column-addressing arrays, have been developed to address this limitation, they inevitably compromise imaging quality (e.g., SNR, resolution) in favor of speed. Coupled with the high equipment cost of 2D arrays, these compromises hinder the widespread adoption of 3D ultrasound imaging technologies in clinical settings. In this presentation, we introduce an innovative transducer clip-on device comprising a water-immersible, fast-tilting electromechanical acoustic

reflector and a redirecting reflector to enable real-time 3D ultrasound imaging using common 1D array transducers. We will first introduce the principles underlying our novel technique, followed by validation studies incorporating simulation and experimental data. We will also demonstrate the feasibility of using the clip-on device to achieve a high 3D imaging volume rate that is suitable for advanced imaging modes such as shear wave elastography, blood flow imaging, and super-resolution ultrasound localization microscopy.

11:40

2aBAa11. Three-dimensional coherence beamforming using row-column arrays and matrix arrays. Joseph Thomas T. Hansen-Shearer (Bioengineering, Imperial College London, Royal School of Mines, Imperial College London, London SW7 2AZ, United Kingdom, jh12718@ic.ac.uk), Matthieu Toulemonde, Kai Riemer, Biao Huang (Bioengineering, Imperial College London, London, United Kingdom), Johanna Tonko (Univ. College London, London, United Kingdom), Jipeng Yan, Marcelo Lerendegui, Tan Qingyuan, Christopher Dunsby, Meng-Xing Tang, and Peter Weinberg (Bioengineering, Imperial College London, London, United Kingdom)

Much advancement in 3D ultrasound imaging has been achieved in recent years, allowing for its implementation in many research capacities. Translation of 3D imaging to a clinical setting has been hindered by small fields of view, expensive equipment and poor image quality. For ultrafast imaging, these challenges are accentuated by the need for a small number of transmission events. The row-column array is an emerging technology which can produce large field-of-view 3D ultrasound images, however, they can produce major artefacts when operated at high frame rates. Here we present some techniques which have been developed to improve the image quality and reduce the artifact level of row-column arrays by exploiting the incoherence of the transmission schemes. Presented will be the Row-Column specific Frame Multiply and Sum beamforming along with a technique for implementing Acoustic Sub-Aperture Processing which improves signal-to-noise ratio whilst also increasing the effective frame rate. Additional coherence-based algorithms, which have previously been developed for 2D imaging, are implemented and presented here for 3D imaging using a matrix array probe. All of the above algorithms have been optimised for super-resolution but should improve all types of 3D ultrasound imaging.

Session 2aBAb

Biomedical Acoustics, Education in Acoustics and Physical Acoustics: Return of the Writer

Julianna C. Simon, Cochair
Graduate Program in Acoustics, Pennsylvania State University, Penn State, 201E Applied Sciences Building, University Park, PA 16802

Karla Patricia Mercado-Shekhar, Cochair
Biological Engineering, Indian Institute of Technology Gandhinagar, Academic Block 6/207, Gandhinagar 382355, India

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

Chair's Introduction—8:00

Invited Papers

8:05

2aBAb1. On the elements of an excellent research paper. Alfred Yu (Univ. of Waterloo, 200 University Ave. West, Waterloo, ON N2L3G1, Canada, alfred.yu@uwaterloo.ca)

Journal publications play a key role in the dissemination of research findings in the biomedical acoustics community. However, authors are often unsure of how to write a paper that will be considered as excellent by their peers, because the notion of excellence is arguably subjective and ill-defined. To demystify such an issue, this presentation will discuss four major elements (novelty, significance, rigor, writing quality) that are often found in excellent journal papers in biomedical ultrasound. Since 2022, these four elements have been adopted as the peer review criteria by the flagship ultrasonics journal of the IEEE press. Multi-day writing workshops with interactive, hands-on tutorials have been designed to engage prospective authors on how to plan and prepare journal manuscripts with the four elements of excellence. Reviewer training workshops have also been offered to inform journal reviewers about these evaluation criteria and expectations. Through these initiatives, we aim to establish a quality-driven publishing emphasis within the biomedical acoustics community. Achieving so will have long-term benefits on the reputation and growth of this research field.

8:25

2aBAb2. Systematic literature review in five steps. Costas Arvanitis (School of Mech. Eng., Dept. of Biomedical Eng., Georgia Inst. of Technol., 311 Ferst Dr. Northwest, Atlanta, GA 30332, costas.arvanitis@gatech.edu)

A systematic review requires a considerable amount of time and resources. But if conducted properly it gives the opportunity to (i) summarize the current state of the art, (ii) identify the next questions that need to be answered to move one's field forward, (iii) Inform one's own research, (iv) support effective writing, and (v) receive recognition from your peers. This presentation will discuss the following five steps for conducting effective systematic literature review: (i) frame the question for the review; (ii) identify relevant work; (iii) assess the quality of studies; (iv) summarize the evidence; and (v) interpret the findings. The PRISMA (Preferred Reporting Items for Systematic Reviews and Meta-Analyses) flow diagram and checklist will be presented. A case study related to the estimation of cavitation thresholds during microbubble-enhanced focused ultrasound targeted drug delivery in the brain from a recent review paper by the author (Schoen *et al.*, Advanced Drug Delivery Reviews, 2022) will be discussed.

8:45

2aBAb3. Teaching the next generation of engineers and scientists to write: Taking responsibility and overcoming challenges. Michael Alley (College of Eng., Penn State, 201 Hammond Bldg., Penn State Univ., University Park, PA 16802, mpa13@psu.edu)

In educating engineering and science students as authors, four challenges exist today that did not exist thirty years ago. First, the decreasing attention over the past three decades on teaching grammar to students has left us with many students unable to analyze writing at the sentence level. Second, the increased focus by writing teachers on the writing process to the exclusion of the product has given many students the impression that properly finishing a paper is not important. No doubt, developing a process to write is important, but so is the final product. Third, many engineering and science instructors mistakenly assume that the responsibility of teaching students to write resides solely with English departments. To develop as authors in their disciplines, engineering and science students also need thoughtful feedback from experts in the field, particularly with regard to the precision of technical terms and the proper emphasis of details. Fourth, the emergence of artificial intelligence has led many students to assume that they need not learn anymore how to write. This paper presents three undertaught writing skills that engineering and science instructors should emphasize to the next generation of engineers and scientists for them to achieve excellence in their documents.

9:05

2aBAb4. But I did science so I wouldn't have to write essays. Eleanor P. Stride (Univ. of Oxford, Inst. of Biomedical Eng., Oxford OX3 7DQ, United Kingdom, eleanor.stride@eng.ox.ac.uk)

A dislike of writing, sometimes bordering on a phobia, seems to be a common experience for many science students and indeed practising scientists. This represents a serious challenge, as regular publication is a key requirement for most areas of research. In seeking to understand the origins of this aversion, a few interesting findings have arisen. First, there are some crucial differences between technical and creative writing, but these are rarely highlighted as part of either high school or university science courses. Second, there are surprisingly few degree courses in which the scientific method is introduced and discussed explicitly. In this talk, strategies for addressing these deficiencies and taking the pain out of report and paper writing will be discussed.

9:25

2aBAb5. Writing for a "different" audience. Kat Setzer (Acoust. Society of America, Acoust. Society of America, Salem, MA 11747, ksetzer@acousticalsociety.org), Micheal Dent (Psych., Univ. at Buffalo, SUNY, Buffalo, NY), and Arthur N. Popper (Biology, Univ. of Maryland, College Park, MD)

Written communication is a critical part of scholarship. However, scholarly writing is often incomprehensible to a broad audience, though we know that broader communication has great value in helping non-experts understand and support science. To bring acoustics to a larger audience, ASA publishes Acoustics Today (AT) to communicate the breadth of the field to members and to the public. To be successful, AT articles must be written so that they are scholarly, but also readable, understandable, and engaging to a college freshman, a newspaper reporter, a regulator, a funder, or one's bosses. Thus, a challenge for those of us working on AT is to help authors understand how to mold their ideas and material to our audience. Most authors find writing for AT a challenge since it is so different from their normal writing. As a result, we may go through many iterations of an article to help authors reach our audience. However, the reward is not only a well-written article that authors can share with colleagues in other disciplines, their bosses, and their parents, but many report that they have a new understanding of communicating with a wider audience.

9:45–10:00 Break

10:00

2aBAb6. Best practices for publishing in the Journal of the Acoustical Society of America. James Lynch (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., 86 Water St., Falmouth, MA 02543, jlynch@whoi.edu) and Elizabeth A. Bury (Acoust. Society of America, Melville, NY)

Scientists, engineers, and practitioners all have good reasons to publish in technical journals such as *The Journal of the Acoustical Society of America* (*JASA*), and so need to understand the procedures and best practices involved in doing so. While *JASA* has a technical focus in acoustics, it is also similar to other journals in many aspects of its writing and procedural requirements. We will address both the "business communication" and technical content aspects of journal writing. We will also stress some of the standard rules and protocols of journal writing, such as (1) polite and professional communications; (2) following specific journal requirements; (3) quoting and citing appropriately; (4) responding to comments from editors and reviewers; (5) complying with ethical standards and journal policies, etc. Useful references will be supplied to help with particular topics.

10:20

2aBAb7. Learnings from developing a Scientific Writing Certification program at IIT Gandhinagar. Karla Patricia Mercado-Shekhar (Dept. of Biological Sci. and Eng., Indian Inst. of Technol. Gandhinagar, Academic Block 6/207, Indian Inst. of Technol. Gandhinagar, Gujarat 382355, India, karlamshekhar@iitgn.ac.in)

Effective written communication is essential to an impactful career. The Certification in Scientific Writing initiative at the Indian Institute of Technology (IIT) Gandhinagar was developed to encourage researchers to work on scientific writing from an early stage in their careers and to promote skill development. This program serves as a milestone for gauging students' skills and knowledge in scientific writing. Since the certification program was established in July 2020, over 200 Ph.D. students and postdoctoral fellows from all departments at IIT Gandhinagar passed the program with approximately a 60% success rate. This program spans 2 months throughout the semester and offers training, evaluation, and feedback to students on their scientific writing competencies. In this talk, I will share the development of the program from inception to implementation and the feedback received from the IIT Gandhinagar community.

10:40

2aBAb8. How to write a peer-polished proposal in 15 weeks. Christy K. Holland (Internal Medicine, Div. of Cardiovascular Health and Disease, and Biomedical Eng., Univ. of Cincinnati, Cardiovascular Ctr. Rm. 3935, 231 Albert Sabin Way, Cincinnati, OH 45267-0586, Christy.Holland@uc.edu)

Creating a meticulously crafted proposal requires a strategic approach and systematic planning. An overview of a semester-long graduate course on how to write successful NIH grant applications will be provided. Particular emphasis is given to developing proposals for the Ruth L. Kirschstein Predoctoral Individual National Research Service Award (https://researchtraining.nih.gov/programs/fellowships/F31) or to disease-based foundations. The writing process involves drafting components in several key phases. The initial four weeks focus on understanding the proposal requirements, identifying the target audience, organizing a brilliant biosketch and

remarkable resources and environment pages, and establishing a clear hypothesis and specific aims. An extensive literature review is conducted in the subsequent two weeks to contextualize the proposal, identify a gap in knowledge, and stress the significance and innovation of the proposed work. Weeks 7 and 8 are devoted to the development of a robust research approach and methodology, including data collection, analysis techniques, expected outcomes, potential challenges, and alternative approaches. The review process, refinement and enhancement take center stage for the remaining weeks. Peer review and feedback mechanisms are incorporated to iteratively improve each proposal's coherence, logic, and persuasiveness. This systematic 15-week timeline emphasizes iterative refinement through peer input, ensuring a polished proposal ready for submission.

Contributed Papers

11:00

2aBAb9. Graduate-level research communication course: Reflections on the first offering. Julianna C. Simon (Penn State Univ., Penn State201E Appl. Sci. Bldg., University Park, PA 16802, jcsimon@psu.edu)

In Fall 2022, I offered a new course on research communication within the Graduate Program in Acoustics at Penn State. With 8 students from Acoustics and Biomedical Engineering, the course was split into three modules of grant writing, paper writing, and outreach. I found the students were reasonably adept at critically reading single papers. They also generally understood the mechanics of writing sentences, although combining sentences into paragraphs, connecting ideas, reducing fluff, and adding emphasis to key points were areas that significantly improved throughout the semester. In the middle of the course, two mock grant review panels were held, which highlighted a need to help students develop strategies to cope with critical feedback. Thus, for the final project, students had the option to either write a journal paper draft or revise their proposal; they also presented orally on their final project. Additionally, communicating with the public and developing outreach activities were included in the course, which culminated in an Acoustics Day at a science center. Overall, I found identifying gaps in the literature using multiple sources and developing a cogent argument on how their proposed work would fill the gap needed the most improvement. [Partially supported by the NSF CAREER:1943937].

11:15

2aBAb10. Empowering research: The role of writing accountability groups. Nour Al Rifai (Internal Medicine, Univ. of Cincinnati, Cincinnati, OH), Eric Smith (Dept. of Internal Medicine, University of Cincinnati College of Medicine, Cincinnati, OH 45221-0663, Smithep4554@gmail.com), Danielle N. Tap, Yolanda Y. Wess, and Kevin J. Haworth (Internal Medicine, Univ. of Cincinnati, Cincinnati, OH)

Productive writing is a challenge for researchers balancing multiple roles. Writing Accountability Groups (WAGs) can increase productivity by providing dedicated writing time. Patterned after a validated structure established by Kimberly Skarupski, Ph.D. (Johns Hopkins School of Medicine), the University of Cincinnati Department of Internal Medicine initiated four 10-week-long WAGs. Weekly virtual one-hour writing sessions were organized by staff with protected time and included an emphasis on principles of time management. Two WAGs were for junior faculty and two for residents/postdoctoral fellows, with 3–6 participants each. Attendance was approximately 80%, and 18 of 20 participants continued for the entire 10-week period. 14 of 20 participants completed a post-WAG survey. 85% reported improved time management skills. Additionally, most participants felt the WAG provided communal support and accountability. Rigorous writing project completion tracking is a future goal. Motivated by the experience, one postdoctoral fellow participant established a laboratory-based WAG. The participants included one faculty member, two postdocs, one Ph.D. student, two research staff, and ten undergraduate students. WAG participation was 50 ± 17 % over 44 weeks. In summary, similar to the experience of Skarupski, the UC WAGs facilitated consistent focus on writing, fostered a sense of community, and provided accountability.

11:30–12:00
Panel Discussion

Session 2aCA

Computational Acoustics: Computational Acoustics Methods Evaluation

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

Contributed Papers

8:00

2aCA1. A learned born series for highly scattering media. Antonio Stanziola (Medical Phys. and Biomedical Eng., Univ. College London, Dept. of Medical Phys. and Biomedical EngineeringUniversity College London, London WC1E 6BT, United Kingdom, a.stanziola@ucl.ac.uk), Simon Arridge (Univ. College London, London, United Kingdom), Ben Cox (Dept. of Medical Phys. and Biomedical Eng., Univ. College London, London, United Kingdom), and Bradley Treeby (Univ. College London, London, United Kingdom)

The Helmholtz equation with heterogeneous material properties is essential in various fields requiring single-frequency wave simulations, including optics, seismology, acoustics, and electromagnetics. Traditional methods for solving this equation, such as the Born Series, face limitations in converging for high-contrast scattering potentials. To address this challenge, we introduce a novel method called the Learned Born Series (LBS). The LBS is derived from the convergent Born Series but employs components that are learned through training. It demonstrates significantly improved accuracy compared to the conventional convergent Born Series, especially in scenarios with high-contrast scatterers, while maintaining similar computational complexity. The LBS can rapidly generate reasonably accurate predictions of the global pressure field with only a few iterations, and the errors decrease as more iterations are learned. We show its effectiveness through experiments on simulated datasets. The LBS offers promising prospects for accelerating simulations in scenarios with strong sound speed contrasts, potentially revolutionizing applications in transcranial treatment planning and full waveform inversion.

8:15

2aCA2. Application of differentiable programming to wave simulation. Antonio Stanziola (Medical Phys. and Biomedical Eng., Univ. College London, Dept. of Medical Phys. and Biomedical EngineeringUniversity College London, London WC1E 6BT, United Kingdom, a.stanziola@ucl.ac.uk), Simon Arridge (Univ. College London, London, United Kingdom), Ben Cox (Dept. of Medical Phys. and Biomedical Eng., Univ. College London, London, United Kingdom), and Bradley Treeby (Univ. College London, London, United Kingdom)

Wave simulations play a crucial role in a wide range of scientific and engineering applications, including seismic imaging, optical design, and acoustic modeling. Here we explore the advantages of differentiable programming in the context of acoustic wave simulation. Differentiable programming enables us to treat wave simulation as a differentiable function, allowing for the automatic computation of gradients with respect to any continuous input parameter. We demonstrate how this approach can be applied to various types of wave simulations, such as Pseudo-spectral time-domain solvers or iterative solvers. Implementing wave simulators via differentiable programming achieves several benefits. First, it enables efficient and accurate sensitivity analysis: This is particularly valuable for optimization and uncertainty quantification tasks. Second, it facilitates the incorporation of wave simulations into machine learning frameworks, enabling the integration of simulation-based models with data-driven approaches. Third, differentiable programming can accelerate the calibration and inversion of wave simulation models, making it easier to match simulated results to observed data. We present practical examples and discuss potential applications in fields such as geophysics and medical imaging. Our findings highlight the potential of this approach to advance the state-of-the-art in wave simulation techniques and their integration into larger computational pipelines.

8:30

2aCA3. Propagation of sonic booms around urban landscapes utilizing ray tracing. Brett A. Schankin (Phys., Farmingdale State College, 2350 NY-110, Farmingdale, NY 11735, schaba5@farmingdale.edu) and Kimberly A. Riegel (Phys., Farmingdale State College, Farmingdale, NY)

With the increasing interest from airlines and technology startups to return usage of commercial supersonic flight, there are still issues that need to be addressed. One of the more prominent barriers remains to be the high level of noise pollution produced by the shockwaves of sonic booms. While NASA and Lockheed Martin are currently developing the X-59 QueSST, an experimental aircraft with the aim of reducing the noise caused by these shockwaves, a closer look at the phenomena surrounding sonic booms can be examined by building computer models. The research being done looks to code a model that simulates the behavior of sound waves produced by sonic booms when they interact with urban geometries and structures. This is accomplished by utilizing a ray tracing method in a developed Python script. This script was initially developed using Fortran, but was ported to Python to take advantage of the language's ease of use and extensive libraries. The Python code must be vigorously validated to ensure it is correctly calculating the physical properties and direction of the sonic booms. Two types of atmospheres are included in the numerical simulations, a uniform atmosphere and the standard atmosphere. Comparisons of both types of atmospheres are shown compared to other numerical prediction models and measurement from the NASA SonicBOBS field measurements from 2009.

8:45–9:00 Break

9:00

2aCA4. Stochastic ray tracing implementation for sonic boom propagation modeling. Chloe A. Marini (Phys. and Astronomy, Univ. of Alabama in Huntsville, 6854 Governors West NW, Apt 2211, Huntsville, AL 35806, chloe.marini@yahoo.com) and Kimberly A. Riegel (Phys., Farmingdale State College, Farmingdale, NY)

Sonic boom propagation in large urban areas needs to be understood to determine the impact it will have on residents. In previous work, a combination of ray tracing and radiosity method was used to model the reflections of the sonic booms around large structures. Radiosity is a memory intensive method which requires extensive computational resources as the environment becomes more complex, whereas stochastic ray tracing does not require substantially more resources for more complicated environments. This study examines the feasibility of using stochastic ray tracing to simulate the diffuse reflections of sonic booms. Receiver graphs comparing stochastic ray tracing and radiosity methods for several environments will be shown, along with their computation times.

9:15

2aCA5. Fourie series decomposition of a modified propagation operator in wide-angle parabolic diffraction models: Accuracy analysis. Philippe Blanc-Benon (CNRS, 36 Ave. Guy de Collongue, Ecully 69134, France, philippe.blanc-benon@ec-lyon.fr), Elena O. Konnova, Maria M. Karzova, Vera Khokhlova, and Petr V. Yuldashev (Lomonosov Moscow State Univ., Moscow, Russian Federation, Moscow, Russian Federation)

Wide-angle parabolic equations are often used to model acoustic wave propagation in inhomogeneous media assuming weak backscattering effects. However, for three dimensional problems, efficient implementation of numerical methods to solve such equations using Padé approximations of the propagation operator is still a challenge. Fourier series approximation of the propagation operator can be used as an alternative approach since operator splitting methods are applicable to each Fourier harmonic of the series. Then calculations along two coordinates perpendicular to the primary wave propagation direction can be separated. High accuracy of the Fourier approximations can be achieved by constructing modified version of the propagator, which is smooth and continuous in a periodic window of the angular spectrum space under constraints of the maximum diffraction angle. Segments of the modified propagator complex valued function are connected using high-order Hermite interpolation method, which is capable to provide continuity up to the sixth order derivatives. Several free parameters available in the modified propagator can be used to diminish the approximation error appearing due to the Gibbs oscillations. The accuracy of the proposed Fourier approximations is tested for different maximum propagation angles and numbers of Fourier harmonics. [The work was supported by RSF 23-22-00220.]

9:30

2aCA6. Enabling support for Octave within the FOCUS software package. Jacob S. Honer (Elec. and Comput. Eng., Michigan State Univ., East Lansing, MI) and Robert J. McGough (Elec. and Comput. Eng., Michigan State Univ., 428 S. Shaw Ln., Rm. 2120, Michigan State University, East Lansing, MI 48824, mcgough@egr.msu.edu)

FOCUS, the "Fast Object-Oriented C++ Simulator" (https://www.egr.msu.edu/~fultras-web/), is a free software package that enables rapid calculations of continuous-wave and transient pressure fields generated by single transducers and phased arrays. FOCUS achieves small errors in relatively short computation times through linear memory-efficient calculations with the fast nearfield method. This capability extends to quick and accurate biomedical imaging simulations within FOCUS. Moreover, FOCUS supports some nonlinear pressure calculations, such as the continuous-wave and transient Khokhlov-Zabolotskaya-Kuznetsov (KZK) equations for circular and spherical transducers. Additionally, the Angular Spectrum Approach (ASA), a frequency-domain solution to linear computational methods, ideal for large volumetric field computations, is also included within the FOCUS package. Initial success with MATLAB motivates the creation of an Octave version that replicates the core functionalities of the original FOCUS package. GNU Octave is similar to MATLAB, but unlike MATLAB, Octave is a free software. Building and compiling FOCUS in Octave required a few minor alterations due to discrepancies in syntax and function calls. The use of Octave-compiled MEX files simplified the conversion process while introducing addition memory overhead compared to rewriting C++ code for Oct-file structuring. The performance and capabilities of FOCUS in Octave are demonstrated and discussed.

Session 2aEA

Engineering Acoustics, Physical Acoustics and Structural Acoustics and Vibration: Multidomain Modeling of Acoustical Systems

Stephen C. Thompson, Chair
Graduate Program in Acoustics, Penn State University, 201 Applied Science Building, University Park, PA 16802

Chair's Introduction—10:15

Invited Paper

10:20

2aEA1. Coupled models for designing and predicting performance of active sonar systems. Thomas E. Blanford (Univ. of New Hampshire, University of New Hampshire, Durham, NH 03824, thomas.blanford@unh.edu), Shawn Johnson, Jason Philtron, and Daniel C. Brown (The Penn State Univ., State College, PA)

Active sonar systems used to survey and image the seafloor typically transmit a pulse on a projector, receive echoes on an array of hydrophones, and condition and digitize the signals using on-board electronics. The quality of the data produced by these sensors is a function of the transducers, the electronics, and the environment. Simultaneous modeling of all these factors is necessary for accurate prediction of the system performance. Data products from these systems are often uncalibrated because absolute calibration is not required for their applications. However, modeling active sonar systems in their natural units within each domain provides clear benefits such as allowing system limitations to be accurately identified early during the design phase and easing validation of system performance in comparison with experimental data. This presentation will discuss the use of coupled multi-domain models in the design of two active sonar systems: the Sediment Volume Search Sonar and a multibeam echosounder architecture based on sigma-delta conversion. By coupling electro-mechanical transducer models and electrical hardware models with the Point-based Sonar Signal Model (PoSSM), it was possible to identify critical design parameters and sensitivities in the design of these sensors.

Contributed Papers

10:40

2aEA2. Multidomain modeling of acoustical transducers and arrays using electrical network theory. David A. Brown (ECE/CIE, Univ. of Massachusetts Dartmouth, 151 Martine St., UMass Dartmouth, Fall River, MA 02723, dbAcousticsdb@gmail.com)

Modeling of piezoceramic acoustic transducers involves the transformation of energy in the electrical, mechanical, acoustical radiation domains. By using generalized coordinates and Lagrangian mechanics, in which principle of least action is applied in each domain, the electrical, piezoelectric, mechanical and sound radiation problems can be solved separately and then combined in a multi-contour equivalent electro-mechanical circuit with each mechanical vibrational resonant modes representing separate degrees of freedom in the coupled electrical circuit. This energy approach involves the calculation of the potential and kinetic energies of each practical mode of vibration, which can be determined analytically, experimentally, or by finite-element-analysis. This is an alternative to the use of Newtonian mechanics which requires an accurate description of the real boundary conditions in the device and is common in many FEA modeling approaches. The availability of software (e.g., Matlab, Python, LTSpice, etc.) to model electrical circuits (networks in the case of multi-resonant devices) makes for a powerful and efficient modeling approach. The historical emergence of this approach stems from adapting the Rayleigh-Ritz method for mechanical and electrical domains with the advent of piezoelectric and magnetostrictive bodies. Several example of piezoceramic transducers are presented.

10:55

2aEA3. Abstract withdrawn.

Invited Paper

11:10

2aEA4. Modeling nonlinearities and thermal effects in moving coil speakers. Stephen C. Thompson (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, sct12@psu.edu)

Simple explanations of moving coil speaker behavior assume that the magnetic force factor, the mechanical spring stiffness and the coil electrical inductance are all constants of the design. The work of Klippel identifies all of these as varying with speaker cone displacement. The electrical resistance also varies with temperature, which varies because of electrical heating in the coil. Methods of measuring these nonlinearities are available in commercial equipment. Using this data to model speaker performance requires a model that considers behavior I the electrical, mechanical, thermal and acoustical domains. This paper describes an open source method to use the measured nonlinear parameters to calculate the time waveform of radiated pressure with large signal excitation for speakers.

Contributed Paper

11:30

2aEA5. Hydrostatic pressure effects in periodically structured polyurethane acoustic tiles. Luke R. Hacquebard (Defence Res. and Development Canada (DRDC), 2635 Provo Wallis St., Halifax, NS B3K 5X5, Canada, luke.hacquebard@ecn.forces.gc.ca) and Jeffrey P. Szabo (Defence Res. and Development Canada (DRDC), Halifax, NS, Canada)

Traditional underwater acoustic materials commonly make use of periodic air voids to improve their ability to absorb or deflect sound through wave-mode conversion and scattering mechanisms. These air voids have the potential to deform or completely collapse under hydrostatic pressure loading, which may significantly affect the material's acoustic performance. This study investigates the effects of hydrostatic pressure on the acoustic response of two different types of fabricated polyurethane samples. Developed numerical finite element simulations help identify the main factors associated with the acoustic pressure dependence of such tiles. The results of this work may aid in the development of pressure resistant materials that can better maintain acoustic performance at depth.

Session 2aMU

Musical Acoustics: Winds Instruments I

Gary Scavone, Cochair
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Jonas Braasch, Cochair
School of Architecture, Rensselaer Polytechnic Institute, School of Architecture, 110 8th Street, Troy, NY 12180

Chair's Introduction—8:40

Invited Papers

8:45

2aMU1. Acoustical behaviour of "baroque" trumpets with vent holes. Murray Campbell (School of Phys. and Astronomy, Univ. of Edinburgh, James Clerk Maxwell Bldg., Peter Guthrie Tait Rd., Edinburgh EH9 3FD, United Kingdom, d.m.campbell@ed.ac.uk), Arnold Myers (Royal Conservatoire of Scotland, Glasgow, United Kingdom), and Michael Newton (Reid School of Music, Univ. of Edinburgh, Edinburgh, United Kingdom)

The trumpets for which baroque composers including Bach and Handel wrote viruoso solo parts were natural (valveless) instruments approximately twice the length of a modern C trumpet. Playing chromatically in the high register of a natural trumpet requires great skill, since only the approximately harmonic resonances of the fixed-length air column are available. In the second half of the twentieth century several makers collaborated with performers to develop variants of the natural trumpet incorporating three or four finger-holes. Such instruments, widely used in modern ensembles playing baroque music, are best described as 'vented trumpets'. The way in which the finger-holes are used differs from the use of tone-holes on the cornett, the keyed trumpet and the keyed bugle. In general, vented trumpets are designed to approach the timbre of natural trumpets while facilitating control of intonation and accuracy in playing. In the present paper, the acoustical foundations for the function of the three- and four hole systems are explored. The study draws on input impedance measurements, computational modelling, and playing tests of typical vented and natural trumpets. Parameters quantifying the timbre of vented trumpets are compared with corresponding values for natural trumpets.

9:05

2aMU2. A comparison of modeled and measured impedance of brass instruments and mouthpieces. Miranda Jackson (Music Res., CIRMMT, McGill Univ., Schulich School of Music, Montreal, QC H3A 1E3, Canada, miranda.jackson@mail.mcgill.ca) and Gary Scavone (Music Res., CIRMMT, McGill Univ., Montreal, QC, Canada)

The impedance of a brass instrument has an important influence on its playability and sound timbre. The geometry of the mouthpiece has various features, such as the cup volume and shape, opening diameter, and length, that determine the characteristics of the overall impedance of the instrument-mouthpiece combination. Brass instruments, and especially mouthpieces, are designed for specific purposes, and horns or mouthpieces are chosen depending on the musical requirements. In order to investigate the relationship between the physical parameters and the impedances of instruments and mouthpieces, they have been modeled with transfer matrix and finite element model techniques, and the results are compared with impedance measurements of instruments, mouthpieces, and combinations of instruments and mouthpieces. Trumpets, flugelhorns, (French) horns, trombones, and the corresponding mouthpieces have been used. A detailed analysis of the estimation of the viscothermal losses has been performed, as the loss estimation in the narrow throat of the mouthpiece and in the flaring part of the brass instrument bell departs from the ordinary transfer matrix calculations. The effect of varying the physical parameters of mouthpieces and instruments is investigated by means of impedance considerations and sound synthesis, and the resulting influences on intonation, playability, and timbre are presented.

9:25

2aMU3. Pitch change of the stopped French horn. Robert Pyle (Stephens Brass Instruments, 11 Holworthy Pl., Cambridge, MA 02138, rpyle@icloud.com)

Composers sometimes ask for a form of muting on the French horn called "hand stopping," where the player closes the bell with the heel to the right hand as completely as possible. This alters the timbre of the instrument and its pitch as well. Players are taught to compensate for the pitch change by transposing down a semitone on the F horn. Stopping thus appears to have raised the pitch of the F horn by a semitone. However, if the heel of the hand is gradually closed from its normal "open" position to full stopping while the horn is played, the pitch falls smoothly. So does hand stopping raise the pitch, or lower it? This has been the topic of sometimes heated discussion among hornplayers for perhaps 250 years. An experiment shows that both viewpoints can be considered correct. A mouthpiece was fitted with an earphone and a microphone so that the acoustical round-trip time through the horn can be determined from the response to

a click emitted by the earphone. As the hand is closed, the round-trip time increases ("pitch falls") but as the hand approaches full stopping, an earlier reflection appears and eventually dominates ("pitch rises").

9:45

2aMU4. Power and good music: The Indigenous southern plains flute tradition. Paula Conlon (School of Music, Univ. of Oklahoma, 593 MacLaren St., Ottawa, ON K1R 5K8, Canada, pconlon@ou.edu)

In traditional Indigenous southern plains culture, a young man could not talk to a young woman alone when they were not yet married. Instead, he would play his flute at the edge of the encampment in the evenings, and each young man had his own love song. In southern plains flute origin stories, power is attributed to good music. If a flute song achieves its intended goal of convincing the young woman to marry the flute player, one can assume the song would be considered "good." But what criteria distinguish good from bad? Which elements typify a "good" flute song? What about the flute itself? Which features epitomize the quintessential flute? Another set of possible criteria in determining that quality is information about the flute player. In her chapter, "Culture and Aesthetics," ethnomusicologist Marica Herndon (1980) observes the community-centered perspective of Indigenous North America. Our last set of criteria involves an assessment of the moral character of the flute player regarding service to their tribal community. This talk discusses the "good music" of two master Indigenous southern plains flute players—Belo Cozad (1864-1950) (Kiowa) and Doc Tate Nevaquaya (1932-1996) (Comanche).

10:05–10:20 Break

10:20

2aMU5. Impedance phase-dependent correction factor for a tonehole model. Michael Prairie (Elec. and Comput. Eng., Norwich Univ., 158 Harmon Dr., Northfield, VT 05663, mprairie@norwich.edu)

The upper and lower end corrections of an open tonehole in a woodwind instrument are often accommodated by adding them to the physical thickness of the wall to create an effective thickness used in calculating the impedance of the tonehole. A rough estimate of the correction often used is 1.5 time the tonehole radius (1.5b), while other more rigorous treatments yield values in this vicinity. These values are usually independent of frequency or are constrained to a range of geometric tonehole and bore dimensions. The data from this study showed a strong and significant frequency dependency that produced a large range of correction values between about 0.5b and 1.5b. This variation correlated with the phase of the impedance of the highest open tonehole, and generally increased with increasing sounding frequency. This paper will discuss the conditions under which the measurements were made and how the correction factors were obtained, and will propose how the observed frequency dependence can be incorporated into the correction factor through the phases of the impedances of the open toneholes.

10:40

2aMU6. Multiphonics in the recorder: Some observations on the effects of selective basis tone removal induced by localized perturbations of window-region air flow. Katherine L. Saenger (Phys., Auburn Univ., Ossining, NY, klsaenger@yahoo.com) and Nicholas Giordano (Phys., Auburn Univ., Auburn, AL)

Multiphonic tones in the recorder are readily produced with certain fingerings and pressure/flow conditions. These tones typically comprise two strong basis tones that do not have a harmonic relationship, along with weaker tones whose frequencies are linear combinations of the basis tone frequencies. In this work, an artificial blower was used to generate multiphonic tones in three recorder body types (each attached to the same Yamaha soprano recorder head): (i) a forked fingering for G5 in a standard tapered body; (ii) a C5 in a cylindrical body; and (iii) a C5 in a capped cylindrical body. It was found that a thin post placed in the recorder's window region could selectively remove one of the basis tones (as well as the combination tones) with little effect on the sound power of the other basis tone. Navier-Stokes simulations were performed to help understand why a post in this position can have these effects. [Work supported by NSF under Grant No. PHY2306035.]

11:00

2aMU7. Measuring the behavior of the acoustic standing wave exiting a flue organ pipe: Is the decay sinusoidal or exponential? Lauren K. Schefter (Phys., Rollins College, 1000 holt Ave., Box #6539, Winter Park, FL 32789, lschefter@rollins.edu), Whitney L. Coyle, Ashley E. Cannaday (Phys., Rollins College, Winter Park, FL), and Eric Rokni (Phys., Rollins College, Mequon, WI)

Correctly predicting the playing frequencies of a musical instrument is dependent on the length of the resonator with the addition of an end correction. There are multiple theories describing this end correction, perhaps the simplest being that the end correction of a pipe is a physical extension of the sinusoidal pressure standing wave inside the pipe. However, recent optical imaging of the flow in a flue organ pipe found an unexpected exponential decay of pressure just outside of the pipe. This work looks to validate those findings acoustically. A flue organ pipe was played at the 1st, 5th, and 7th harmonics and the pressure just inside and immediately outside the end of the pipe played was measured using a zero-degree PU Match Microflown sound intensity probe. These measurements were fit to both exponential and sinusoidal curves and compared to the optical images. While an exponential trend is in fact apparent in some cases, the goodness-of-fit appears to be dependent on which harmonic is sounding. Future work includes exploration of a potential transitional region, assessing the impact of altered pipe geometry (both cross-sectional shape and size), and investigating potential sensor interference by using other measurement equipment.

Contributed Papers

11:20

2aMU8. Spectral behavior of recorder tones during transitions between notes. Nicholas Giordano (Phys., Auburn Univ., College of Sci. and Mathematics, Auburn, AL 36849, njg0003@auburn.edu) and Katherine L. Saenger (Phys., Auburn Univ., Ossining, NY)

We have studied note transitions in a soprano recorder as tone holes are opened or closed. Experimental results with an instrument excited by an artificial blowing machine are compared to Navier-Stokes-based simulations for an essentially identical instrument geometry. Several different cases have been studied. (1) Transitions involving the opening or closing of a single tone hole, with the notes separated by two semitones; e.g., transitions between C5 and D5. (2) Transitions involving the opening or closing of several tone holes simultaneously; e.g., transitions between C5 and G5. (3) Transitions involving a single tone hole but for notes separated by a major fifth; transitions between C5 and G5 using a forked fingering. Our work has focused on tone hole openings and closings at speeds typical of a human player but interesting behavior was also observed for longer tone hole switching times for cases (2) and (3), in which the sound field inside and outside the instrument was found to take much longer than the tone hole switching time to reach steady state. Work supported by NSF grant PHY2306035

11:35

2aMU9. Some observations on the sound powers of flute combination tones produced by singing while playing. Katherine L. Saenger (Independent Researcher, Ossining, NY, klsaenger@yahoo.com)

Singing while playing" is an extended flute technique that can produce weak, but audible, combination tones that are heard along with the note being sounded on the flute and the note that the player is singing or humming. The present work was motivated by an interest in obtaining a quantitative understanding of the factors determining the sound powers of these combination tones in hopes of making them louder and/or more audible. The tones under study were produced in a conventional flute powered by an artificial blower apparatus whose pressure/flow could be modulated at audio frequencies (by a speaker connected to the gasline) to mimic player singing. While quantitative results were elusive, one takeaway finding was clear: The combination tones are loudest when they have a frequency that aligns with one of the flute's passive resonances, a situation which can be facilitated with the use of forked fingerings that preserve the strength and position of the resonance used for the sounded flute note while shifting the positions of the unused resonances so that one of them better coincides with the frequency of the desired combination tone.

TUESDAY MORNING, 14 MAY 2024 ROOM 205, 8:00 A.M. TO 11:15 A.M.

Session 2aNSa

Noise, Computational Acoustics and Psychological and Physiological Acoustics: Advanced Air Mobility Noise: Noise from New Air Transportation in Urban and Underserved Communities I

Matthew Boucher, Chair
NASA Langley Research Center, 2 N. Dryden St., M/S 463, Hampton, VA 23681

Chair's Introduction—8:00

Invited Paper

8:05

2aNSa1. Optimization of propellers under consideration of aeroacoustic and aerodynamic goals and installation effects. Michael Schmähl (Chair of Aircraft Design, Tech. Univ. of Munich, Boltzmannstraße 15, Garching 85748, Germany, michael.schmaehl@tum.de) and Mirko Hornung (Chair of Aircraft Design, Tech. Univ. of Munich, Garching bei München, Germany)

Advanced Air Mobility (AAM) vehicles like air taxis and cargo unmanned aerial vehicles (UAVs) operate close to urban areas. Therefore, it is necessary to keep the noise footprint of such aerial vehicles at a minimum to gain societal acceptance for AAM vehicle operations. Cargo UAVs are typically highly integrated in terms of functions leading to a high extent of aerodynamic interactions between propellers and the airframe. As a result of these installation effects, unsteady loading noise can become the dominant part of the aerial vehicle noise emissions which leads to a situation, where classical propeller noise reduction measures like blade tip Mach number reduction do not necessarily lead to a reduction in the overall noise emissions. Overcoming these uncertainties in propeller design necessitates propeller noise optimization on an aircraft configuration level. In this work, an existing propeller optimization framework for isolated propellers was extended accordingly: Consideration of propeller inflow velocity perturbations due to installation in the blade element momentum theory-based propeller performance prediction and utilization of a blade element method code (BEM) to consider the scattering of propeller noise on the airframe accounts for configuration noise aspects. Optimization results of a pusher propeller UAV configuration conclude this work.

8:25

2aNSa2. Computational optimization of trailing-edge designs to reduce airfoil self-noise. Behzad Amirsalari (Dept. of Mech. and Aerosp. Eng., Carleton Univ., 1125 Colonel By Dr., Ottawa, ON K1S 5B6, Canada, behzadamirsalari@cmail.carleton.ca) and Joana Rocha (Dept. of Mech. and Aerosp. Eng., Carleton Univ., Ottawa, ON, Canada)

This paper presents an investigation and optimization of Trailing Edge (TE) design to reduce airfoil self-noise using Computational Fluid Dynamics (CFD). The case for study is a NACA0012 airfoil with a chord length (C) of 0.2286 m, a varying Angle of Attack (AoA) between 0° and 15°, and free stream velocity between 35 and 70 m/s. The flow domain consists of a c-type domain with a length and height of 18C and 9C, respectively. The parametric mesh maintains a structured mesh on the entire domain for different designs and TE shapes. Simulations employ a hybrid Stress-Blended Embedded Large-Eddy Simulations (SB-ELES) model to calculate the flow properties. Different turbulence models are tested to address their performance in determining pressure fluctuations. A correlation length also accounts for spanwise effects in the Ffowcs-Williams and Hawkings (FW-H) acoustic analogy approach to forecasting the far-field noise. Furthermore, multi-objective optimization is employed to determine the optimum airfoil TE configuration for different flow velocities and AoAs. The optimum designs generate the lowest Sound Pressure Level (SPL) without significantly sacrificing the aerodynamic performance of the airfoil within specified parameters.

8:40

2aNSa3. Multirotor broadband noise modulation. Ze Feng (Ted) Gan (Aerosp. Eng., The Penn State Univ., 229 Hammond Bldg., University Park, PA 16802, tedgan@psu.edu), Vitor Tumelero Valente, Kenneth S. Brentner, and Eric Greenwood (Aerosp. Eng., The Penn State Univ., State College, PA)

Rotor broadband noise spectra are typically analyzed over time scales on the order of one or more rotor periods. However, modulation of the broadband noise spectrum with the blade passage frequency (BPF) has been shown to be significant for noise levels and perception of wind turbines and helicopters. In contrast, time-varying broadband noise has not been extensively studied for aircraft with many rotors, such as unmanned aerial vehicles (UAVs) or advanced air mobility aircraft. In this work, significant broadband noise modulation was measured in flight and anechoic chamber tests of hexacopter UAVs at various observer angles. This modulation is aperiodic with the BPF such that the modulation amplitude varies substantially between blade passages, even when the BPFs are controlled to be nearly constant between all rotors at all times. Furthermore, the azimuthal phasing between rotors greatly affects the measured modulation, such that the modulation of multiple rotors may be less than or greater than for a single rotor, depending on the phase offsets. The effects of phase variations on acoustic interactions between rotors is studied by comparing the sum of the modulation of individual rotors to the modulation of those rotors operating simultaneously. This is done not only using measurements, but also noise predictions made using PSU-WOPWOP. These results contribute understanding to how the noise modulation of rotors sum together, including the resulting directivity and aperiodicity.

Invited Papers

8:55

2aNSa4. Use-case study for a smaller German community and beyond, with realistic traffic scenario and three alternative vertiport locations. Michael W. Bauer (Munich Aeroacoustics, Kirchheim, Kirchheim 85551, Germany, sub@muc-aero.com)

Future individual air traffic will be based on air vehicles, such as drones and air-taxis. While drones, e.g., for delivery of goods may be processed decentralized, passenger air-taxis will be operated at vertiports in city centers, e.g., connected to big rail hubs, close to civil airports, but also in smaller communities in the vicinity of larger cities. Here, air-taxi noise—mainly take-off and approach—can impact residential areas inside the community but also in its neighborhood. In cooperation with a midsized community near to Munich/Germany, three potential vertiport locations in the community's area are investigated under the aspect of noise and potentially related noise annoyance. To create a use-case with results, transferable to similar situations, realistic numbers for daily air-taxi movements between the use-case vertiport and five neighbor cities are included. These connections imply two different types of air-taxis: one tilt-wing eVTOL for the ranges up to 75 miles, and one multicopter aircraft serving on distances up to 30 miles. The UAM noise use-case study is performed for all three vertiport positions, optimized flight paths, and different flight profiles. The results regarding community noise, but also for en-route noise between the neighboring vertiports, are discussed in this paper.

9:15

2aNSa5. Risks and benefits of air taxis as perceived in Germany. Hinnerk Eißfeldt (FL-SEG, DLR German Aerosp. Ctr., Bachstraße 94, Braunschweig, Lower Saxony 22083, Germany, hinnerk.eissfeldt@dlr.de)

In a telephone survey on the acceptance of civilian drones in Germany conducted at the end of 2022 a certain part of questions was dedicated to air taxis explicitly. While the attitude towards civilian drones tended to be slightly more positive than in the preceding study, in the again nationwide representative study attitudes concerning air taxis were revealed to be relatively balanced, with a slight negative tendency. Inferential statistical analyses showed factors such as age, gender, active experience with drones, and interest in environmental protection to be significantly associated with the attitudes towards both, civilian drones and air taxis. According to prior findings on the relevance of noise concerns for drone acceptance in general, the current study asked for some noise related information, revealing subjective noise sensitivity and general noise annoyance being associated with the acceptance of civilian drones as well as air taxis. Evaluating information from answers to free format questions about air taxis, this contribution reveals reasons why in Germany—although the contributing factors are the same—the general attitude towards air taxis is less positive compared to civil drones, and what role noise considerations might play.

9:35–9:50 Break

Contributed Papers

9:50

2aNSa6. Aeroacoustic and aerodynamic analysis of a Conceptual Unmanned Aerial Vehicle (UAV) using Computational Fluid Dynamics (CFD). Mayur Nunkoo (Carleton Univ., Ottawa, ON, Canada) and Joana Rocha (Carleton Univ., 1125 Colonel By Dr., Ottawa, ON K1S 5B6, Canada, Joana.Rocha@carleton.ca)

The current investigation aims to conduct a comprehensive analysis of the aerodynamics and aeroacoustics of Manta, a conceptual Unmanned Aerial Vehicle (UAV) in the Bio-inspired Environmentally Friendly Aerial Vehicle (BEFAV) project at Carleton University. The primary objective involves exploring the effects of a Blended-Wing-Body design on critical parameters such as lift, drag, and Sound Pressure Level (SPL), through Computational Fluid Dynamics (CFD). This analysis considers the cruise phase of flight, featuring an airspeed of 67 m/s at an altitude of 1500 m. The incorporation of bio-inspired elements into Manta's design, aiming at increased operational efficiency and noise reduction, is also investigated.

10:05

2aNSa7. The reduction of noise and vibration of composite panels in aircraft through multi-dimensional particle swarm optimization. Noah Veenstra (Carleton Univ., 1125 Colonel By Dr., Ottawa, ON K1S 5B6, Canada, NoahVeenstra@cmail.carleton.ca) and Joana Rocha (Carleton Univ., Ottawa, ON, Canada)

The reduction of noise and vibrations on aircraft poses a unique challenge. This paper explores the use of multi-dimensional Particle Swarm Optimization (PSO) to reduce vibrations and transmitted noise in composite fuselage panels excited by turbulent boundary layer flow. Turbulent boundary layer Power Spectral Density (PSD) data is used to simulate the excitation of simply supported aircraft panels. A comparison is made between isotropic aluminum 2024-T3 panels and optimized composite panels of varying composition for the assessment of success in noise transmission reduction as well as structural function. The algorithm for PSO is implemented in Python and iterates through composites with varying thickness, ply makeup, ply orientation, and materials. Python allows for complex algorithms to be easily interfaced with Finite Element Analysis (FEA) programs. In the current study, ANSYS is used to determine the spectral response of the panels subject to turbulent flow using the superposition of the modes of vibration. During the optimization process, optimization parameters are updated in each iteration based on the success of reducing the spectral response of the panel without compromising its structural integrity or increasing its weight beyond a reasonable threshold.

10:20

2aNSa8. Acoustic characterization and optimization of a subsonic closed loop wind tunnel. Nicholas P. Cunnington Bourbonniere (Mech. and Aerosp. Eng., Carleton Univ., Ottawa, ON, Canada), Joana Rocha (Carleton Univ., 1125 Colonel By Dr., Ottawa, ON K1S 5B6, Canada, Joana.Rocha@carleton.ca), and Peter Waudby-Smith (Aiolos Eng. Corp., Toronto, ON, Canada)

A low-speed closed-loop wind tunnel with an open-jet test section having a maximum windspeed of $60\,m/s$ is being modified for future aeroacoustic testing. The wind tunnel is new to Carleton University as of 2021 and no acoustic characterization tests have been completed until recently. The objective of this research is to acoustically characterize the wind tunnel before and stepwise through its acoustic modifications. Multiple tests have been made using 1/4-inch microphones at different locations inside the plenum to record the existing out-of-flow background noise. The first planned wind tunnel modification is a height extension to the current plenum with

dimensions $0.74 \times 1.8 \times 1.3$ *m* ($H \times L \times W$) to increase the observer distance for the microphones. Once the plenum volume is increased, acoustic treatment will be applied to the interior of the plenum to create a hemi-anechoic measurement environment. An in-flow microphone mount was designed and manufactured with computer-controlled vertical traversing; it will be installed after the plenum modifications are complete.

Invited Papers

10:35

2aNSa9. Individual response trends to urban air mobility noise in a laboratory study. Aaron B. Vaughn (Structural Acoust. Branch, NASA Langley Res. Ctr., 2 N Dryden StMS 463, Hampton, VA 23681-2199, aaron.b.vaughn@nasa.gov) and Andrew Christian (Structural Acoust. Branch, NASA Langley Res. Ctr., Hampton, VA)

Advanced Air Mobility, of which Urban Air Mobility (UAM) is a subset, presents new opportunities for more dynamic aviation transportation systems. It is essential to understand the human response to the noise of these vehicles for sustainable operations. This research aims to investigate the relationship between the noise level and number of events on individual human annoyance to UAM vehicle noise. A "noise and number" laboratory psychoacoustic study was conducted in the Exterior Effects Room at NASA Langley Research Center in January 2023. A total of 38 participants listened to 4-minute audio clips of UAM vehicle flyover noise and provided their annoyance response on an 11-point numerical scale. A test hypothesis was that annoyance to multiple flyovers can be found by adding the annoyance responses to individual flyovers. Results for the overall test subject pool suggest that annoyance grows faster than hypothesized as the number of aircraft events increases. This trend of faster annoyance growth only seems to hold when the individual flyovers are heard as distinct events. Parsing the data down to individuals provides further insight into the mechanisms contributing to the increased annoyance with number of events.

10:55

2aNSa10. Studying holistic perception and response to unmanned aerial vehicle sound within acoustic environments: Bridging the gap between annoyance and soundscape. Marc C. Green (Acoust., Univ. of Salford, Newton Bldg., University of Salford, Salford M5 4BR, United Kingdom, m.c.green@salford.ac.uk), Michael J. Lotinga (Acoust., Univ. of Salford, Salford, United Kingdom), and Antonio J. Torija Martinez (Acoust. Res. Ctr., Univ. of Salford, Manchester, United Kingdom)

Given the projected increase in Unmanned Aerial Vehicle (UAV) deployment in the coming years, there is increased interest in studying their auditory impact. Most previous studies into perception of UAV sound have narrowly focused on noticeability and annoyance, typically presenting sound events in isolation. Whilst it is useful to understand these effects, isolated stimuli are not reflective of how these sources are likely to be experienced within ambient soundscapes. Furthermore, asking only about annoyance could bias participants, in effect prompting them to consider certain sound sources annoying. This can be seen as an instance of the conventional "environmental noise" approach, with the focus on disturbance and annoyance, in contrast with a "soundscape" approach, which adopts a more holistic focus. In soundscape, environmental sound is considered as a resource, with the potential for positive or negative effects, rather than only as a waste byproduct (noise). Here we present the results of a study involving UAV stimuli embedded within recorded acoustic environments. Participants evaluated sounds using a "Self-Assessment Manikin" to rate their affective response alongside a conventional noise annoyance scale. Associations between these two approaches are investigated, in an attempt to link "environmental noise" and "soundscape" literature with regards to UAV sound.

Session 2aNSb

Noise, Architectural Acoustics and ASA Committee on Standards: Soundscape—Focus on Applications I

Brigitte Schulte-Fortkamp, Cochair
HEAD Genuit Foundation, Ebert Straße 30 a, Herzogenrath, 52134, Germany

Bennett M. Brooks, Cochair
Brooks Acoustics Corporation, 49 N. Federal Highway, #121, Pompano Beach, FL 33062

Chair's Introduction—8:30

Invited Papers

8:35

2aNSb1. Applying soundscape: A matter of participation. Brigitte Schulte-Fortkamp (HEAD Genuit Foundation, Ebert Straße 30 a, Herzogenrath 52134, Germany, bschulte_f@web.de) and Bennett M. Brooks (Brooks Acoust. Corp., Pompano Beach, FL)

Defined as the acoustic environment understood by people in context, the soundscape encapsulates the myriad sounds that shape our daily lives (ISO12913 series). In the intricate tapestry of our sensory experiences, the soundscape emerges as a profound element, weaving together the diverse threads of auditory stimuli and social implications that surround us. Soundscape is not merely an assortment of sounds; rather, it is a complex interplay of natural and human-made elements that contribute to the acoustical and social identity of a place. Participation in the planning and modification of an acoustic environment requires an active and creative approach, it calls for a targeted engagement with the proposed redesigns or reorganizations. Participating in the soundscape approach also entails promoting acoustic ecology and the responsible management of acoustic environments. This involves efforts to minimize noise pollution, protect natural soundscapes, and create spaces where diverse sounds can coexist harmoniously. The paper will discuss different aspects of participation that offers numerous benefits, and as a participatory experience it opens new avenues for creativity, well-being, and community building in our ever-evolving acoustic living situations.

8:55

2aNSb2. Soundscape techniques applied to urban planning goals. Bennett M. Brooks (Brooks Acoust. Corp., 49 N. Federal Hwy., #121, Pompano Beach, FL 33062, bbrooks@brooksacoustics.com) and Brigitte Schulte-Fortkamp (HEAD Genuit Foundation, Herzogenrath, Germany)

The soundscape method is a powerful tool for understanding the perception of our acoustical environment. Urban planning is the technical and political process by which we manage and mold our cities. Smart growth is a movement within urban planning which emphasizes the livability of our surroundings. These disciplines all seek to shape the form of our communities to improve the quality of life for those who live, work, and play there. Yet, there has been little formal interaction between soundscaping and urban planning. It is imperative that soundscapers who wish to implement improved urban designs and interventions adopt the language and methods of urban planners in order to succeed and accomplish their mutual goals. Soundscape protocols must engage the urban planning process, including the comprehensive, or master, plan to achieve implementation on a city-wide scale. Several recent initiatives which illustrate the integration of soundscape techniques with urban planning methods are presented and discussed.

9:15

2aNSb3. Exploring soundscape methods and interventions in the Urban Soundscape of 3 cities. Gary W. Siebein (Siebein Assoc.,Inc., 625 NW 60TH St., Ste. C, Gainesville, FL 32607, gsiebein@siebeinacoustic.com), Keely M. Siebein, Marylin Roa, Gary Siebein Jr., Nicolas Ospina (Siebein Assoc.,Inc., Gainesville, FL), and Martin Gold (School of Architecture, Univ. of Florida, Gainesville, FL)

Vital mixed-use urban communities in the United States struggle to achieve balances among citizens moving back into cities, commercial activities, and entertainment venues that many people find desirable in live, learn, work, and play environments. Dynamic documentation, analysis, design, codes, and enforcement activities are required to achieve this balance in rapidly evolving, sustainable cities. Active engagement of the acoustical communities and the full range of stakeholders in each case were essential in understanding and addressing the issues. Multiple meetings with parties individually and in groups provided ways for all to understand the points-of-view of others as a building block to achieve consensus. Simple, but sophisticated, measurement and modeling of the soundscape were necessary elements of the methods used. Case studies in 3 large cities are presented of needs, issues, methods, analysis, and proposed solutions to a wide variety of acoustical issues encountered in the cities. Reflections on soundscape theory posed by the case studies and possible adjustments to theory are discussed.

9:35

2aNSb4. A soundwalk tutorial and exercise in Sydney, Australia. David s. Woolworth (Roland, Woolworth & Assoc., 356 County Rd. 102, Oxford, MS 38655-8604, dwoolworth@rwaconsultants.net)

The previous Acoustical Society of America's meeting #185 in Sydney, Australia joint with Australian Acoustical Society (AAS), Western Pacific Commission for Acoustics (WESPAC) and Pacific Rim Underwater Acoustical Conference (PRUAC) provided an opportunity to for an introductory tutorial on soundwalks followed by a walk through the city to gather measurement data and perceptive feedback from the participants. This presentation will provide the structure of the tutorial and preparation and results of the walk with analysis.

9:55–10:10 Break

10:10

2aNSb5. Psychoacoustic analyses of urban noise from long-term monitoring—Findings and outlook. Andre Fiebig (Eng. Acoust., TU Berlin, Einsteinufer 25, Berlin 10587, Germany, andre.fiebig@tu-berlin.de), Moritz Schuck (Eng. Acoust., TU Berlin, Berlin, Germany), Timo Haselhoff, and Susanne Moebus (Inst. for Urban Public Health, University Hospital Essen, Univ. Duisburg-Essen, Essen, Germany)

Urban noise is usually assessed using sound pressure level indicators to assess harmful noise effects. However, studies have shown that human responses to environmental noise are driven by psychoacoustic properties even beyond sound pressure level indicators. Although the value of psychoacoustic parameters for an improved characterization of environmental noise appears plausible and has often been shown in laboratory studies, a systematic examination of psychoacoustic parameters over long measurement intervals is lacking. In the research project SALVE+, long-term acoustic measurements at several locations in a German city were subject to psychoacoustic analyses and the location-dependent behavior of psychoacoustic parameters over time was examined. The measurements were analyzed to determine the acoustic quality of urban sound beyond sound pressure level and noise annoyance. Based on statistical analyses, the psychoacoustic properties of urban locations considering their land use are discussed regarding an improved characterization of acoustic environments. Based on this approach, the suitability of psychoacoustic parameters for mapping of spatial and temporal patterns is investigated. The presentation aims to highlight the value of noise monitoring in cities and analyze opportunities for improved urban noise management that additionally considers issues such as restoration quality and health promotion.

10:30

2aNSb6. Soundscape, attention and cognitive load. Adrian K. C. Lee (Speech & Hearing Sci., Univ. of Washington, 1417 NE 42nd St., Seattle, WA 98105, akclee@uw.edu)

The cocktail party problem was first coined by E Colin Cherry in 1953 and it describes the archetypal challenge of listening in a complex soundscape (e.g., multiple talkers conversing vying for your attention in a crowded restaurant). In the past two decades, the field of psychoacoustics has steadily marched towards understanding how we listen in these naturalistic environments. Many studies have focused on studying the psychological and physiological aspects of auditory scene analysis and object-based attention as well as how these processes differ in typical listeners from others with listening difficulties. In recent years, there has also been a burgeoning interest in understanding how cognitive load and listening effort affect our ease of listening in different situational contexts (e.g., talking while driving). In this talk, a brief survey of modern psychoacoustic experimental approaches will be presented in the hope to spur new collaborative research ideas with those who study soundscape.

10:50

2aNSb7. Soundscape augmentation for people with dementia requires accounting for disease-induced changes in auditory ccene analysis. Arezoo Talebzadeh (Ghent Univ., 126 Tech Ln. Ghent Sci. Park, Ghent 9052, Belgium, arezoo.talebzadeh@ugent.be), Dick Botteldooren, and Paul Devos (Information Technol., Ghent Univ., Ghent, Belgium)

Recently, there has been an increased interest in adapting the sonic environment to support people with cognitive difficulties, such as dementia. Research shows that incorporating "pleasant" sounds into the environment positively impacts behaviour and reduces psychological symptoms of dementia. Introducing sound into the acoustic environment creates an enhanced auditory experience, an augmented soundscape, resulting in an improved interpretation of the environment. People with dementia experience changes in their perception, which includes misperceptions, misidentifications, hallucinations, delusions, and time-shifting. Sound augmentation can support a better understanding of the environment and help in navigation time during the day. Dementia is a broad name for a degenerative disease; different types of dementia result in different syndromes and diverse auditory scene analyses. Some key auditory symptoms of different variants of the disease are auditory hallucinations, auditory disorientation, increased sound sensitivity, auditory agnosia (difficulty processing auditory input), agnosia for environmental sounds, amusia (tonal deafness) and Musicophilia. These different syndromes of auditory perception need more understanding when designing a soundscape augmentation for people with dementia. This talk aims to discuss different auditory symptoms of dementia based on a literature review and introduce ways to design an augmented soundscape to foster individual auditory needs.

11:10

2aNSb8. The Catalogue of Soundscape Interventions (CSI) project—A tool to bridge soundscape research and practice. Francesco Aletta (Univ. College London, Central House, 14 Upper Woburn, London N19DD, United Kingdom, f.aletta@ucl.ac.uk), Xiaochao Chen (Univ. College London, London, United Kingdom), Cleopatra Moshona (TU Berlin, Berlin, Germany), Jian Kang, Andrew Mitchell, Tin Oberman (Univ. College London, London, United Kingdom), Brigitte Schulte-Fortkamp (HEAD Genuit Foundation, Herzogenrath, Germany), and Andre Fiebig (TU Berlin, Berlin, Germany)

The Catalogue of Soundscape Interventions (CSI) project addressed the growing interest in urban soundscapes by establishing a comprehensive taxonomy for soundscape design. The project development is described, emphasizing its role in connecting soundscape research with practical applications. The proposed concepts for soundscape design interventions and other soundscape interventions are discussed, which seek to inform the creation of "soundscape design briefs." These briefs, intended for use by local authorities, aim to enhance communication among soundscape consultants and researchers, and other relevant stakeholders. This work also presents the intersection of the CSI project with the technical specifications outlined in the ISO/AWI TS 12913-4 Acoustics Soundscape Part 4: Design and Intervention, currently in development. By aligning with emerging standards, the CSI project seeks to contribute to a unified framework for soundscape practices, and to establish a model to describe phases and necessary action points in the life cycle of a soundscape design intervention.

Contributed Paper

11:30

2aNSb9. Taipei performing arts center surrounding soundscape study and its application. Roxana Ghadiri (Dept. of Architecture, National Taiwan Univ. of Sci. and Tech., 43 Keelung Rd. Sec. 4, RB809, Taipei 106, Taiwan, M11213803@mail.ntust.edu.tw), Shiang-I Juan, Stijn Zeger van Brug, Nikita Grace Manullang, Juliana Manuela Muet, Tuan Sanh Diep, Khaing Thinzar, Ni Made Putri Indriyani, Phoa Angela Grace Wibowo, Gabriela Niederberge, Clarissa Averina, Chau N. Truong, and Lucky S. Tsaih (Dept. of Architecture, National Taiwan Univ. of Sci. and Tech., Taipei, Taiwan)

Taipei Performing Arts Center designed by architect Rem Koolhaas and his firm OMA, features three theaters stacked vertically. The cube-shaped structure features a distinctive geometric facade, serving as the seating areas for the three main theaters. These giant facades, due to their sloped seating surfaces, could also function as significant sound-reflecting and focusing panels, potentially amplifying the nearby MRT and traffic noise. The project was strategically located near Shihlin Nightmarket for the area's cultural and economic growth. ISO-12913 soundscape study method is used to document essential sound sources, paths, and levels. The average sound pressure level of 67.4 decibels around the building provides an indication of the ambient noise in the immediate vicinity. While the acoustic comfort level in the plaza beneath the prominent facades may not be optimal, it is crucial to acknowledge their dual role as guiding individuals into the performance hall and serving as a buffer between external traffic and internal events. In addition to evaluating the impact of building mass and envelope design decisions on sound levels, this study uncovers a noteworthy finding related to urbanized traffic beeping. The observed and mapped patterns of local traffic sounds reveal unique cultural messages embedded in the auditory landscape.

Session 2aPA

Physical Acoustics, Noise, Engineering Acoustics, and Signal Processing in Acoustics: Wind Noise

Gregory W. Lyons, Cochair
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W. C. K. Alberts, Cochair
CCDC-Army Research Laboratory, 2800 Powder Mill Road, Adelphi, MD 20783

Chair's Introduction—8:00

Invited Papers

8:05

2aPA1. Predicting infrasonic wind noise levels using local topographic features. Garth Frazier (NCPA, Univ. of MS, NCPA, University of MS, P.O. Box 1848, Oxford, MS 38677, frazier@olemiss.edu), Roger M. Waxler (NCPA, Univ. of MS, University, MS), and Claus Hetzer (NCPA, Univ. of MS, Tempe, AZ)

This presentation describes a method for predicting RMS wind noise levels within a user-specified infrasonic frequency band (e.g., 1–10 Hz) using local topographic features. This capability is especially valuable when performing site selection for infrasound sensor array deployment. The method is based on building models using measured infrasound data from known sensor site locations and the local topographic features corresponding to the site. The presented results correspond to using features such relative terrain elevation and relative vegetation height obtained from publicly available sources. Mean wind speed and direction are also included as predictive parameters and these can be from data measured at the infrasound sensor site or from estimates obtained from a tool such as the Weather Research Forecast (WRF) model as was utilized for this investigation. A novel feature of the approach is that the terrain specification is not uniquely site specific but also depends on wind direction. Essentially, the topographic features are specified relative to wind direction and not in absolute coordinates. This enables a much richer set of samples from which to build the predictive models. While many options are potentially available for model building, this work focused on the use of multiple-layer artificial neural networks as a basis for regression. The results presented correspond to data from several infrasound sensor sites in the U.S Array Project.

8:25

2aPA2. The use of *in situ* infrasound calibration to correct wave parameter estimation. Samuel Kristoffersen (CEA, DAM, DIF, Bruyeres-le-Chatel, France), Alexis Le Pichon (CEA, DAM, DIF, Arpajon, France), Michaela Schwardt (PTB, Braunswchweig, Germany), Paul Vincent (CEA, DAM, DIF, Bruyeres-le-Chatel, France), Benoit Doury (CTBTO, Vienna, Austria), Franck Larsonnier (CEA, DAM, DIF, Bruyeres-le-Chatel, France), Christoph Pilger (BGR, Hannover, Germany), and Patrick Hupe (BGR, Stilleweg 2, Hannover 30655, Germany, patrick.hupe@bgr.de)

As part of the Comprehensive Nuclear-Test-Ban Treaty (CTBT), the International Monitoring System (IMS), which includes infrasound stations, has been established to monitor nuclear testing, as well as many other infrasound sources of interest to the scientific community. To minimize the effects of wind-noise, wind-noise reduction systems (WNRS) are installed as part of each sensor. In order to provide the best wave parameter (azimuth, trace velocity, amplitude) estimates, the response of the WNRS must be properly taken into account. Therefore, it is important to perform an *in-situ* calibration of the sensor using a co-located reference sensors and ambient signals to determine this response. This *in-situ* calibration can be used to monitor the status of the sensors, and provide feedback to the station operators. Experiments were performed at the IS26 site in Germany using a temporary WNRS to provide quantitative measurements of the effects of the WNRS and its calibration on the wave parameter estimation. These calibration results were used to provide corrections to the raw signal data for retrieval of the corrected wave parameters and were compared to the IS26 measurements, demonstrating that accurate measurements can be retrieved using the *in-situ* calibration.

8:45

2aPA3. Richard Raspet's contributions to our understanding of low-frequency wind noise. Jeremy Webster (Los Alamos National Labs, LANL, MS F665, Bikini Attoll Rd., Los Alamos, NM 87544, jwebster@lanl.gov)

Richard Raspet's research in wind noise spanned three decades and took the acoustics community from a rudimentary understanding of how low-frequency wind noise is generated in pressure sensors to a firm theoretical understanding derived from first principles in fluid dynamics. This talk will present a technical overview of his work starting with early quantitative studies, progressing through the development of the theoretical framework for wind noise studies, and finishing with a summary of research including wind noise at the ground surface, wind noise under tree canopies, wind fences, and wind induced seismic noise.

9:05

2aPA4. Towards a rapid-distortion theory model of turbulent flow for wind noise within an impermeable windscreen. Gordon M. Ochi (U.S. Army ERDC, 2902 Newmark Dr., Champaign, IL 61822, Gordon.M.Ochi@erdc.dren.mil) and Carl R. Hart (U.S. Army Engineer Res. and Development Ctr., Cold Regions Res. and Eng. Lab., Hanover, NH)

Current models for the wind noise reduction (WNR) of microphone windscreens in atmospheric turbulence generally work under the assumption of a homogeneous surface pressure averaging theorem. Under this theorem, the surface pressures should average in such a way that the noise spectrum at low wavenumbers relative to the windscreen dimension should approach that of an unscreened microphone. Contrasting this, experiments have actually observed a significant WNR at these wavenumbers. In this work, we examine a Rapid Distortion Theory model for linking turbulent distortion to the unsteady pressure received inside the windscreen. The theoretical background and underlying assumptions of Rapid Distortion Theory are examined in detail [R. Zamponi, *et al*., J. Fluid Mech. 915**:**A27 (2021)], then results for the case of an impermeable cylinder are presented. Future work on the project involving both theoretical development and experimental verification are then discussed.

9:25

2aPA5. Pseudospectrum-based methods for estimating the wind speed and direction using coherence decay-dependent steering vectors. Daniele Mirabilii (WSAudiology, Henri-Dunant-Straße 100, Erlangen, Bavaria 91058, Germany, daniele.mirabilii@wsa.com) and Emanuël A. Habets (Int.. Audio Labs., Erlangen, Bavaria, Germany)

In a recent work, we employed acoustic array processing to measure the wind speed and direction based on wind-induced noise recorded with multiple microphones. In particular, we proposed beamforming and signal subspace-based methods for estimating the wind speed and direction via pseudo-spectrum maximization. Based on experimental observations using a compact microphone array, we make more stringent assumptions on the wind-induced noise advection to delineate its weakly coherent propagation. To account for the wind turbulence dissipation across space and frequency, we introduce a speed- and direction-dependent anisotropic decay factor that models the relative propagation of the wind-induced noise source between microphones. The decay term is incorporated in the relative transfer functions of the noise source and, consequently, in the steering vectors used for the computation of the pseudo-spectra. Finally, we re-evaluate the proposed methods in terms of speed and direction accuracy under these new assumptions.

9:45–10:05 Break

Contributed Papers

10:05

2aPA6. Abstract withdrawn.

10:20

2aPA7. Dealing with wind noise on resource constrained acoustic sensors. W. C. K. Alberts (DEVCOM Army Res. Lab., 2800 Powder Mill Rd., Adelphi, MD 20783, william.c.alberts4.civ@army.mil) and Gregory W. Lyons (DEVCOM Army Res. Lab., Adelphi, MD)

Single and multiple microphones in small or hand-held devices, i.e., mobile phones, are often single port and without windscreens. In a typical outdoor situation, this configuration can be subject to strong impulsive wind noise events and wind-induced tones. Smart mobile phones can sometimes utilize their significant computational power to employ sophisticated signal processing to reduce or remove wind-associated noise. However, in systems where battery conservation is paramount and computational processing resources are severely limited, alternative strategies must be employed. This presentation will discuss exemplar wind noise data from a single channel, resource-constrained system, methods for wind noise reduction, and results.

10:35

2aPA8. Wind noise due to flow induced around a windscreen in large-scale turbulence. Gregory W. Lyons (DEVCOM Army Res. Lab., 2800 Powder Mill Rd., Adelphi, MD 20783, gregory.w.lyons3.civ@army.mil)

Turbulent wind noise on microphones is a ubiquitous problem for sound measurement and recording outdoors. Windscreens, such as open-cell foam spheres, are effective at reducing wind noise. The unsteady pressure sources induced on the windscreen surface must contribute less noise on average than the stagnation pressure on a bare microphone, or windscreens would not be effective. To quantify this noise reduction mechanism, a model is presented for the unsteady pressure within an impermeable windscreen of arbitrary shape immersed in an initially isotropic turbulent flow. Only turbulent scales larger than the windscreen are considered, which contribute the most intense low-frequency band of the noise spectrum in most practical situations. The inhomogeneous velocity field around the windscreen is modeled in terms of an induced irrotational blocking field that meets the kinematic boundary condition. The interior pressure is expressed as a convolution of the surface pressure field with a filter function in wavevector space. The wind noise wavevector spectrum is then shown to be only a function of the inflow isotropic velocity spectrum and the windscreen shape. Solutions are obtained for the pressure frequency spectrum within a spherical windscreen for a von Kármán turbulence model and compared with prior results in literature.

10:50

2aPA9. Very short time Fourier transform for turbulent analysis. Roger Oba (Acoust. Div., US Naval Reseach Lab., 4555 Overlook Ave. S.W., Washington, DC 20375, roger.oba@nrl.navy.mil) and Ravi Ramamurti (Naval Res. Lab., Washington, DC)

Nominally stationary acoustic sensors, e.g., buoy suspended or bottom anchored, are subject to various environmental flows. Direct numerical simulation of hydrodynamic flow around the acoustic sensor housing show turbulence along the housing surface. Time-space analysis of pressure variation along the surface of the housing shows significant acoustic-like surface interaction at low frequencies, even at very low mean flow velocity. In this case turbulence manifests as intermittently generated, irregular, coherent vortex structures that separate from the surface of the housing and advect downstream. These structures drive rapid surface pressure transients that propagate within the housing to the acoustic sensors as noise. Fractional Fourier Analysis provides joint time-frequency methods to extract transient features of turbulence. Use of Hermite-Gauss functions and phase relations are of particular interest. [This research is supported by 6.2 NRL base program sponsored by the Office of Naval Research. Distribution Statement A: Approved for public release. Distribution unlimited.]

11:05

2aPA10. A reduction in near-field hydroacoustic flow noise using shark skin-inspired surfaces. Jonathan Stocking (US Naval Res. Lab., 4555 Overlook Ave., SW, Bldg. 2, Rm. 129L, Washington, DC 20375, jonathan.stocking@nrl.navy.mil), Kaushik Sampath (KS Res. Inc., Greenbelt, MD), Nicole Xu (Univ. of Colorado, Boulder, Boulder, CO), Jason Geder (US Naval Res. Lab., Washington, DC), and Silvia Matt (US Naval Res. Lab., Stennis, MS)

Recent studies have demonstrated improved hydrodynamic performance for hydrofoils and uncrewed underwater vehicles (UUVs) using passive surface geometries such as riblets and shark skin-inspired denticles. However, little is known about the impact of added surface roughness on the production of flow-induced noise, a critical design consideration for UUVs in sensitive environments. We present here experimental results of near-field hydroacoustic noise measurements for boundary layer flow over smooth and bioinspired denticle-covered surfaces. A 3 in. wide and 10 in. long segment of staggered denticles was fabricated in-house using photopolymer 3D printing, and flow and acoustic measurements were made in a custom-built flow channel using particle image velocimetry (PIV) and miniature hydrophones. Comparing acoustic power spectra over flat and denticle-covered plates, we find both a scale- and frequency-dependent response in radiated noise. At low flow speeds ($Re \sim 20{,}000$), the denticle surface shows a 20 dB enhancement in radiated noise at frequencies below 1000 Hz, but no change at higher frequencies. While at high flow speeds ($Re \sim 60\ 000$), the denticle surface reduces radiated noise by $\sim$10 dB for frequencies between 4000 and 8000 Hz. Analysis of PIV data will elucidate the flow physics responsible for these effects. [Work sponsored by Office of Naval Research.]

TUESDAY MORNING, 14 MAY 2024 — ROOM 208, 9:30 A.M. TO 11:20 A.M.

Session 2aPP

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

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

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

Invited Papers

9:30

2aPP1. Gender-expansive listeners invite us to reconsider voice categorization as a simple matter of pitch and timbre. Jay Marchand Knight (Individialized (Science), Student, 7547 rue Centrale, Lasalle, QC H8P 1K4, Canada, juanita.marchand@gmail.com)

Past literature has focused on pitch and timbre cues as the main determinants of the voice of an actor (a speaker or a singer). But in real life, judges rely on non-auditory cues such as the actor's facial features, height, body size and shape. Resisting these visual biases could provide more accurate assessment of a voice but requires a stronger dissociation between auditory attributes and physical appearance, an ability hypothetically acquired by gender-expansive participants. In three online studies, we examine *Faching* (allocating voices into traditional categories). In study 1, 166 participants (85 cis, 81 gender-expansive) rated 144 audio (A) samples from 18 different actors (3 from each major *Fach* category, excluding countertenor) along a slider labeled low/dark to high/bright. Participants then guessed the voice types of the same actors in silent videos (V), before rating them in AV combinations. The cis group exhibited 30% more visual bias than the gender-expansive group. To further understand how gender-expansive participants obtain this benefit, we manipulated the fundamental frequency (in study 2) or the vocal tract length (in study 3) by $\pm$ 3 semitones from the original stimuli. Preliminary findings suggest that the two groups differ in the face of timbre but not pitch manipulations.

9:55

2aPP2. Comparing LTASS across languages using natural and AI speech samples. Abhijit Roy (Commun. Sci. and Disord., Northwestern Univ., 124 Callan Ave., Apt. 2B, Evanston, IL 60202, abhijitroy2025@u.northwestern.edu), Ann Bradlow (Linguist, Northwestern Univ., Evanston, IL), and Pamela E. Souza (Commun. Sci. and Disord., Northwestern Univ., Evanston, IL)

Hearing aid prescription protocols rely on estimates of speech spectra. One such estimate, the International Long-Term Average Speech Spectrum (ILTASS), is a common reference for the distribution of spectral energy within speech signals. However, recent research has suggested differences in speech spectra between some languages and the ILTASS standard. These disparities raise questions regarding the applicability of a language-general approach to hearing aid gain application. This study presents a comparative analysis of Long-Term Average Speech Spectra (LTASS) across various languages. Utilizing speech samples from both human and AI sources, LTASS was measured for speakers of different languages. Subsequently, these LTASS profiles were contrasted with the established

ILTASS reference. Results reveal that while LTASS was very similar across languages, deviations were observed between LTASS derived from the human speech and AI speech data compared to ILTASS. Specifically, measured LTASS demonstrated a reduced 500 Hz peak in contrast to ILTASS. This research underscores the importance of accounting for spectral variations introduced during the recording process in the field of speech and hearing research. It also emphasizes the importance of standardized recording protocols to enhance the precision of hearing technologies.

Contributed Papers

10:20

2aPP3. The relationship in phonotactics between cross-linguistic sonority principles and within-language probabilistic distribution of segment sequences. Peiman Pishyar-Dehkordi (Linguist Dept., Univ. of Canterbury, University of Canterbury, Private Bag 4800, Christchurch 8140, New Zealand, peiman.pishyardehkordi@pg.canterbury.ac.nz)

The worlds' languages contain a variety of cross-linguistic phonotactic patterns, in which certain sound sequences are more likely to occur in languages than others. Are these patterns also reflected as probabilistic distributions within individual languages? We investigate this question in the context of sonority constraints on syllable formation. Phonotactic patterns in syllable structure have been argued to be governed by sonority hierarchies which are loosely based on the acoustic intensity of speech sounds. Cross-linguistic sonority principles argue that the simplest syllable is one with the maximal rise in sonority at the beginning, and the minimal drop in sonority at the end. We also know that in addition to categorical constraints on their phonology, languages contain probabilistic phonotactic patterns within the sequences that they allow, meaning that in each language some sequences tend to be over-represented, and some sequences under-represented. If cross-linguistic patterns are also reflected as within-language probabilistic constraints, we hypothesize that in a language that allows both simple and complex syllables, simpler syllables will be more frequent than more complex ones. To test this, syllables within 8 languages were explored. Our findings are generally compatible with the above hypothesis for all the 8 languages, suggesting that there is a general alignment between sonority principles as cross-linguistic universals and within-language probabilistic distribution of segment sequences within syllables.

10:35

2aPP4. Lack of racial diversity in language science journals. Olivia Tobin (Linguist, Univ. of Iowa, Iowa City, IA), Paras B. Bassuk, Samantha Chiu (Psychol. Brain Sci., Univ. of Iowa, Iowa City, IA), and Ethan Kutlu (Linguist, Univ. of Iowa, 459 Phillips Hall, Iowa City, IA 52242, ethankutlu@gmail.com)

Numerous studies and reports have pointed out the lack of diversity in scientific spaces (Liu *et al*., 2023; Singh *et al*., 2023) which has a direct impact on shaping our research practices, questions, and scientific pursuits in general (Kutlu and Hayes-Harb, 2023). While the lack of diversity in science has been documented widely, there is no report on the diversity of editorial board members of language science journals, who play an integral role in diversifying science. This ongoing study aims to document to what extent language science journals engage with diversity and how diversity is represented in these spaces. To achieve this goal, we looked at 100 language journals and their editorial board members list. We hand-coded the editorial board members' perceived gender and race. We used multiple sources to check our coding (e.g., Wikipedia, websites, videos). Our preliminary findings suggest that while there is gender balance in editorial spaces, there was a significant lack of racial representation in editorial space (i.e., more White scholars in editorial spaces). Our next step is to collect archival data on editorial board members, document systematic inequalities, and provide suggestions for how racial diversity can be increased in editorial spaces.

10:50–11:20
Panel Discussion

Session 2aSA

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

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

Alexey Titovich, Cochair
Naval Surface Warfare Center, Carderock Division,

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

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

Dylan Kovacevich, Cochair
Mechanical Engineering, University of Michigan, 2350 Hayward St., Ann Arbor, MI 48109

Abigail D. Willson, Cochair
Acoustics, Penn State University, PO Box 30, Mail Stop 3220B, State College, PA 16804

Chair's Introduction—9:00

Invited Paper

9:05

2aSA1. Energy absorption from an oscillating fluid using an engineered acoustic diode. Rico Schmidt (Mech. and Aerosp. Eng., Univ. at Buffalo (SUNY), Buffalo, NY), Indradip Roy (Mech. Eng., Purdue Univ., West Lafayette, IN), Hosam Yousef (Mech. and Aerosp. Eng., Univ. at Buffalo (SUNY), Buffalo, NY), Alex Aboueria, Carlo Scalo (Mech. Eng., Purdue Univ., West Lafayette, IN), and Mostafa Nouh (Mech. and Aerosp. Eng., Univ. at Buffalo (SUNY), 240 Bell Hall, Attn: Mostafa Nouh, Buffalo, NY 14260, mnouh@buffalo.edu)

Over the past decade, engineered subsurface structures inspired by acoustic metamaterial configurations have shown promising results in boundary layer control through different energy stabilization mechanisms. Despite the potential shown via numerical simulations, the underlying interfacial dynamics between a fluid column and a compliant metamaterial remain largely unexplored. Our preliminary studies have shown that conventional phononic band gaps are insufficient for effective control since the bulk of the absorbed energy remains confined to the fluid-structural interface, a direct consequence of localized modes associated with band gap truncation resonances. In this talk, we will discuss the mathematical intricacies of impedance boundary conditions (IBCs) at the interface of a one-dimensional fluid and an engineered structure, shedding light on both linear and non-linear subsurface behaviors. Through which, we will present a new pathway to effectively transmit a pressure wave from an oscillating fluid away from the interface, and into a structural medium.

Contributed Papers

9:25

2aSA2. Variational study of a model for near-perfect transmission through lossy media with acoustic sources. Nathan P. Geib (Appl. Res. Labs., Univ. of Texas at Austin, 1587 Beal Ave Apt 13, Ann Arbor, MI 48105, geib@umich.edu), Samuel P. Wallen, Michael R. Haberman, and Christina Naify (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Complementary acoustic metamaterials have been proposed as a means of compensating for the high impedance mismatches of aberrating layers that disrupt the acoustic field and hence distort acoustic images. Recently, a complementary acoustic metamaterial featuring active components was shown in principle to compensate for both the impedance mismatch and energy attenuation of lossy materials, but a physical realization of the concept has not yet been implemented. Here, we present results from a one-dimensional acoustic model showing how a plane wave incident on a lossy material can be augmented by point monopole and dipole sources to allow for near-perfect transmission, thus rendering the lossy medium acoustically transparent. We present general expressions for source magnitudes that are dimensionless with respect to frequency, material thickness, and the background medium. We explore the sensitivity of the performance to variations in each of the model parameters, considering both theoretical and practical limitations to the proposed method. We show that these findings are consistent with three-dimensional finite element simulations.

9:40

2aSA3. Study of defect mode characteristics in acoustic metamaterials.
Vinod Ramakrishnan (Mech. Sci. and Eng., Univ. of Illinois at Urbana-Champaign, 1324 MEL,105 S Mathews Ave., Urbana, IL 61801, vinodr@illinois.edu) and Kathryn Matlack (Mech. Sci. and Eng., Univ. of Illinois at Urbana-Champaign, Urbana, IL)

The development of novel material architectures incorporating unconventional geometric designs (e.g., microstructural instabilities, multi-length scale geometries), or adaptive constituent elements (e.g., piezo electrics) has created a powerful tool to design metamaterials for efficient dispersion manipulation and precise control of acoustic waves. Various aspects of the acoustic dispersion (e.g., wave speed, damping ratio, band gap position and widths) have been investigated with a primary focus on wave characteristics in the acoustic pass bands, promoting their utility in applications, e.g., wave guiding, energy harvesting. However, recent studies on topological waves, truncation and bandgap resonances have highlighted potential benefits of leveraging acoustic waves in the bandgap, e.g., signal propagation, passive flow control. Motivated by this, we explore the dispersion behavior of finite mass-spring-damper acoustic metamaterial models with material defects. The eigen analysis reveals a (defect) resonance in the acoustic band gap, producing a highly localized defect mode. The dependence of this resonance frequency, the degree of localization and phase response of the eigenmode, on the defect parameters are explored. Subsequently, the potential utility of these defect modes in flow control applications is also explored and presented.

9:55

2aSA4. Enhanced viscous dissipation of sound in phononic supercrystal.
Arkadii Krokhin (Dept. of Phys., Univ. of North Texas, 1155 Union Circle # 311427, Denton, TX 76203, arkady@unt.edu), Martin Ibarias, and José Sánchez-Dehesa (Wave Phenomena Group, Universitat Politecnica de Valencia, Valencia, Valencia, Spain)

Sound wave propagating in a homogeneous viscous medium decays due viscous losses with decay coefficient $\gamma_{;0}{\sim}\eta\omega^2$, where η is shear viscosity. Presence of a hard wall strongly increases viscous losses which now occurs mainly within a narrow boundary layer δ. Viscous losses in a 2D phononic crystal are analytically calculated in the low-frequency limit for arbitrary Bravais lattice. The decay coefficient is expressed through series over reciprocal lattice vectors. If the phononic crystal possesses less than 3-fold rotational symmetry it behaves as anisotropic viscous medium. Otherwise, the decay coefficient is isotropic. Depending on the crystal structure and the filling fraction of solid cylinders the decay coefficient $\gamma_{;\mathrm{ph}}$ may exceed $\gamma_{;0}$ by two-four orders of magnitude. The decay coefficient may be further enhanced in a supercrystal – a structure with doubly periodicity. The supercrystal is a 2D structure of two sets of aluminum cylinders in air. A periodic set of larger cylinders is imbedded in another set of smaller cylinders arranged in a lattice with smaller period. We present analytical, numerical, and experimental results for decay of sound in a hexagonal supercrystal of aluminum cylinders in air. [This work is supported by the NSF under EFRI Grant No. 1741677.]

10:10–10:25 Break

10:25

2aSA5. Localization-antilocalization of soundwaves in a disordered phononic crystal with mirror symmetry.
Michael McKinstry (Phys., Univ. of North Texas, 1155 Union Circle, Denton, TX 76203, michaelmckinstry@my.unt.edu), Dmitrii Shymkiv, and Arkadii Krokhin (Phys., Univ. of North Texas, Denton, TX)

Anderson localization in a disordered potential leads to exponential decay of an incoming wave. However, if a 1D disordered potential possesses mirror symmetry, $V(-x) = V(x)$, then the eigenstates are either even or odd functions of x. As such, a wave localized on one side of such a symmetric disordered potential should elicit a corresponding antilocalized peak at the symmetric position on the other side. The distribution of pressure in a symmetric disordered potential is similar to the wave function profile in a symmetrical double-well potential. This similarity opens a way to demonstrate quantum tunneling using acoustic waves. This effect has an important application to secure communications, as the transmitted signal is reduced by disorder to the level of noise, thus excluding the possibility of signal interception, then enhanced by the symmetric part of the potential at the receiver. This is a secure method of information transmission without the need for encryption and decryption. We developed a methodology for identifying the frequency spectrum consisting of narrow doublets that correspond to eigenstates of different parities. A 2 $\times$ 30 phononic crystal with orientational disorder was fabricated for experimental observation of antilocalization. [This work is supported by the NSF under EFRI Grant No. 1741677.]

10:40

2aSA6. Nonreciprocal energy transmission in short discrete systems with strong spatiotemporal modulations.
Jiuda Wu (Concordia Univ., Montreal, 1455 De Maisonneuve Blvd. W., Rm. EV-4.139, Montreal, QC H3G 1M8, Canada, w_jiuda@encs.concordia.ca) and Behrooz Yousefzadeh (Concordia Univ., Montreal, Montreal, QC, Canada)

Introducing spatiotemporally varying properties in a phononic lattice can enable nonreciprocal transmission of energy. The hallmark of this phenomenon is the unidirectional energy transmission in infinitely long systems. In short lattices, however, identifying nonreciprocity through differences in transmitted energies (norm bias) becomes challenging because the primary contributor to nonreciprocity is the transmitted phases. Although stronger modulation can achieve a higher norm bias, it can also result in parametric instabilities and trigger large-amplitude oscillations that lead to device failure. To better understand the tradeoff between stability and norm bias in strongly modulated systems, we investigate the parametric stability of a finite one-dimensional lattice with spatiotemporally modulated elasticity. We use Floquet theory to compute the stability charts for strongly modulated systems. Thus, we can identify operating conditions that allow for a relatively large norm bias while maintaining a stable, bounded response.

10:55

2aSA7. Inverse design of three-dimensional architected elastic metamaterials using graph neural networks.
Yu-tong Wang (Acoust., Penn State Univ., 201 Appl. Sci. Bldg., Graduate Program in Acoust., University Park, PA 16802, ybw5392@psu.edu), Mourad Oudich (Acoust., Penn State Univ., State College, PA), Marco Maurizi, Xiaoyu Zheng (Mater. Sci. and Eng., Univ. of California Berkeley, Berkeley, CA), and Yun Jing (Acoust., Penn State Univ., State College, PA)

Architected metamaterials exhibit novel mechanical properties shaped by the spatial arrangement of periodic structures, rather than their constituent materials. Truss lattices, a notable subtype, are recognized for their high strength-to-weight ratio; thus, they hold significant potential for applications in robotics, aerospace engineering, and other fields. Despite recent advances in deep learning (DL) have revolutionized traditional design, researchers have mainly focused on the inverse design of quasi-static properties, leaving a gap in addressing dynamic behavior or simultaneous considerations for both aspects. To overcome this gap, we develop a novel inverse design framework to generate truss metamaterials with tailored quasi-static (stress-strain curve) and dynamic (transmission curve) properties. Our data-driven framework, based on graph neural networks, integrates a forward model into an inverse model, trained using deep reinforcement learning. . To demonstrate the model performance, finite element method simulations, uniaxial compression tests, and vibration tests are conducted to verify the properties of the optimized structures. The successful realization of user-desired properties in both quasi-static and dynamic domain can potentially accelerate the inverse design of novel materials towards applications such as lightweight and high-strength vibration isolators.

11:10

2aSA8. Mitigating low-frequency broadband aircraft noise through a structured assembly of acoustic material and devices. Tenon Charly Kone (Flight Res. Lab., National Res. Council Canada, 1200 Montreal Rd., Ottawa, ON, Ottawa, ON K1A 0R6, Canada, tenoncharly.kone@nrc-cnrc.gc.ca), Sebastian Ghinet (Flight Res. Lab., National Res. Council Canada, Ottawa, ON, Canada), Raymond Panneton (Ctr. de Recherche Acoustique-Signal-Humain, Université de Sherbrooke, Sherbrooke, QC, Canada), and Anant GREWAL (Flight Res. Lab., National Res. Council Canada, Ottawa, ON, Canada)

Addressing the challenge of attenuating low-frequency broadband noise emerges as a critical concern within the fields of aeronautics, ground transportation, and construction industries, requires innovative solutions for enhanced acoustic control. Over the last few decades, the literature has seen an increase in low-frequency noise control solutions centered around acoustic metamaterial designs. These proposed technologies exhibit promising acoustic performance, especially proving superior to conventional sound insulation materials in constrained spaces, such as in aerospace applications. Despite the efficacy of typical metamaterials in attenuating tonal noise through narrow resonant frequency maxima, practical applications reveal some challenges, as even slight variations in tonal noise frequencies can compromise the overall effectiveness of such solutions. In response to this, the present paper introduces a novel thin acoustic metamaterial design aimed at improving broadband noise attenuation at low frequencies. This design uses carefully arranged structured metamaterials within a fiberglass layer to create optimal resonance frequency bands for maximum low-frequency noise attenuation. Performance assessment in the low-frequency domain employed COMSOL Multiphysics finite element methods, predicting sound absorption coefficient and transmission loss. Results confirm the effectiveness of the proposed metamaterial design, showcasing broad noise attenuation at low frequencies.

11:25

2aSA9. Active acoustic metamaterials with independently programmable bulk modulus and full mass density tensor. Dylan Kovacevich (Mech. Eng., Univ. of Michigan, 2350 Hayward St., Ann Arbor, MI 48109, dkovac@umich.edu) and Bogdan-Ioan Popa (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

Active acoustic metamaterials consist of unit cells with sensor-driver pairs that produce a coherent response to incident waves. The effective acoustic properties of the metamaterials depend on the gains programmed between the sensing and driving components. The strength of the monopole response to the local pressure determines the effective bulk modulus and the dipole response to the local particle velocity determines the effective mass density. The unit cells are controlled individually, so in theory the metamaterials can be scaled to an arbitrary size and geometry, but prior realizations were limited to only a few cells with 1D operation. Here, we present an active acoustic metamaterial of nine cells in 2D with programmable bulk modulus and mass density tensor. We demonstrate the ability to independently program the property components (bulk modulus and four mass density elements) for any desired set within stable limits, including previously unattainable acoustic properties necessary for the fabrication of transformation acoustics devices. We also demonstrate complex effective properties, specifically a lossy layer with impedance matched to the background, such that incident waves are absorbed with no reflection. The effective properties of the active metamaterial are validated by comparing the experimental total and scattered fields to ideal simulation results.

Session 2aSC

Speech Communication: VowelFest: Honoring the Past and Celebrating the Present I

Ewa Jacewicz, Chair
Speech and Hearing Science, The Ohio State University, 1070 Carmack Road, 110 Pressey Hall, Columbus, OH 43210

Chair's Introduction—8:00

Invited Papers

8:05

2aSC1. A vowel research retrospective. Robert A. Fox (Speech and Hearing Sci., The Ohio State Univ., 1070 Carmack Rd., 110 Pressey Hall, Columbus, OH 43210-1002, fox.2@osu.edu)

One of the earliest attempts at describing/characterizing vowel quality was Daniel Jones' Cardinal Vowel system (1918) representing the total range of vowel quality in languages. It was based on auditory quality (evaluated by a trained phonetician). Much later, Ladefoged (1967) demonstrated that despite auditory equivalence, vowels were acoustically different—a function of physiological and production among speakers. The acoustic nature of vowel quality was elaborated through the introduction of easy spectral analysis (using the sonograph) exemplified by the seminal work by Peterson and Barney (1952). In the past five decades we have seen how vowels differ as a function of speaker sex, speaker size, age, speech disorder and dialect group and development of vowel spaces in second-language acquisition (much of the research through contributions of speakers in this session). In concert with these acoustic studies, we have better understanding of the nature of the perception of vowels through multidimensional scaling studies, adaptation/perceptual magnet/categorical perception studies, cross-linguistic studies, and developmental studies. Most recently, we have seen work done with regard to brain mapping and neural models of vowel perception. This talk will provide a review of past vowel research (voluminous as it is) and possible future trends.

8:25

2aSC2. Modeling vowel-inherent spectral change in varying consonant context. Michael Kiefte (Commun. Sci. and Disord., Dalhousie Univ., 1256 Barrington St., Halifax, NS B3J 1Y6, Canada, mkiefte@dal.ca)

A wide variety of experimental evidence has shown that vowel inherent spectral change (VISC) is important in vowel identification. This evidence is drawn from production data, statistical pattern recognition, and perceptual experiments with both synthetic or manipulated naturally produced speech. Experimental studies often consider vowels in a constant consonant context which makes it difficult to factor out context effects from the VISC itself. In order to examine the magnitude of these consonant effects, we developed a statistical procedure inspired by Broad and Clermont [(2014). J. Phon. 47, 47–80] in which vowel formant frequencies are approximated by a linear combination of vowel and consonant influences which vary as a function of time. Vowels extracted from a database of both spontaneous and read speech were analyzed to produce context-normalized vowel-formant tracks. Results show that vowel formant frequencies vary systematically across their duration in both spontaneous and read speech and that all consonants in both onset and coda position show significant effects on vowel production across their entire duration. Although these formant patterns are seemingly complex, perceptual evidence suggests that listeners may only attend to onsets and offsets and that deviations from a straight-line interpolation between onset and offset must be relatively large for listeners to discriminate them.

8:45

2aSC3. American English monophthong tenseness. Richard McGowan (CReSS Books, 1 Seaborn Pl., Lexington, MA 02420, mcgowan.richard.s@gmail.com)

In General American English, the point vowels of the F2 – F1 verus F1 quadrilateral, [i], [u], and [ɑ] are tense. More generally, Lindau noted the connection between acoustic peripherality and vowel tenseness [Lindau, M. (1975). "Vowel Features." *Working Papers, Phonetics Laboratory, Lund University*, **11**, p. 1]. In the case of [ae], it is the tensing of this point vowel that initiated the Northern Cities Chain Shift [Labov, W. (1994). *Principles of Linguistic Sound Change: Internal Factors*. Blackwell Publishers, Malden, MA]. The theses of this talk are firstly that a high degree of acoustic sensitivity to area function change is the reason that peripherality is highly correlated with tenseness. However the reason for the acoustic sensitivity of [ae] is different than it is for the other three point vowels. Second, a more fundamental characterization of tenseness is in its articulatory kinematics during production. The reason for this has to do with the persistence of tenseness with diachronic sound change away from the periphery. Perhaps, though, tense/lax changes appear to be more likely on the periphery than elsewhere.

9:05

2aSC4. Lifespan anatomic and vowel acoustic studies confirm prepubertal sexual dimorphism of the human vocal tract and absence of lengthening in aging. Houri K. Vorperian (Waisman Ctr., Univ. of Wisconsin-Madison, University of Wisconsin-Madison, Waisman Ctr., 1500 Highland Ave., Vocal Tract Development Lab # 427, Madison, WI 53705, vorperian@waisman.wisc.edu) and Raymond D. Kent (Dept of Commun. Sci. & Disord., Univ. of Wisconsin-Madison, Madison, WI)

To study anatomic-acoustic relations across the lifespan, anatomic studies of the developing vocal tract (VT) using medical imaging studies were related to vowel acoustic space as synthesized from published data. However, the aggregate acoustic data highlighted the need to acquire sex-specific vowel acoustic data across the lifespan using controlled methodology for data collection and analysis; also, to acquire measurements of the higher formants. Lifespan anatomic and acoustic (4-to-92 years) data helped resolve two questions. First, anatomic studies revealed significant prepubertal sex differences in the oral region of the VT (3-to-7 years) that are masked by growth rate differences between males and females. Three-dimensional anatomic findings also revealed prepubertal sexual dimorphism in select regions of the mandible and pharynx. Second, anatomic findings confirmed that VT length does not increase with aging. Acoustic studies similarly confirmed sex differences in fundamental frequency emerging at age 7; and the presence of significant prepubertal sexual dimorphism of the higher formant frequencies (F3-F4) at the earliest age studied (4 years). Findings for adults showed significant age-related decreases in fundamental frequency in women only, and no changes in formant frequencies in either sex across several decades. The results underscore the importance of sex-specific and non-uniform growth of the VT structures. [NIDCD funding support R01DC006282.]

9:25

2aSC5. Vowel perception research from formant thresholds to sentence intelligibility. Diane Kewley-Port (Speech and Hearing Sci., Indiana Univ., 3803 E St Remy Dr., Bloomington, IN 47401, kewley@indiana.edu)

Historically, research on the contribution of vowels to speech understanding lagged that of consonants. Speech synthesis techniques were then developed that established that the primary acoustic features of vowels are formant frequency, fundamental frequency, speech dynamics and naturalness. For speech perception research, precise control of the features in vowel stimuli was required. Starting in the 1980s, the Klatt formant synthesizer was the first important tool, and by 2000 Kawahara's STRAIGHT synthesizer generated nearly natural speech. As a baseline for vowel perception, my research sought to determine psychophysical thresholds for F1 and F2 under ideal conditions. This talk reports contributions from my vowel perception studies on three questions. First, how do formant thresholds change with speech context (isolated vowels up to sentences), across age and with hearing impairment? Second, do vowels or consonants contribute more to intelligibility in noise-interrupted sentences? Third, given formant dynamics, how is intelligibility affected as consonant-vowel boundary conditions are manipulated across age and hearing impairment? Significant results include: (1) Vowels carry more information about sentence intelligibility than consonants for both young and older listeners; (2) Even though older listeners' performance is reduced compared to young, hearing impairment has a greater negative impact than age-related cognitive decline.

9:45–10:00 Break

10:00

2aSC6. Vowels in clear and conversational speech: Intelligibility for listeners with hearing loss. Sarah H. Ferguson (Commun. Sci. and Disord., Univ. of Utah, 390 South, 1530 East, Rm. 1201, Salt Lake City, UT 84112, sarah.ferguson@hsc.utah.edu)

While my research during the past two decades has explored numerous perceptual properties of clear versus conversational speech, it all began with vowel intelligibility. In this presentation, I review my research on vowel intelligibility in listeners with hearing loss and whether and how it changes when talkers adopt a clear speaking style. I also discuss acoustic properties that may explain these changes and the potential interaction between clear speech acoustic properties and listener hearing status.

10:20

2aSC7. What babies bring to our understanding of vowel perception. Linda Polka (School of Commun. Sci. & Disord., McGill Univ., 2001 McGill College Ave., 8th Fl., SCSD, McGill University, Montreal, QC H3A 1 G1, Canada, linda.polka@mcgill.ca)

Vowels are produced and perceived very early in infancy and occupy a central role in speech communication across the lifespan. Many phenomena first uncovered in vowel research with adults were later explored with infants. This produced important findings that inform our understanding of vowel perception and production development. In this talk, I will highlight several perception findings that were initially discovered in infants which now also direct fruitful lines of research with adults. I will focus on two perceptual biases—the focal vowel bias and the infant talker bias—each involving information conveyed by vowels. The focal vowel bias identifies a universal vowel perception bias that is germane across the lifespan. Elaborating the mechanism(s) behind this bias can lead us to a more principled understanding of basic vowel perception processes. The infant talker bias – first identified in infants and now also in adults – reveals a robust bias favoring infant conspecific vocalizations. This bias appears to impact infant development directly and also indirectly via its' positive effect on parenting behaviors. Going forward, the interplay of research across age groups will continue to bring us a deeper understanding and appreciation of the ubiquitous role of vowels in human cognition.

10:40

2aSC8. Perception of age, gender, and talker height from children's vowels. Peter F. Assmann (Psych., Univ. of Texas at Dallas, 800 W Campbell Rd., Richardson, TX 75080, assmann@utdallas.edu), Abbey L. Thomas (Brain and Behavioral Sci., The Univ. of Texas at Dallas, Richardson, TX), and Santiago Barreda (Linguist, UC Davis, Davis, CA)

In his 1989 review of vowel perception [J. Acoust. Soc. Am. 85, 2088–2113], Terry Nearey observed that the acoustic properties of vowels vary substantially across talkers due to differences in talker size. Taller speakers tend to have lower formant frequencies and lower fundamental frequencies than shorter speakers, and listeners appear to take these relationships into account in vowel identification. In this talk, we follow up some of the questions raised in that paper, reviewing recent findings on the perception of age, gender, and

talker height in children's voices. Children exhibit substantial age-related changes in fundamental and formant frequencies due to growth of the larynx and vocal tract, presenting an interesting opportunity to explore the perceptual consequences of these changes. Listeners' perceptual judgments of age and gender reveal complex interdependencies, consistent with the idea that these indexical attributes are jointly estimated, in a manner that reflects their shared dependence on acoustic parameters related to perceived talker height.

11:00

2aSC9. My great big problem with vowels. James E. Flege (Speech & Hearing Sci., Univ. of Alabama at Birmingham, Via del Moro 12, Tuscania, Viterbo 01017, Italy, jimflege@gmail.com)

How vowels are produced in a second language (L2) is determined by several factors. One is cross-language mapping: how/if vowels in the L2 phonetic system are categorized in terms of native language (L1) vowels and the extent to which they are judged to resemble the L1 vowels perceptually. Another is input: the quantity and quality of the L2 vowels to which learners have been exposed. L1-L2 mapping patterns are sometimes reported in studies of L2 vowel learning, but they may differ across individuals and change as learners gain L2 experience. Coarse estimates of quantity of L2 input are usually reported but estimates of exposure to foreign-accented L2 input are virtually non-existent. Even if we had adequate measures of L2 input, other serious problems would remain. Once the L1 phonetic system is established, language users may attend less to the surface phonetic properties of vowels they hear when conversing in the L1 and also in the L2. This talk will focus on problems inherent in (a) understanding how the phonetic input received in an L2 is used, and (b) gauging degree of L2 learning success.

11:20

2aSC10. Unexpected findings from L2 speech research can help us understand how humans process vowels. Ocke-Schwen Bohn (Aarhus University, Sejts Alle 20a, Risskov DK-8240, Denmark, engosb@hum.au.dk)

Research on nonnative speakers' vowel production and perception has provided us with a number of surprising insights on how learners cope with nonnative vowels. Three of these initially unexpected, yet later solidly replicated, findings will be presented: (1) A decline in intelligibility of nonnative vowels as general proficiency improves (when an increase in intelligibility will be expected), (2) the use of acoustic cues in L2 vowel perception that cannot be attributed to transfer from the native language, and that are nonfunctional in the L2, and (3) the maintenance from infant speech perception in adult cross-language vowel perception of a bias favoring peripheral vowels. This presentation will discuss implications of these findings for our understanding of how vowel categories coexist in the minds of multilinguals, and of universally (native-language independent) preferred ways of vowel perception.

Session 2aSP

Signal Processing in Acoustics, Biomedical Acoustics, Physical Acoustics, Computational Acoustics, and Underwater Acoustics: Data Augmentation in Signal Processing: Advancing Performance through Artificial Data Generation

Yongsung Park, Cochair
University of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0238

David R. Dowling, Cochair
Mechanical Engineering, University of Michigan, Dept. of Mech. Eng., Univ. of Mich., 1231 Beal Avenue, Ann Arbor, MI 48109

Chair's Introduction—8:00

Invited Papers

8:05

2aSP1. The origins of frequency-difference and frequency-sum beamforming. Shima Abadi (Univ. of Washington, 185 Stevens Way, Paul Allen Ctr. – Rm. AE100R, Seattle, WA 98195, abadi@uw.edu), David R. Dowling (Dept. of Naval Architecture and Marine Eng., Univ. of Michigan, Ann Arbor, MI), Heechun Song (SIO, UCSD, La Jolla, CA), and Kevin J. Haworth (Dept. of Internal Medicine, Univ. of Cincinnati, Cincinnati, OH)

Beamforming the signals recorded by an array enables the determination of sound source location(s) or the arrival directions of ray paths between a sound source and the receiving array. Frequency-difference and frequency-sum beamforming are beamforming techniques that provide out-of-band information from in-band signal frequencies. Interestingly, the out-of-band frequencies can be chosen by the user, within limits set by the signal recordings, to achieve desired properties of the beamformed output, such as: increased resolution, reduced sidelobes, or greater robustness to random scattering. Both techniques are general and are not limited to any particular acoustic environment, frequency range, or array geometry. Frequency-sum beamforming, generates higher-frequency information from lower frequency signal components, enhancing beamforming results in scenarios with random scattering between the source and the receivers. However, it is limited by artifacts arising from cross-terms when multiple source signals are present in the same bandwidth. Conversely, frequency-difference beamforming manufactures lower-frequency information from higher frequency signal components, effectively mitigating the impact of spatial aliasing in situations where the receiving array is sparse. This presentation delves into the origins of frequency-difference and frequency-sum beamforming, presents the fundamental mathematics underlying their algorithms, and showcases their performance via simulations and experimental results. [Work supported by ONR.]

8:25

2aSP2. Beamforming applications of the frequency-difference acoustic autoproduct. Alexander S. Douglass (Oceanogr., Univ. of Washington, 2010 AL, 1231 Beal, Ann Arbor, MI 48109, asdougl@umich.edu), Heechun Song (SIO, UCSD, La Jolla, CA), and David R. Dowling (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

Frequency-difference beamforming (Abadi *et al.*, 2012, JASA, 132, 3018–3029) is an array signal processing technique that overcomes the limitations of the spatial Nyquist criterion by utilizing the acoustic autoproduct to shift the processing to below-band frequencies. This is accomplished using a quadratic product of complex signal amplitudes at different frequencies, resulting in wave propagation information at the out-of-band difference-frequency. The resulting field is capable of mitigating many of the in-band challenges often associated with high frequency acoustic signal processing, including sparse receiver arrays and features of the physical environment that are on a scale significant for the in-band wavelength. This presentation uses both laboratory and ocean-based experiments to demonstrate capabilities of the method. The mitigation of sparse array aliasing effects on beamforming (caused by elements that are spaced by many wavelengths) in both a laboratory water tank and an ocean environment utilizing data collected during the KAM11 experiment are considered. Additionally, the method is used to localize a high frequency source in the presence of strong, random scatterers in a water tank experiment. [Work sponsored by NAVSEA and ONR.]

8:45

2aSP3. Augmenting sparse arrays in ocean acoustics: A Gaussian Process approach. Zoi-Heleni Michalopoulou (Dept. of Mathematical Sci., New Jersey Inst. of Technol., 323 M. L. King Boulevard, Newark, NJ 07102, michalop@njit.edu) and Peter Gerstoft (Univ. of California, San Diego, La Jolla, CA)

Source localization and geoacoustic inversion performance depends on the location of receiving phones, their spacing, their number, and the array aperture. Performance suffers when practical design issues limit the capabilities of the array. In this work, sparse arrays are augmented leading to the generation of virtual arrays for better sampling of the ocean and, thus, improved estimation performance. To that effect, Gaussian Processes are employed, which are shown to create high-fidelity field "measurements" at virtual, densely spaced hydrophones. Kernel functions are key building blocks in the implementation of Gaussian Processes, as they quantify the field coherence at neighboring spatial points. Functions of interest are the squared exponential, Matern, and plane wave kernels. We validate our method with application to synthetic data as well as data collected during the Seabed Characterization Experiment conducted in 2022. [Work supported by ONR.]

9:05

2aSP4. A modular neural network to quickly approximate the modal dispersion in coastal waters. Arthur Varon (GIPSA-Lab, 11 Rue des Mathématiques, Saint-Martin-d'Hères, Isère 38400, France, arthur.varon@gipsa-lab.grenoble-inp.fr), Jérome Mars (GIPSA-Lab, Saint-Martin-d'Hères, France), and Julien Bonnel (Woods Hole Oceanographic Inst., Woods Hole, MA)

Low-frequency acoustic propagation modeling in coastal waters usually relies on numerical models based on modal theory such as Kraken and Orca. These models compute the modal parameters (e.g., modal wavenumbers and depth functions) that can be used in the calculation of the acoustic field. Their repeated use in broadband applications, or for inversion purposes, comes with a notable computational cost. To mitigate this, a modular neural network (NN) was trained to approximate modal parameters for varying modes and frequencies, across diverse environments, with variable water sound speed profile and variable seabed geoacoustic parameters. The training dataset is generated using Kraken and the NN is evaluated on many environments not seen during training. Once trained, the NN can make broadband predictions without prior knowledge on the number of modes, even when the number of modes changes over the frequency-band of interest. This approach reduces computation time compared to the original forward propagation model, while maintaining high precision. The effectiveness of our method is demonstrated through transmission loss calculations and a simulated geoacoustic inversion scenario.

Contributed Papers

9:25

2aSP5. Frequency difference-wavenumber analysis for direction-of-arrival estimation in sparse vertical array. Donghyeon Kim (Korea Maritime and Ocean Univ., 727 Taejong-ro, Yeongdo-Gu, Busan 49112, Korea (the Republic of), donghyeon.ual@gmail.com), Gihoon Byun (Korea Maritime and Ocean Univ., La Jolla, CA), and J. S. Kim (Korea Maritime and Ocean Univ., Busan, Korea (the Republic of))

Frequency–wavenumber (*f–k*) analysis can determine the direction-of-arrival (DOA) for broadband signals captured by a vertical array [M. J. Hinich, J. Acoust. Soc. Am., **69**, 732–737 (1981)]. The sparse vertical array generates numerous sidelobes in the *f–k* domain due to aliasing errors resulting from the spatial sampling. This presentation introduces the frequency difference-wavenumber ($\triangle f$–k) analysis, expanding the application of *f–k* analysis to the sparse vertical array by adopting the concept of frequency difference. The relationship between the frequency difference beamforming and the $\triangle f$–k analysis is also discussed. Experimental results verify the effectiveness of the proposed $\triangle f$–k analysis in estimating the DOA of snapping shrimp clicks (11–24 kHz) recorded using a sparse vertical array in a shallow water experiment. During the experiment, a vertical array with an element spacing of 3.75 m (i.e., design frequency = 200 Hz) is utilized, which is extremely sparse because it corresponds to 27.5 wavelengths at the lowest frequency.

9:40–9:55 Break

9:55

2aSP6. High-resolution matched autoproduct processing using sparse Bayesian learning for multiple source localization. Ze Yuan (State Key Lab. of Acoust., Inst. of Acoust., Chinese Acad. of Sci., Inst. of Acoust., Chinese Acad. of Sci., No. 21 North 4th Ring Rd., Haidian District, Beijing, China, Beijing 100190, China, yuanze@mail.ioa.ac.cn), Haiqiang Niu (State Key Lab. of Acoust., Inst. of Acoust., Chinese Acad. of Sci., Beijing, China), Zhenglin Li (School of Ocean Eng. and Technol., Sun Yat-sen Univ., Zhuhai, China), and Wenyu Luo (State Key Lab. of Acoust., Inst. of Acoust., Chinese Acad. of Sci., Beijing, China)

Matched autoproduct processing (MAP) is a nonlinear array-signal-processing technique employed for source location estimation within the framework of matched field processing (MFP). Operating by matching frequency-difference autoproducts instead of the pressure field, MAP induces a downshift in frequency. While offering reduced sensitivity to environmental mismatch, MAP presents drawbacks such as compromised spatial resolution, broadened mainlobe, and a diminished peak-to-sidelobe ratio within the ambiguity surface. These characteristics make the application of MAP challenging in scenarios involving multiple sources, especially when dealing with close proximity or a weak source alongside a stronger one. To address these limitations, MAP estimation is formulated as a sparse signal reconstruction problem, solved through sparse Bayesian learning (SBL). In the context of preserving frequency-difference robustness, the proposed method demonstrates a narrower mainlobe and diminished sidelobe levels compared to the original MAP. This improvement extends the applicability of MAP to multi-source localization scenarios. Finally, we validate the superior performance of the proposed method through both simulations and experimental data.

10:10

2aSP7. Artificial data augmentation using braided human features for underwater acoustics. Ananya Sen Gupta (Dept. of Elec. and Comput. Eng., Univ. of Iowa, 103 S Capitol St., Iowa City, IA 52242, ananya-sengupta@uiowa.edu), Andrew J. Christensen (Elec. and Comput. Eng., Univ. of Iowa, Iowa City, IA), Anjali Mathews (Dept. of Elec. and Comput. Eng., Univ. of Iowa, Iowa City, IA), Subhajit Das (Subhangik Dance Co., Bandel, West Bengal, India), and Ivars P. Kirsteins (NUWCDIVNPT, Newport, RI)

Underwater acoustics datasets provide rich environmental information that can potentially be mined using popular machine learning architectures. However, such knowledge discovery is typically limited by the high feature uncertainty, lack of robust ground truths, limited availability of public-domain data, and difficulty of reproducing experimental conditions in field experiments. This poses a data science challenge to study persistent target features, as they can only be learnt and classified robustly if large scale robust training and testing datasets are available. This “nano-data” problem can be potentially solved by augmenting the training and testing data repositories with a two-pronged approach: (i) physics-driven simulations of desired features, which offer interpretable ground truths but typically cannot emulate practical open-sea experiments; (ii) geometric-proxy augmentation using creatively constructed non-domain datasets that emulate the desired feature geometries and environmental effects. For example, a skilled dancer can emulate specific features geometries through geometric contortions of the human body. Such choreographed dance movements can be reproduced reliably against different types of stage environments, structured and unstructured, to emulate the oceanic environment under different conditions. We will present preliminary results in data augmentation from both approaches and discuss the trade-offs between physics-oriented simulations and geometry-driven proxy augmentation. [ONR grant N000142312503.]

10:25

2aSP8. Increasing the accuracy of ISO 354 and ASTM C423 through modal identification. Dario D’Orazio (Univ. of Bologna, Viale Risorgimento, 2, Bologna 40126, Italy, dario.dorazio@unibo.it), Andrea Zaccarini (Univ. of Bologna, Bologna, Italy), and Fiorella Falciano (Univ. of Bologna, Bologna, Italy)

Measuring the acoustic properties of materials is a highly challenging task. Previous studies have highlighted errors in the absorption coefficient above 200 Hz, leading to absorption coefficient values higher than one. These errors are often attributed to differences in modal decays with and without the specimen. This effect is amplified in the case of highly reactive materials, such as innovative materials, that significantly impact the acoustic field and, consequently, the modal behaviour of the room. This work aims to reduce the difference between measured reverberation time values, with and without the specimen, through Data Augmentation. Modal identification was performed on measured impulse responses of porous specimens, where the alpha is known, allowing the determination of modal decay times and reverberation times within the frequency range of 70–180 Hz. A statistical analysis of natural modes and numeric solutions enables the identification of outliers. In this way, even in a non-well-optimized reverberation room, it is possible to enhance the accuracy of acoustic absorption measurements.

Invited Paper

10:40

2aSP9. Frequency-difference beamforming and sparse processing. Yongsung Park (Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0238, yongsungpark@ucsd.edu) and Peter Gerstoft (Univ. of California, San Diego, La Jolla, CA)

Frequency-difference processing enables the estimation of the direction of arrival (DOA) for sources beyond the spatial aliasing frequency. The beamforming method takes advantage of the frequency difference between multiple frequencies, enabling processing at a lower frequency than the aliasing frequency. During this process, the number of DOAs we need to estimate becomes squared, and we require a high-resolution DOA estimator. Sparse processing has demonstrated high-resolution capabilities in DOA estimation, resulting in beamforming spectra with sharp peaks and enhancing resolution for accuracy. This presentation proposes the utilization of both frequency-difference processing and sparse processing. The suggested method is an aliasing-free beamformer for high-frequency sources using frequency-difference processing and achieves high-resolution capability using sparse processing.

Session 2pAAa

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

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

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

Contributed Papers

1:00

2pAAa1. How can desks and ceilings influence room acoustic criteria in classrooms? Findings from measurements and wave-based simulations. Giulia Fratoni (Univ. of Bologna, Viale del Risorgimento 2, Bologna, BO 40136, Italy, giulia.fratoni2@unibo.it) and Dario D'Orazio (Univ. of Bologna, Bologna, Italy)

Sound diffusion is essential in achieving the required speech intelligibility criteria in classrooms ($T_{;30}$, $C_{;50}$, STI). Predictive formulas employed and recommended by international standards generally rely on the ideal diffuse-field theory, notwithstanding its applicability limitation in most feasible real-world scenarios. Yet, along with the sound-absorbing features of the surfaces, it is crucial to quantify any potential diffraction effect caused by furnishing elements and ceiling treatments. The present work uses acoustic measurements and wave-based numerical models to explore the impact of desk layouts and materials in false ceilings on sound diffusion. The single desk arrangement increases the sound diffusion, here quantified in terms of measured $T_{;30}$ standard deviation, more than other furniture layouts, e.g., desks arranged in circles. Moreover, preliminary results from acoustic measurements and finite-element analysis show that inhomogeneous treatments of the suspended ceiling – porous and perforated modules – increase sound diffusion through material discontinuity even at low frequencies. The match between predictive formulas, experimental results, and simulations confirms the reliability and accuracy of the acoustic design process when involving diffraction effects.

1:15

2pAAa2. Acoustical enhancement of a multi-purpose auditorium in an academic setting. Anthony Steven Garcia Mora (Escuela Superior Politécnica del Litoral, Guayaquil, Ecuador), Luis Bryan Vanegas Cruz (Escuela Superior Politécnica del Litoral, Guayaquil, Ecuador), Galo Durazno (Escuela Superior Politécnica del Litoral, Guayaquil, Ecuador), Carlos Yoong (Wood PLC, 2020 Winston Park Dr #700, Oakville, ON L6H6X7, Canada, carlos.yoong@woodplc.com), and Pedro Segovia Gonzalez (Universidad de las Artes, Guayaquil, Ecuador)

The Coastal Polytechnic School in Guayaquil, Ecuador is the leading engineering school in the country. Their Faculty of Mechanical Engineering currently have an auditorium that it is being used for different type of events such as conferences and artistic presentations. The design of the auditorium never considered acoustics, and the space currently presents issues with high reverberation time, deficient speech intelligibility and poor sound system. As part of a capstone project, a group of students from the Faculty conducted a detailed assessment of the space and develop solutions to enhance its acoustical properties for the required space functionality. Onsite measurements were taken and incorporated into a specialized software to calibrate the properties of the current space. Various solution scenarios were then evaluated computationally to find a solution suitable for a multi-purpose space.

1:30

2pAAa3. The investigation of sound and position in rehearsal rooms, the data is telling. Carolyn Dzul (Recording Arts & Sci., Peabody Inst. of The Johns Hopkins Univ., 606 Saint Paul St., P.O. 435, Baltimore, MD 21202, carolyn.dzul@gmail.com) and Ian B. Hoffman (Recording Arts & Sci., Peabody Inst. of The Johns Hopkins Univ., Baltimore, MD)

This paper includes the investigation of music rehearsal environments. Mainly focused at the high school level, it measures criteria that impact the health and performance of young musicians. This study goes beyond the ISO 3382: Measurements of room acoustic parameters and the Norwegian standard NS 8178 considerations. It assesses sound exposure and risks as they relate to room acoustics goals and data. The measurements of at least five rehearsal rooms in the Baltimore/DC area are conducted with orchestra and wind ensembles. A questionnaire to obtain observation data from music directors was considered in the analysis. The collected measured data focus on the actual sound exposure levels experienced by the music director and students. A consistent, repeated, triangulated setup of measurement locations allowed for both exposure and cross-room communication evaluations. The results revealed potential inconsistencies between healthy sound exposure and the goals/targets noted in the standards mentioned above.

1:45

2pAAa4. A survey of stage acoustic conditions for choral performances. Francesco Martellotta (Dept. Architecture, Construction and Design, Politecnico di Bari, Via Orabona 4, Bari 70125, Italy, francesco.martellotta@poliba.it), Chiara Rubino (Dept. Architecture, Construction and Design, Politecnico di Bari, Bari, Italy), and Stefania Liuzzi (Dept. Architecture, Construction and Design, Politecnico di Bari, Bari, Italy)

Choral performances, due to a very large number of non-professional groups often operating in small cities, frequently take place in non-dedicated spaces like churches, open atria, and other spaces with unusual acoustic conditions. This results in performances taking place in spaces without a real "stage" and consequently, in acoustic conditions that, as reported by several interviews, lack support and communication among singers. In order to better understand the effective acoustic conditions experienced in real spaces that frequently host choral performances, an on-site survey was carried out in six venues including one semi-open atrium and five churches of different styles and dimensions. Measurements were carried out according to ISO 3382-1 requirements, also taking into account the most recent proposals in terms of stage support to account for actual source directivity and presence of reflecting elements close to the source. Results showed that smaller spaces where vertical surfaces where more likely to be closer to the performers behaved better than large spaces where side volumes (like transepts or high domes) mostly withdrew acoustic energy yielding STearly values much lower than −12 dB.

2p TUE. PM

2:00

2pAAa5. Acoustical investigations of multipurpose spaces at community facilities. Mandy Chan (2000 Argentia Rd., Plaza 1, Ste. 203, Mississauga, ON L5N 1P7, Canada, machan@hgcengineering.com), Michael Kundakcioglu (Mississauga, ON, Canada), and Alex Lorimer (Mississauga, ON, Canada)

Multipurpose spaces in community and recreational facilities hold significant value, catering to a variety of uses and accommodating users of diverse age groups. This article discusses investigations conducted at nine municipally owned facilities including community centres, arenas, and a senior centre related to reportedly poor acoustic conditions in multipurpose spaces. The acoustical measurements of the as-found conditions, relevant criteria, and analysis are outlined. The sound isolation, acoustical treatment for interior reverberation control, and background noise from HVAC are discussed as the contributing factors for the subjective acoustic concerns.

2:15

2pAAa6. Understanding diffusion versus scattering devices and how they work. Richard L. Lenz (RealAcoustix LLC, 2361 B Ave., Ogden, UT 84401, RL@RealAcoustix.com)

Nearly 50 years ago, Dr. Manfred Schroeder defined the word "Diffuser" as both a process and a product. Since then, many product manufacturers, acousticians and academics have come to use the term for many different products and designs. A careful reading of Dr. Schroeder's 1975 and 1978 papers reveals a lot of significant information as to why certain products are diffusers and why other products do not meet his criteria for diffusion. This paper will show, with the aid of Schroeder's interpretations and modern particle simulations, exactly what he had in mind when he, and others he was working with, created what ultimately became the Quadratic Residue Diffuser or QRD. Additionally, it will show his journey from primitive-root theory to his final quadratic designs. A discussion of the term "Scattering" will also be included to define how Schroeder uses the word and why we need to be careful as to how it is used as well. Test data showing phase models will be used to show the importance of phase manipulation in describing diffusion.

2:30

2pAAa7. Test method for diaphragmatic bass traps in a revberation chamber. Richard L. Lenz (RealAcoustix LLC, 2361 B Ave., Ogden, UT 84401, RL@RealAcoustix.com)

For decades the limitations of ASTM C423 and ISO 354 have been confined to frequencies above 100Hz. While many attempts have been made to provide low frequency test data using these standards, they all go beyond the actual standards. A significant problem with testing low frequencies in reverberation chambers has to do with using the diffuse field, or standard test area, of the lab. Low frequencies do not propagate in these areas to the level of the standard test range. Additionally, the devices, specifically diaphragmatic low frequency absorbers, are not used in the diffuse field of rooms. They are generally designed to be placed in, or near, corners of rooms, or modal areas, where low frequencies do propagate. This test method will show experiments and the process of determining how to test diaphragmatic low frequency devices using the modal zone of the chamber. These tests are intended to be presented to ASTM in order to develop an addendum to C423 for testing low frequencies and the lab limitations based on size and other factors.

2:45–3:00 Break

3:00

2pAAa8. An analysis and retrofit of the acoustics at Image Creators Health and Beauty Salon. Donna A. Ellis (Architectural Acoust., Lines by Nature LLC, 415 Riggs Ave., Severna Park Md. 21146, Severna Park, MD 21146, dellisdona@gmail.com)

This paper discusses the analysis and retrofit of the acoustics in a high-volume beauty salon in Severna Park, Md. The major issues in what was intended to be a serene environment are reverberation times of 1–1.68 s in the mid-to upper frequency range and Background noise levels ranging from 73.4 to 64.6 dB in the mid to low frequency range. Employee and customer complaints include ear pain and tinnitus due to prolonged exposure to high background levels, heightened stress, vocal strain, headaches and poor speech intelligibility. Existing analysis and the acoustical retrofit resolution will be demonstrated.

3:15

2pAAa9. A real-world Lexicon 960L reverberation chamber: Simulating a hardware reverberation unit in virtual acoustics. Aybar Aydin (Sound Recording Area, McGill Univ., McGill University, Dept. of Music Res., Montreal, QC H3A 1E3, Canada, aybar.aydin@mail.mcgill.ca), Vlad Baran, Kathleen Ying-Ying Zhang (Sound Recording Area, McGill Univ., Montreal, QC, Canada), Jack Kelly, Richard King, and Wieslaw Woszczyk (Music Res., McGill Univ., Montreal, QC, Canada)

The second generation of the Virtual Acoustic Technology (VAT) Laboratory at McGill University features a new real-time auralizer with a feedback canceller developed by CCRMA at Stanford University, allowing for the simulation of virtual acoustic environments with exceptionally high gain. This study is part of an ongoing research effort focused on integrating algorithmic reverberation tools designed for audio post-production into virtual acoustics at McGill University's VATLab. Previous work has been done using impulse responses (IRs) captured from various acoustic spaces. In contrast, this study focuses on using IRs captured from the legendary Lexicon 960L hardware reverberation unit and using them in the VATLab for recording sessions with musicians. Various 5.1 multichannel presets have been captured as IRs and 3 groups of related 5 channel IRs have been loaded into the existing 15 speaker system of VATLab to simulate a real world, physical Lexicon 960L "environment" through virtual acoustics. Objective measurements following the ISO 3382-1 and 3382-2 standards in the VATLab have been performed to measure the effect of the physical room and analyze the effects of changing different algorithmic reverb parameters such as Diffusion, Early Level Master Control, Early Rolloff, Early or Reflection Delays on the simulated acoustical environments.

3:30

2pAAa10. Musician-led performance perspectives in virtual acoustics. Kathleen Ying-Ying Zhang (Sound Recording Area, McGill Univ., McGill University, Dept. of Music Res., Montreal, QC H3A1E3, Canada, yingying.zhang@mail.mcgill.ca), Aybar Aydin, Vlad Baran, Richard King, and Wieslaw Woszczyk (Sound Recording Area, McGill Univ., Montreal, QC, Canada)

Musical performance is a complex task that involves juggling performance techniques, creative expression and auditory feedback, all of which in turn affect a musician's perception of their acoustic environment. Synthesized here are several recent studies conducted at McGill University in variable acoustic spaces: the Multimedia Room (with a Meyer Constellation system) and the Immersive Media Lab (utilizing a proprietary Virtual Acoustics Technology system). In studying the perception of room response, variable acoustic environments allow researchers to physically and/or virtually change the character of a space around a musician without the need for a physical venue relocation. These experiments explore different methodologies and approaches to studying the perception of room acoustic response of various vocalists and instrumentalists who completed both solo and group performance tasks. After contextualizing these studies with previous work done in this area, lessons recently learned point towards new methodologies for musician-led perspectives on room perception.

3:45

2pAAa11. Virtual acoustics in distributed performance over a closed audio network. Kathleen Ying-Ying Zhang (Sound Recording Area, Dept. of Music Res., McGill Univ., Montreal, QC H3A1E3, Canada, yingying.zhang@mail.mcgill.ca), Vlad Baran, Aybar Aydin, Michail Oikonomidis, Richard King, and Wieslaw Woszczyk (Sound Recording Area, McGill Univ., Montreal, QC, Canada)

We explore an application of active acoustics used to produce a shared virtual environment for live musical performance. As part of a concert given in the Immersive Media Lab at McGill University, musicians and audience members were located in adjacent but acoustically isolated spaces on the

same digital audio network. With computer generated effects and live instruments, the concert consisted of electroacoustic performances that utilized dynamic virtual environments produced by our Virtual Acoustic Technology (VAT) system as an improvisatory partner and a mixing device to blend the diffusion of electronic and acoustic musical sources. The performance, including its evolving virtual environment, was captured using spatial microphone techniques and distributed in real-time to the audience over an immersive loudspeaker system in an adjacent control room. Audience members were given the opportunity to visit the musicians' performance space in order to compare the reproduction to the original environment. Overall, the blending of computer-generated and acoustic sources created a specific use case for virtual acoustics, while the immersive capture and distribution method examined an avenue for producing a real-time shared experience. Future work in this area includes audio networks with multiple virtual acoustic environments and distributions.

4:00

2pAAa12. Acoustics of two Hindu temples in southern India. Shashank Aswathanarayana (Performing Arts, American Univ., Jack Child Hall, Rm 202, American University, Washington, DC 20016, shashank@american.edu) and Braxton Boren (Performing Arts, American Univ., Washington, DC)

Acoustically important aspects of Hindu worship include chants, bells, conch-shells, and gongs. Conch-shells and gongs are used at various times during puja rituals (Prasad and Rajavel, 2013), throughout which texts from the Vedas and other Sanskrit scriptures are chanted. These Vedic chants have phonetic characteristics such as pitch, duration, emphasis, and uniformity (Beck, 1995; Prasad, 2013). Traditional methods of acoustic characterization (e.g., for churches) are based on time domain characteristics like reverberation time and clarity. We term this as, "time domain soundscape of worship." This is vastly different from Hindu worship which involves extensive use of bells, conch-shells, and gongs in puja rituals, all of which produce unique sonic characteristics making frequency of sound very crucial which we term, "frequency domain soundscape of worship." Beck (1995, 2006, 2012) has explored the sonic aspects of Hindu tradition as they relate to the religion and drawn comparisons with religious practices in other cultures where pertinent. However, a comprehensive acoustic analysis of temples and the characteristics of these sounds within temples is yet to be done. In this paper, we analyze the impulse responses and decay curves measured at the Virupaksha and Vijaya Vittala temples in Southern India.

4:15

2pAAa13. Development of an acoustic design support tool for HVAC ventilation units. Malek Khalladi (MJM Acoust. Consultants Inc., 753 Rue Sainte-Hélène, Longueuil, QC J4K 1K5, Canada, mkhalladi@mjm.qc.ca) and Hong Tong (MJM Acoust. Consultants Inc., Longueuil, QC, Canada)

HVAC equipment is one of the main sources of noise inside or outside a building. Furthermore, many people are exposed daily to HVAC noise, which can lead to health-related troubles. Therefore, an HVAC mechanical unit must be selected to provide an acceptable sound level transmitted to the occupied spaces of a building and not disturb the community. This paper presents a prediction tool named Sound Prediction Tool (SPT) developed by MJM Acoustical Consultants Inc. in 2021 that allows the engineer (i) to design an HVAC unit with specific mechanical components (filter, coil, etc.) according to his input parameters and (ii) to predict and optimize the sound levels produced by the selected design through its casing and openings (inlet and outlet air). This tool has been validated firstly with acoustical tests ''*in situ*'' on several separate Air Handling Units (AHU) manufactured by our customer and secondly with Finite Elements Method (FEM) models using Comsol-Multiphysics. Several examples are presented to demonstrate the usefulness of the proposed tool.

4:30

2pAAa14. The acoustical consultant in the context of the construction of a building. Hong Tong (MJM Acoust. Consultants Inc., 753 Ste-Helene St., Longueuil, QC J4K 3R5, Canada, htong@mjm.qc.ca)

Acoustical engineering or consulting is a niche field. However, it touches on many aspects of building construction and will inevitably cross path with other disciplines such as the architect and the mechanical engineer. The purpose of this article is to expose the role of the acoustical consultant as an important stakeholder in the context of the construction of a building. From his or her role at the design phase of a project to the construction up to the reception. The article will also discuss about the different challenges and constraints as an acoustical consultant and will show the impacts of not implicating one. The technical aspect of the acoustical consultant is important, but there is also aspects of project management that needs to be considered. The moment there is a requirement for acoustics, an acoustical expert must be part of the project team at the earliest stages to deliver a successful project.

Session 2pAAb

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

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

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

Chair's Introduction—3:20

Invited Papers

3:25

2pAAb1. Planes, trains, and automobiles: Three case studies of building envelope design addressing transportation noise. Jessica S. Clements (Acoust. Studio, Newcomb & Boyd, LLP, 303 Peachtree Ctr. Ave. NE, Ste. 525, Atlanta, GA 30303, jclements@newcomb-boyd.com), John Garretson (Acoust. Studio, Newcomb & Boyd, LLP, Atlanta, GA), and Kirsten Barringer-Cook (Acoust. Studio, Newcomb & Boyd, LLP, Atlanta, GA)

This presentation will focus on three case studies of designs to limit transportation noise through the building shell to support the successful use of the building. The first project focuses on an office building located close to beautiful water views that desired a building shell made primarily of glass. However, it will be located directly under the flight path of a nearby US Airforce base. The second project considers a hotel conference center located directly adjacent to a heavy rail line. The third project address movie sound stages located adjacent to a major highway and directly under the flight path of Hartsfield Jackson International Airport, known as the busiest airport in the world. Site measurements were taken for each of these locations and a detailed review of the data used to determine which noise sources and frequencies of concern would be the focus of the design. The presentation will present examples of the collected data, the design strategies used, and discuss the challenges faced by the design team.

3:45

2pAAb2. Transit-oriented development: A case study of exterior-to-interior noise control for buildings near transit and highway noise sources. Leisa Nalls (Wilson Ihrig, 5900 Hollis St., Ste. T1, Emeryville, CA 94608, lnalls@wilsonihrig.com), Deborah Jue (Wilson Ihrig, Emeryville, CA), and Silas Bensing (Wilson Ihrig, New York, NY)

A case study of acoustical design for mixed-used, transit-oriented development is presented. The project is two, six-story mixed-used residential buildings near a rapid transit station and heavily traveled elevated highway in the San Francisco Bay Area. The buildings are metal frame construction over concrete podiums. The California Building Code requires the building shell to be designed to provide interior noise from exterior noise sources not to exceed 45 dBA Ldn. Exterior noise due to trains and buses serving the transit station, and traffic on the elevated highway, required high sound attenuating windows and acoustically rated walls and roofs. Initially, none of the exterior wall assemblies included batt insulation in the stud cavities, only continuous rigid insulation behind the façade. Modeling was done to compare the exterior wall assemblies with and without batt insulation in the stud cavities combined with various window configurations. Estimates of the building shell transmission loss are presented along with a summary of noise control recommendations for building roof, windows, and doors.

4:05

2pAAb3. Residential sound attenuation: Criteria and approaches. Joseph Keefe (Ostergaard Acoust. Assoc., 1460 US Hwy. 9 North, STE 209, Woodbridge, NJ 07095, jkeefe@acousticalconsultant.com)

A few case studies regarding residential building envelope sound isolation case studies will be presented. The case studies will focus on interior and exterior criteria, measured exterior sound exposure data, and prediction methods/strategies for achieving appropriate wall/window/door/roof sound attenuation values. The presentation will also discuss how to effectively communicate technical and subjective information to project stakeholders.

4:25

2pAAb4. Silencing the rolling thunder: An almost unthinkable acoustic isolation facade solution. Shane J. Kanter (Threshold Acoust., 141 W Jackson Blvd, Ste. 2080, Chicago, IL 60604, skanter@thresholdacoustics.com), Nicolaus T. Dulworth, and Carl P. Giegold (Threshold Acoust. LLC, Evanston, IL)

Situated along a popular motorcycle route, the Lindemann Performing Arts Center at Brown University posed a unique challenge to its design team — isolating the performance space from external road noise while adhering to the demanding RC 15 performance standard. This presentation provides a comprehensive exploration of the intricate development and testing of a robust yet relatively lightweight acoustic isolation façade system. The ultimate approach employed unconventional damping methods, ensuring compliance with dew point requirements. The paper delves into the nuances of the design process, with a focus on key elements such as glazing systems, roofing solutions, access doors, and gypsum buildups that push the boundaries of conventional construction, challenging the possible maximum length of a drywall fastener. Attendees will gain valuable insights into the innovative strategies implemented to achieve effective acoustic isolation without compromising the high-performance standards demanded by the Lindemann Performing Arts Center. This presentation serves as a valuable case study, offering a deeper understanding of the intersection between architecture, acoustics, and the practical challenges encountered in the design of performance spaces.

4:45

2pAAb5. Design condiserations for secondary glazing systems. Jennifer Levins (Acentech, 33 Moulton St., Cambridge, MA 02138, jlevins@acentech.com) and Benjamin E. Markham (Acentech, Cambridge, MA)

Addressing high noise levels from intermittent environmental sources, such as from train pass-bys, presents both acoustical and architectural challenges. One effective noise mitigation approach is to provide secondary storm glazing, but this is not always favored by architects, developers, and builders due to operational and aesthetic shortcomings, budget considerations, and installation challenges. These pitfalls are not always well-communicated among the design team, leading to a compromised solution. In this case study, we will discuss the concerns related to secondary glazing and the collaborative process used with the design team—informed by lessons learned on prior projects—to find a solution that addresses the various project goals.

5:05

2pAAb6. New York City noise code and open windows. David Manley (DLR Group, 6457 Frances St., Omaha, NE 68106, dmanley@dlrgroup.com) and Steph Ahrens (DLR Group, Omaha, NE)

The New York City Noise Code requires compliance with noise levels from stationary sources such as HVAC equipment to residential receivers to be measured inside the receiver location with an open window or patio door. The presentation will include the review of recent case studies completed by DLR Group with school renovations and calculated noise impact to adjacent properties. Discussion of the corrections used for window openings will be included.

Session 2pAB

Animal Bioacoustics, Computational Acoustics, Acoustical Oceanography, Underwater Acoustics, and Signal Processing in Acoustics: Data to Information, Navigating the Application of Acoustic Data for Conservation II

Megan F. McKenna, Cochair
Cooperative Institute for Research in Environmental Sciences, University of Colorado Boulder, 442 Gibson Ave., Pacific Grove, CA 93950

Carrie Wall, Cochair
University of Colorado, 325 Broadway, Boulder, CO 80305

Contributed Papers

2:45

2pAB1. Acoustic monitoring of marine environmental quality: An example from the Estuary and Gulf of St. Lawrence. Florian J. Aulanier (Fisheries and Oceans Canada, Pêches et Océans Canada, 850, Rte. de la Mer, P.O. Boîte 100, Mont-Joli, QC G5H 3Z4, Canada, florian.aulanier@dfo-mpo.gc.ca), Yvan Simard, Clément Juif (Fisheries and Oceans Canada, Mont-Joli, QC, Canada), and Samuel Giard (Pêches et Océans Canada, Mont-Joli, QC, Canada)

As a part of the Canada's Ocean Protection Plan, Fisheries and Oceans Canada has joined the efforts to better understand and monitor the effects of anthropogenic noise on marine environmental quality. Since 2017, underwater acoustic observatories were put in place across endangered whale habitats leading to the acquisition of big underwater acoustic dataset to process and analyze. In the Estuary and Gulf of St. Lawrence, underwater noise has been continuously monitored at 13 locations (6 to 10 simultaneously) between 2018 and 2023 at sampling rate up to 256 ksps in order to better understand the effect of shipping noise on marine environmental quality of the endangered St. Lawrence estuary beluga habitat. In this presentation, the data analysis pipeline from *in situ* sampling to processing is detailed, including recording schemes, data quality and control, soundscape cube, source separation, multi-scale statistics on noise levels and risk of impacts on habitat quality and visualization. These steps are used to identify, characterize and quantify daily to interannual spectral variability of underwater noise and their relationship with local environmental forcings such as shipping, wind, ice, tides, and currents at targeted locations. Ultimately, these results are used to provide support to (1) marine conservation and spatial planning initiatives from DFO and the Saguenay St. Lawrence Marine Parc; and (2) assess the predictability power of the outputs of soundscape modeling.

3:00

2pAB2. Soundscape characterization in the Central Arctic Ocean ecosystem (Svalbard) using acoustic indices. Andrea Lynn (Environ. Studies, Antioch Univ. New England, 40 Avon St., Keene, NH 03431-3516, alynn1@antioch.edu)

An increasingly warmer, less frozen Arctic is opening further to vessel traffic, transforming dominant ambient noise sources in the underwater soundscape. Chief sources may be shifting from wind to ship noise. Collecting data that help explain the changing composition of the soundscape may offer insights to guide regulation of noise in the ocean, furthering management and conservation efforts. Hydrophones connected to field recorders were used in this study to characterize the soundscape and to study marine mammal presence and ship noise. Hydrophones were deployed at 44 locations from a 13.8 m Ovni 445 sailing vessel between 9° E and 19° E and 69° N and 80° N in April 2023. Acoustic indices were utilized to assess soundscape composition. Vessel locations were confirmed using Automatic Identification System (AIS) marine traffic data. Wind, waves, and ice (geophony) dominated the soundscape's acoustic signature in remote locations, while human-caused sounds (anthrophony) were significant near Arctic shipping routes, fishing areas, and in fjords. Marine mammal vocalizations were detected near the ice edge, at fjord mouths, and in fjords. This acoustic characterization study provides a glimpse into the sonic sources and balance of sounds in the soundscape at present, essential data as the region rapidly transforms.

3:15

2pAB3. Bulk analysis of underwater sound spectra from vibratory pile driving of cylindrical steel piles in coastal waters. James Caplinger (Office of Protected Resources, National Oceanic and Atmospheric Administration (NOAA) Fisheries, 1315 East-West Hwy., Silver Spring, MD 20910, james.caplinger@NOAA.gov), Cara Hotchkin (Office of Protected Resources, National Oceanic and Atmospheric Administration (NOAA) Fisheries, Silver Spring, MD), Reny Tyson Moore (Contractor with Ocean Assoc. Inc., Office of Protected Resources, National Oceanic and Atmospheric Administration (NOAA) Fisheries, Silver Spring, MD), and Shane Guan (Div. of Environ. Sci., Bureau of Ocean Energy Management, Sterling, VA)

Vibratory pile driving is a common means of pile installation in both near-shore and offshore environments. Spectral knowledge of the noise produced by this activity is crucial in modeling and assessing the impacts to aquatic species. With the goal of providing generic representative spectra to better inform propagation modeling and related impacts, approximately 80 near-pile measurements of vibratory pile driving of cylindrical steel piles from more than 19 hydroacoustic reports were analyzed. Included pile diameters range from 24–48 inches and all measurements were of activities in U.S. coastal waters. Results include representative spectra based on the mean and median spectra binned by pile size and the presence of noise mitigation systems, an examination of the variation in the data, marine mammal hearing group weighted spectra, broadband sound pressure levels, and comparisons with general knowledge of vibratory pile driving spectra characteristics in previous literature. Finally, data availability and larger efforts aimed at a comprehensive underwater pile driving acoustic spectral database are discussed.

3:30

2pAB4. Using before-after control-impact methodology to quantify effects of full ship shock trial explosions on marine fauna. Kerri D. Seger (Appl. Ocean Sci., 2127 1/2 Stewart St., Santa Monica, CA 90404, kerri.seger.d@gmail.com), Shyam Madhusudhana (Curtin Univ., Perth, Western Australia, Australia), Holger Klinck (K. Lisa Yang Ctr. for Conservation Bioacoustics, Cornell Univ., Ithaca, NY), Kevin D. Heaney (Appl. Ocean Sci., Fairfax Station, VA), and John Boyle (Appl. Ocean Sci., Seattle, WA)

In the summer of 2021, the US Navy conducted a Full Ship Shock Trial (FSST) for the USS Gerald R. Ford. This involved three large underwater explosions off the coast of Florida, USA. We collected underwater acoustic recordings, using low-sensitivity recorders, for the Naval Undersea Warfare Center to validate their underwater acoustic propagation models. We also deployed SoundTraps on the shallow moorings to collect additional acoustical biologics data in hopes of measuring any acoustic responses of marine fauna to the explosions. The acoustic energy of the explosions did not propagate up the continental slope and were hence not captured by the SoundTraps on the shallow moorings. However, our analyses did yield some notable changes in acoustic behavior after as compared to before the explosions. Here, we describe how our field plan was designed for before-after control-impact (BACI) hypothesis testing and discuss our analyses and findings of the few significant cases. These results lend insight for improving the impact assessments and conducting behavioral response studies during a future FSST or other large underwater explosions.

3:45

2pAB5. Acoustic monitoring of harbor porpoises in Hood Canal, WA: Efforts to improve detection of a highly cryptic cetacean. Asila Ghoul Bergman (Res. and Undersea Test Ranges, U.S. Navy, NUWC Div., Keyport, Res. & Undersea Test Ranges, Code 214, Bldg 1074L, Keyport, WA 98345, asila.m.bergman.civ@us.navy.mil) and Dawn Grebner (Res. and Undersea Test Ranges, U.S. Navy, NUWC Div., Keyport, Keyport, WA)

Harbor porpoises (*Phocoena phocoena*) are the smallest, most abundant cetaceans occurring within the waters of the US Navy's Northwest Training and Testing Ranges. The ability to detect harbor porpoise and monitor their presence over time is essential to assessing potential impacts of Navy activities. This can be difficult due to their highly cryptic nature exhibited in their surface behavior (vessel avoidance) and acoustic repertoire (producing only high-frequency clicks). This presents significant challenges for both visual and acoustic (PAM) survey methods. Here, we discuss the Navy's efforts to improve harbor porpoise monitoring capability using an automated acoustic detection system. We tested multiple detection algorithms and evaluated their performance in identifying and extracting harbor porpoise clicks from acoustic recordings collected in Hood Canal, WA. Using this detection data combined with data collected from concurrent visual surveys of Hood Canal, we explored how acoustic and visual methods may be used in concert to improve harbor porpoise monitoring. Hood Canal provides a unique opportunity to do this, as it is the only known US Navy range with a resident population of harbor porpoise. These efforts will also inform future behavioral response studies needed to measure the impact of Navy sounds to individual animals.

4:00–4:15 Break

4:15

2pAB6. Automated detection of fin whale calls recorded with distributed acoustic sensing. Quentin Goestchel (School of Oceanogr., Univ. of Washington, 1503 NE Boat St., Seattle, WA 98115, qgoestch@uw.edu), William S. Wilcock (School of Oceanogr., Univ. of Washington, Seattle, WA), and Shima Abadi (Univ. of Washington, Seattle, WA)

As the global biodiversity crisis intensifies, wildlife monitoring has become a necessity. From an oceanographic perspective, the tracking of baleen whales allows studies of their habitat use and can support call density estimates, which are indicative of ocean health. Distributed Acoustic Sensing (DAS) is a promising observational technique that measures strain rates along a spare optical fiber with a spatial resolution of tens of meters to distances of $\sim$100 km. Over 4-days in November 2021, a public domain DAS dataset was collected on the submarine cables of the Ocean Observatories Initiative Regional Cabled Array off the coast of central Oregon. The experiment recorded the acoustic signals from tens of thousands of fin whale calls as well as blue whale calls, ship signals and T-phases. We will describe preliminary efforts to develop automated methods to detect and localize fin whale calls using small representative examples of the DAS data. Our longer-term goal is to extend this approach to the full data set to understand the sensitivity of DAS to fin whale calls as function of cable geometry and seafloor characteristics and to study the distribution of calling fin whales in coastal waters off central Oregon. [Work supported by ONR.]

4:30

2pAB7. Integrating ocean soundscape modelling and mapping into marine environment quality assessment and marine spatial planning. Florian J. Aulanier (Fisheries and Oceans Canada, Pêches et Océans Canada, 850, Rte. de la Mer, P.O. Boîte 100, Mont-Joli, QC G5H 3Z4, Canada, florian.aulanier@dfo-mpo.gc.ca), Patrice Lebel (Ismer, UQAR, Trois-Rivières, QC, Canada), and Yvan Simard (Fisheries and Oceans Canada, Rimouski, QC, Canada)

As on land, underwater anthropogenic noise and its potential impacts on marine ecosystems have been a growing concern in the past three decades. Initially focusing on louder noise sources, acute physical and behavioral impacts on marine mammals and commercial fish, governmental agencies started to integrate soundscapes into marine spatial planning. However soundscape science has to deal with a large number of metrics and variables (time, space, frequencies, species, types of impacts, sound sources,…) and uncertainties to be able to bring a scientifically robust and reliable support to decision making process. This is a real challenge to integrate and communicate to a vast diversity of stakeholders. To address this challenge, we present a study of the impact of shipping noise on the St Lawrence Estuary and Gulf ecosystems and in particular on endangered marine mammals. The methodology uses probabilistic underwater acoustic modelling to produce 3D-maps of acoustic-field statistics and risk of impacts at daily, weekly, monthly and annual scale over few year-cycles. Those maps are then fed into a web application capable of handling terabytes of geospatial raster's which allows to produce statistics on user-defined area interactively in order to explore and support collaborative decision making process between marine spatial planner and stakeholders. Details on probabilistic methodology and practical examples will be shown, with concluding remarks on gaps and remaining challenges.

4:45

2pAB8. Moving cargo, keeping whales: Investigating solutions for ocean noise pollution. Vanessa M. ZoBell (Scripps Inst. of Oceanogr., 812 Ocean Surf Dr., Solana Beach, CA 92075, vmzobell@ucsd.edu), John Hildebrand, and Kait Frasier (Scripps Inst. of Oceanogr., La Jolla, CA)

Human activities introduce high levels of noise into the ocean. Commercial shipping, in particular, has increased to the point that ships make a larger contribution to ocean noise than natural noise sources for most ocean locations and over a broad range of frequencies. Primeval ocean noise levels, those that would have been experienced before the advent of human-made noise in the ocean, are largely unknown. Ocean noise monitoring efforts began post-industrialization, leaving baseline sound levels under which marine organisms evolved unclear. This study modeled primeval (wind-driven) ocean noise levels and modern (ship noise plus wind noise) ocean noise levels in the Santa Barbara Channel off Southern California. The modern noise levels were validated with acoustic measurements from two sites equipped with High-frequency Acoustic Recording Packages. There was good agreement between the modern noise level models when compared to measured levels for high frequencies, and at a site shielded by islands from long range sound propagation. The lower frequency acoustic environment, modeled at 50 Hz, was more degraded than the higher frequency noise levels, modeled at 1000 Hz. This model can be used to identify target regions and times for noise reduction efforts, as well as model future scenarios for noise reduction to identify techniques with the greatest potential for conservation.

5:00

2pAB9. Masking as a means to quantify impact of increased commercial shipping on southern resident killer whales in the Salish Sea. Rianna Burnham (Dept. of Fisheries and Oceans, Inst. of Ocean Sci., Sidney, BC V8L 5T5, Canada, rianna.burnham@dfo-mpo.gc.ca), Svein Vagle, and Maximilian Lauch (Dept. of Fisheries and Oceans, Victoria, BC, Canada)

The push to report underwater soundscape measures in standardized forms allows for comparison between sites and through time. This identifies regions most impacted by anthropogenic noise sources, yet may not fully address how the characteristics of these sources affect the marine life that these areas host. In risk assessments, a move away from the use of noise thresholds encourages the use of tools such as the "maskogram" or consideration of "active" or "listening" space changes. Here, we propose the use of masking metrics, whereby the reduced efficacy of communication and echolocation signals of marine species, can be quantified as a loss of effective range. We show that this may be an effective way to consider potential impacts of noise in a more species-centric way. Using data on southern resident killer whales (SRKW, *Orcinus orca*) in waters of the coast of British Columbia, we consider how near-future scenarios in vessel presence, including increased number and size of vessels to support growing commerce and a port expansion, will alter both the soundscape and SRKW acoustics use. We will then show how masking metrics can be used to understand the impact to SRKW, and better direct management measures to lessen disturbance.

5:15

2pAB10. Using passive acoustic data to understand sustainable vessel operations. Megan F. McKenna (Cooperative Inst. for Res. in Environ. Sci., Univ. of Colorado Boulder, 442 Gibson Ave., Pacific Grove, CA 93950, megan.mckenna@noaa.gov), Lindsey Peavey Reeves (Office of National Marine Sanctuaries, National Oceanic and Atmospheric Administration, Silver Spring, MD), Timothy Rowell (Southeast Fisheries Sci. Ctr., National Marine Fisheries Service, National Oceanic and Atmospheric Administration, Beaufort, NC), Sofie Van Parijs (Southeast Fisheries Sci. Ctr., National Oceanic and Atmospheric Administration, Woods Hole, MA), Carrie Wall (Cooperative Inst. for Res. in Environ. Sci., Univ. of Colorado, Boulder, CO), and Leila Hatch (Office of National Marine Sanctuaries, National Oceanic and Atmospheric Administration, Scituate, MA)

Marine vessels support diverse ocean economic sectors. As society works towards a sustainable ocean economy, a variety of vessel initiatives are emerging. Passive acoustic monitoring (PAM) offers a method to evaluate these initiatives. Here we review efforts related to vessel noise management in marine protected areas (MPA) and benefits of vessel speed reduction programs (VSRs). Monitoring marine vessel activity and related underwater noise across a network of protected areas, like the U.S. National Marine Sanctuary system, helps managers ensure the quality of habitats used by a wide range of species. Network-wide comparisons of vessel noise revealed a spectrum of conditions, providing robust metrics to help prioritize management and inform condition assessments. These same vessel noise metrics provided insight on noise reduction related to changes in vessel operations during multiple VSRs. The metrics are complementary to other noise reduction metrics currently used to evaluate VSRs. With the growth in VSRs, there is a need to communicate noise reduction at scales relevant to the targeted vessels, typically larger than individual VSRs. Coordinated efforts are advancing to meet these needs. With strategic, systematic, and sustained efforts, PAM can continue to provide key insight on efforts to realize sustainable marine vessel operations.

Session 2pAO

Acoustical Oceanography: Acoustical Oceanography Prize Lecture

David R. Barclay, Chair
Department of Oceanography, Dalhousie University, PO Box 15000, Halifax, B3H 4R2, Canada

Chair's Introduction—1:00

Invited Paper

1:05

2pAO1. Solutions to muddy (geoacoustic inversion) problems. Julien Bonnel (Woods Hole Oceanographic Inst., 266 Woods Hole Rd., MS# 11, Woods Hole, MA 02543-1050, jbonnel@whoi.edu)

Waveguide propagation is a ubiquitous topic in ocean acoustics. This presentation focuses on low-frequency ($f < 500$ Hz) sound propagation in coastal oceans (water depth shallower than 500 m). These environments are highly dynamical and complex, and act as dispersive waveguides, bounded by the sea-surface and the seafloor. The underwater acoustic field can thus be described by a set of modes that propagate with frequency-dependent speeds. Propagation of transient acoustic signals to a distant receiver (range > 1 km) accrues multi-modal dispersion information about the propagation medium. To extract relevant information about the oceanic environment from the acoustic recording, physics-based processing methods must be used. In this presentation, I will briefly review modal propagation and time-frequency analysis. I will then show how these approaches can be combined into a non-linear signal processing method dedicated to extracting modal information from a single receiver: this information is the foundation of a transdimensional inversion method used to characterize the oceanic environment. This method will be used to perform geoacoustic inversion on the New England Mud Patch. Using several datasets collected under different oceanographic conditions, I will notably demonstrate a successful estimation of the seafloor properties, consistent with geophysical core measurements and other inversion studies, even when the water column is dynamical and mostly unknown.

Session 2pBAa

Biomedical Acoustics: General Topics in Biomedical Acoustics: Microbubbles

Brandon Helfield, Chair
Physics, Concordia University, 7141 Sherbrooke Street West, L-SP 365.04, Montreal, H4B 1R6, Canada

Contributed Papers

1:00

2pBAa1. *In vitro* comparison of subharmonic-aided pressure estimation sensitivity among microfluidic monodisperse microbubbles, sonazoid, and definity. Ga Won Kim (Radiology, Thomas Jefferson Univ., Henderson Hall (HND), 1013 NE 40th St., Seattle, WA 98105, gxk181@jefferson.edu), Lisa te Winkle, Wim van Hoeve (Solstice Pharmaceuticals, Enschede, the Netherlands), Kausik Sarkar (Mech. and Aerosp. Eng, George Washington Univ., Washington, DC), Flemming Forsberg, and John Eisenbrey (Radiology, Thomas Jefferson Univ., Philadelphia, PA)

The subharmonic response of microbubble-based ultrasound contrast agents (UCAs) typically exhibits an inverse linear relationship with the surrounding ambient pressure. Although SHAPE has been clinically successful, the sensitivity of SHAPE at lower pressures is suboptimal. While previous studies optimized SHAPE using commercially available UCAs without considering their size distribution, in this study, two SHAPE-specific monodisperse microbubble (MMB) UCAs with different bubble diameters were compared with commercial polydisperse UCAs as well as their buoyancy-separated subpopulations. UCAs were imaged in a hydrostatic tank containing 350 ml of deionized water. The pressure within the tank was increased from 0–100 mmHg. Using a modified Logiq E10 scanner (GE HealthCare) with acoustic power optimization, SHAPE data was acquired using subharmonic imaging mode with a C1-6 curvilinear probe at 2.5 MHz (receiving at 1.25 MHz). After identification of the optimal acoustic output power, MMB with a mean diameter 1.66 mm demonstrated a mean SHAPE sensitivity of −0.207 dB/mmHg ($r^2 = 0.94$). The second MMB with a mean diameter 3.45 mm demonstrated a mean SHAPE sensitivity of −0.300 dB/mmHg ($r^2 = 0.92$). In contrast, this was 2.5–4 times more sensitive than the two polydisperse UCAs (slope range of −0.081 to −0.074 dB/mmHg) and 1.45–5 times more sensitive than the buoyancy-separated UCAs (ranging −0.061 to −0.142 dB/mmHg). These results suggest that SHAPE-optimized MMB are more sensitive to measuring changes in pressure.

1:15

2pBAa2. Size and pressure dependent activity of lipid coated bubbles and finite element simulation of the propgation of focused ultrasound through bubbly media. Amin Jafarisojahrood (Phys., Toronto Metropolitan Univ., 77 harbour square, Apt. 2103, Toronto, NS M5J 2S2, Canada, amin.jafarisojahrood@ryerson.ca), Carly Pellow (Physical Sci., Sunnybrook Health Sci. Ctr., Toronto, On, Cayman Islands), Agata A. Exner (Dept. of Radiology, Case Western Reserve Univ., Cleveland, OH), David Goertz (Physical Sci., Sunnybrook Health Sci. Ctr., Toronto, ON, Canada), and Michael Kolios (Phys., Toronto Metropolitan Univ., Toronto, ON, Canada)

Use of nanodroplets as alternative to bubbles is limited to high pressure applications due to their high vaporization threshold (>1 MPa). For low pressure applications (e.g <800 kPa) and stable bubble activity, size isolated micron or sub-micron bubbles may be used to tackle the pre-focal bubble activity and attenuation. Numerical simulations of the Marmottant model were ran for bubble sizes of 0.45, 1, 2, and 4 μm in response to 1 MHz ultrasound with pressures between 10 and 700 kPa. All agents were volume matched to 20 μl/kg of Definity considering inter-bubble interactions. The pressure-dependent attenuation and the total acoustic power (TAP) were calculated for each population. Finite element simulations (FEMS) were run by taking account the pressure dependent attenuation and sound speed. Using size isolated bubbles an experimental passive cavitation case study was performed for the same exposure conditions and sizes. Numerical results show that TAP and attenuation of the bubbles are size dependent. Bigger bubbles have stronger responses at lower pressures. However, smaller agents exhibit a size-dependent pressure threshold behavior (PT) above which their attenuation and TAP grow stronger than their bigger counterparts in qualitative agreement with experiments. FEMS show that the PT of oscillations may be used to reduce pre-focal attenuation for ultrasound propagation with minimal loss.

1:30

2pBAa3. Microstreaming profile of a phospholipid-coated wall-attached microbubble undergoing shape oscillation. Hongchen Li (Erasmus MC, P.O. Box 2040, Deparment of Biomedical Eng., Rotterdam 3000 CA, Netherlands, h.li@erasmusmc.nl), Yuchen Wang, Ruisheng Su (Erasmus MC, Rotterdam, Netherlands), Christian Cierpka (Technische Universität Ilmenau, Ilmenau, Germany), Michel Versluis (Univ. of Twente, Enschede, Netherlands), Antonius F. van der Steen (Erasmus MC, Rotterdam, Netherlands), Martin D. Verweij (Imaging Phys., Delft Univ. of Technol., Delft, Netherlands), Nico de Jong, and Klazina Kooiman (Erasmus MC, Rotterdam, Netherlands)

Ultrasound-activated microbubbles induce shape oscillation and microstreaming. However, a thorough understanding remains elusive. This study investigated the 3D microstreaming profile of clinically relevant lipid-coated microbubbles undergoing shape oscillation when bound to a wall. Size-controlled biotinylated microbubbles (radius 3–8 μm) were produced by a flow-focusing device and bound to a streptavidin-coated glass. Insonification spanned 85–425 kPa at 25,000 cycles and 1.25 MHz. Two cameras, operating at 5 Mfps and 10 kfps, coupled to a microscope, captured microbubble shape oscillation and cavitation microstreaming, respectively. Astigmatic particle tracking velocimetry measured 3D particle trajectories of 500-nm beads. In the dataset (n = 79), four microstreaming profiles were observed: quadrupole, dipole, radial, and patternless. Modal decomposition revealed that quadrupole patterns resulted from self-interaction of the predominant shape mode, while dipole patterns arose from strong interactions of two nearby modes at the same oscillation frequency. Microstreaming reached 0.002–0.01 m/s at varying acoustic pressures. Quadrupole microstreaming generally induced higher shear stresses than dipole and radial patterns at identical acoustic pressure. A lower shape mode induced higher shear stresses than a higher mode at the same acoustic pressure. The unique microstreaming characteristics yield diverse outcomes in mechanical impact, which could aid efficient therapeutic applications of ultrasound-activated microbubbles.

1:45

2pBAa4. Focused ultrasound pulse repetition frequency impacts bubble activity and tracer delivery during blood-brain barrier opening. Stecia-Marie Fletcher (Radiology, Brigham and Women's Hospital/Harvard Med. School, 221 Longwood Ave., EBRC 515b, Boston, MA 02115, sfletcher4@bwh.harvard.edu), Amanda Chisholm (Radiology, Brigham and Women's Hospital/Harvard Med. School, Boston, MA), Yongzhi Zhang (Radiology, Brigham and Women's Hospital/Harvard Med. School, Boston, MA), and Nathan J. McDannold (Radiology, Brigham and Women's Hospital/Harvard Med. School, Boston, MA)

The impact of pulse repetition frequency (PRF) on microbubble activity and blood-brain barrier (BBB) opening, for a fixed total number of pulses, is unclear. We studied bubble response to a range of PRFs (0.125, 0.25, 0.5, 1, and 2 Hz) by monitoring bubble emissions during focused ultrasound (FUS) sonications of the rat brain (274.3 kHz). When a location was treated with a test PRF and a control PRF of 1 Hz, at the same pressure, the change in the harmonic amplitude compared to the control decreased with increasing PRF, with a median change of 73.8% at 0.125 Hz and −38.3% at 2 Hz. Significant changes were not observed with repeated sonications at the same PRF. Furthermore, between PRFs of 0.25 and 1 Hz, no difference was observed in the threshold for broadband emissions, which are linked to the potential for tissue damage. Fluorescence imaging was used to estimate the concentration of Trypan Blue (TB) dye following a fixed-pressure 75-pulse exposure for PRFs of 1 and 0.25 Hz in rats. A 0.25 Hz PRF led to a 68.2% increase in the mean concentration measured at the target, with a 53.9% increase in the mean harmonic sum compared with a 1 Hz PRF. Finally, a harmonic emissions-based controller at a PRF of 0.25 Hz yielded similar TB delivery, with fewer instances of petechiae observed through histology, compared to the same controller at 1 Hz. These results may be adapted to improve the clinical safety margin of FUS-mediated BBB opening and to improve the sensitivity to detecting small harmonic signals from cavitating microbubbles.

2:00

2pBAa5. An *in vitro* investigation into Lumason's utility for subharmonic-aided pressure estimation with direct comparison to Sonazoid and Definity. Hailee Mayer (Radiology, Thomas Jefferson Univ., 1411 Walnut St., Apt. 706, Philadelphia, PA 19102, hailee.mayer@jefferson.edu), Ga Won Kim, Priscilla Machado, John Eisenbrey, Trang Vu (Radiology, Thomas Jefferson Univ., Philadelphia, PA), Kirk Wallace (GE HealthCare, Niskayuna, NY), and Flemming Forsberg (Radiology, Thomas Jefferson Univ., Philadelphia, PA)

Subharmonic-aided pressure estimation (SHAPE) with ultrasound contrast agents (UCAs) has shown promising results *in vitro* and *in vivo*. Most commercial UCAs show an inverse linear relationship between subharmonic signal and hydrostatic pressure. However, conflicting trends have been reported for Lumason. Thus, this study investigated the subharmonic (i.e., SHAPE) response of Lumason and directly compare to Sonazoid and Definity in a static and a dynamic *in vitro* system using a clinical Logiq E10 scanner. SHAPE signals were acquired (in triplicate) using a C1-6 probe (transmit/receive: 2.50/1.25 MHz) in increments of 10 mmHg from 0 to 200 mmHg in the static tank, and peak pressures of 10, 20, 30, and 45 mmHg in the dynamic system. Sonazoid and Definity maintained consistent inverse linearity in both static and dynamic conditions, with average sensitivities of −0.07 dB/mmHg ($r = -0.88$) and −0.04 dB/mmHg ($r = -0.97$) in the static tank, and −0.14 dB/mmHg ($r = -0.86$) and −0.10 dB/mmHg ($r = -0.85$) in the dynamic system, respectively. Lumason exhibited a triphasic behavior; from 0–90 mmHg, SHAPE increased with increasing hydrostatic pressure (0.07 dB/mmHg; $r = 0.98$), while from 90 to 140 mmHg, the response plateaued ($r = 0.32$), before decreasing with increasing pressure from 140 to 200 mmHg (−0.14 dB/mmHg; $r = -0.95$). The subharmonic response of Sonazoid and Definity continues to be well-understood, but further investigations into Lumason's SHAPE response is needed before clinical translation.

2:15

2pBAa6. The effect of flow rate on sonoporation within individual *ex vivo* mesenteric arteries. Stephanie He (Biology, Concordia Univ., 7141 Sherbrooke St. West, Loyola Campus SP-501.06, Montreal, QC H4B 1R6, Canada, stephanie.he@mail.concordia.ca), Davindra Singh (Biology, Concordia Univ., Montreal, QC, Canada), and Brandon Helfield (Phys., Concordia Univ., Montreal, QC, Canada)

This study investigates the impact of vascular flow rate on the efficiency of sonoporation in viable mesenteric arteries ((391 ± 35) μm in intralumenal diameter), shedding light on the applications of ultrasound and microbubble-mediated (USMB) therapies. Isolated viable rat mesenteric arteries, which retained physiological functionality, exhibited reversible responses to vasoactive agents when mounted on a pressure myograph throughout the experiments. We then subjected these vessels to ultrasound (2.25 MHz) and microbubble treatment (Definity) under varied acoustic and flow conditions and assessed sonoporation through propidium iodide (PI) fluorescence microscopy. Results disclosed a strong correlation between microbubble flow rate, duty cycle, and sonoporation efficiency. Higher flow rates and duty cycles were associated with increased PI-positive cell counts, signifying more efficient cellular permeability. Post-treatment viability assays affirmed vessel integrity. These findings underscore the pivotal role of vascular flow rate in shaping therapeutic efficacy within individual vessels. The implications extend to refining USMB therapies in diverse disease scenarios, emphasizing the necessity for meticulous parameter selection to ensure both effectiveness and safety. Overall, the study furnishes valuable insights for enhancing the success and applicability of USMB-based therapeutic approaches in cardiovascular and oncological contexts.

2:30

2pBAa7. Dosage dependent subharmonic response and ambient pressure sensitivity of sonazoid microbubbles for subharmonic aided pressure estimation. Mehmet Yapar (Mech. and Aerosp. Eng, George Washington Univ., Sci. & Eng. Hall, 800 22nd St NW, Washington, DC 20052, myapar23@gwu.edu), Roozbeh H. Azami (Mech. and Aerosp. Eng, George Washington Univ., Washington, DC), Flemming Forsberg, John Eisenbrey (Dept. of Radiology, Thomas Jefferson Univ., Philadelphia, PA), and Kausik Sarkar (Mech. and Aerosp. Eng, George Washington Univ., Washington, DC)

Microbubbles are micron-sized contrast-increasing agents for diagnostic ultrasound imaging. At half of the excitation frequency due to nonlinear oscillations, they generate a subharmonic response, the magnitude of which changes by the ambient pressure. Subharmonic Aided Pressure Estimation (SHAPE) is a noninvasive method to measure local *in vivo* pressure utilizing subharmonic emission from microbubbles using medical ultrasound. In this study, we investigated the *in vitro* dosage effect of Sonazoid, a commercially available microbubble, on SHAPE. We subjected Sonazoid microbubbles to acoustic excitations ranging from 100 to 700 kPa peak negative pressure (PNP) and at a frequency of 3 MHz. We increased the ambient pressure up to 20 kPa and investigated the change in subharmonic signal during eight pressurizing-depressurizing cycles. We observed two different behaviors of subharmonic responses over investigated parameters. Subharmonic response increased significantly with increasing ambient pressure under low PNP excitations for sufficiently high concentrations. Meanwhile, increased ambient pressure at higher PNPs drastically reduced the subharmonic response for sufficiently low concentrations. Reported findings are crucial for characterizing the SHAPE performance of Sonazoid microbubbles, offering valuable insights for improved, consistent, and cost-efficient practice.

2:45–3:00 Break

3:00

2pBAa8. Extracellular matrix stiffness affects microbubble-assisted endothelial permeabilization under flow. Zoe D. Katz (Biology, Concordia Univ., 921 rue du couvent, Home, Montreal, QC h4c 2r7, Canada, katzdaniela15@gmail.com), Elahe Memari (Concordia Univ., Montreal, QC, Canada), and Brandon Helfield (Phys., Concordia Univ., Montreal, QC, Canada)

Cancer immunotherapy has faced challenges in the treatment of solid cancers due to the complex tumor microenvironment (TME), including physical barriers that prohibit immune cell infiltration. Ultrasound (US) stimulated microbubbles present a novel way to potentiate cancer immunotherapy in tumors by permeabilizing the tumor vasculature, however the effect of individual TME parameters on treatment efficacy has not yet been elucidated. Here, we focus on one biophysical parameter, showing that an increase in matrix stiffness increases US-assisted membrane permeabilization. Using a novel setup that allows for real-time visualization under flow, HUVECs seeded on polyacrylamide hydrogels of different stiffnesses (800 and 1600 Pa) showed an increase in sonoporation rates. Collagen models corroborate this trend at two different flow rates for a range of stiffnesses (5, 25, and 75 Pa): for both 5 ml/min and 30 ml/min, there is a relative increase in sonoporation from the stiffer substrate. For a given collagen substrate, there is a significant increase in sonoporation efficiency with increasing flow rate. These data can be used to fine-tune treatments according to different TMEs; further, our findings have applications in designing US parameters for targeted drug delivery and other clinical contexts that consider a variety of tissue stiffnesses.

3:15

2pBAa9. Influence of the liquid ionic strength on the resonance frequency and estimated shell parameters of lipid-coated microbubbles. Amin Jafarisojahrood (Phys., Toronto Metropolitan Univ., 77 Harbour Square, Apt. 2103, Toronto, NS M5J 2S2, Canada, amin.jafarisojahrood@ryerson.ca), Celina Yang (Phys., Toronto Metropolitan Univ., Toronto, ON, Canada), Claire Counil, Pinuta Nittayacharn (Dept. of Radiology, Case Western Reserve Univ., Cleveland, OH), David Goertz (Physical Sci., Sunnybrook Health Sci. Ctr., Toronto, ON, Canada), Agata A. Exner (Dept. of Radiology, Case Western Reserve Univ., Cleveland, OH), and Michael Kolios (Phys., Toronto Metropolitan Univ., Toronto, ON, Canada)

Measurement of the resonance frequency and shell properties of coated microbubbles (MBs) is important in understanding and optimizing their response to ultrasound. In many applications MBs are in ionic liquids; however, the influence of the medium charges on the MB behavior is not well investigated. This study aims to measure the medium charge interactions with MBs by measuring the frequency-dependent attenuation of the same size MBs in mediums of varying charge density. In-house lipid-coated MBs were size isolated to a mean size of 2.35 μm using differential centrifugation. MBs were suspended in distilled water (DW), Phosphate-Buffered Saline solution (PBS1×) and PBS10×. The frequency-dependent attenuation of the MBs solutions was measured using a system of two aligned 100% bandwidth transducers with 10 MHz center frequency. The MB shell properties were estimated by fitting the linear equation to experiments. With increasing salinity, the frequency of the peak attenuation decreased (13, 7.5 and 6.25 MHz in DW, PBS1× and PBS-10×, respectively) and attenuation peak increased (∼140%). Estimated MBs shell elasticity decreased by 64% between DW and PBS-1× and 36% between PBS-1x and PBS-10x. The estimated shell viscosity reduced by ∼40% between DW and PBS-1× and 42% between PBS-1× and PBS-10×. The significant reduction in the fitted stiffness and viscosity is possibly due to the formation of a densely charged layer around the shell and requires proper inclusion in the related MB models.

3:30

2pBAa10. Controlling the stability of monodisperse lipid-coated microbubbles by tuning their buckling pressure. Michel Versluis (Phys. of Fluids, Univ. of Twente, 7522 NB, Enschede, the Netherlands, m.versluis@utwente.nl), Benjamin van Elburg, Guillaume Lajoinie (Phys. of Fluids, Univ. of Twente, Enschede, the Netherlands), and Tim Segers (BIOS Lab-on-a-Chip group, Univ. of Twente, Enschede, the Netherlands)

Through recent years, major hurdles associated with the microfluidic production of phospholipid-coated monodisperse microbubble have been overcome which has granted them a long shelf life. This has heaved the hope of using their unique properties *in vivo* for improved contrast, controlled therapy, or noninvasive pressure measurement. However, these bubbles are also very sensitive to small changes in ambient pressure which compromises their clinical translation: upon intravenous injection, physiological pressures will cause bubble dissolution, degrading their uniformity. Here, we demonstrate the direct relation between shell buckling and bubble dissolution by acoustically measuring the buckling pressure and the response of bubbles to controlled pressure changes. We show that the concentration of PEGylated lipid, necessary for microfluidic operation, can be used to tune the buckling pressure, and thereby increase bubble stability. We show that this concentration can be changed either directly during bubble production or, more conveniently, by heating the microbubbles after production. The proposed heating step exploits the phase-change of the phospholipids within the shell to selectively expel the PEGylated lipids. Doing so, the bubble buckling pressure can be increased from 0 kPa to 27 kPa which is above physiological pressures.

3:45

2pBAa11. Ultrasound and microbubble mediated T cell modulation in peripheral blood mononuclear cells. Ana G. Baez (Biology, Concordia Univ., 2240 Ave. Madison, Apt. #30, Montreal, QC H4B2T6, Canada, anabaezc1i@gmail.com), Davindra Singh, Stephanie He (Biology, Concordia Univ., Montreal, QC, Canada), Mehri Hajiaghayi, Fatemeh Gholizadeh, Peter J. Darlington (Dept. of Health, Kinesiology & Appl. Physiol., Concordia Univ., Montreal, QC, Canada), and Brandon Helfield (Phys., Concordia Univ., Montreal, QC, Canada)

Ultrasound (US)-stimulated microbubbles are emerging as a revolutionary therapeutic technique, notably within the field of cancer immunotherapy. While this approach has been shown to modify cell permeability and to trigger local anti-tumor effects, the interactions between vibrating microbubbles and T-cells is not well understood. Here, we explore the biophysical impact of microbubbles on T-cell permeability and viability, setting the stage for potential advancements in cellular immunotherapy. Following the activation of fresh human peripheral blood mononuclear cells (PBMCs), cells were incubated with FITC-dextran (10kDa)—used here as a surrogate drug– and exposed in the presence of Definity^{TM}(1:217–cell:bubble ratio) US (1MHz, N = 1000, PRF = 5 ms for 2 minutes) over a range of peak-negative pressures (208-563kPa). Viability was assessed immediately and post-treatment concurrently with FITC macromolecule uptake using flow cytometry. T-cell population was identified using CD3 and CD4 antibodies (60% CD3 + and 20% CD4+). All conditions examined resulted > 95% viability. Results indicated a significant increase in viably permeated PBMCs with higher acoustic pressures, ranging from 0% to 30% (sham-corrected). When analyzed separately both CD4 + and CD4- T-cells exhibited similar permeability rates (0%–37.5%). The 563 kPa condition yielded the highest permeability with minimal viability loss. These findings open avenues for enhancing the efficacy of cellular immunotherapy for solid tumors.

4:00

2pBAa12. Possible physical mechanisms of the high echogenicity of lipid coated nanobubbles. Amin Jafarisojahrood (Phys., Toronto Metropolitan Univ., 77 Harbour Square, Apt. 2103, Toronto, NS M5J 2S2, Canada, amin.jafarisojahrood@ryerson.ca), Agata A. Exner (Dept. of Radiology, Case Western Reserve Univ., Cleveland, OH), Michael Kolios (Phys., Toronto Metropolitan Univ., Toronto, ON, Canada), and David Goertz (Physical Sci., Sunnybrook Res. Inst., Toronto, ON, Canada)

Lipid coated nanobubbles (NBs) have attracted a great level of interest as ultrasound (US) contrast agents due to their ability to extravagate through

leaky tumor vasculature. Their linear resonance frequency is in the range of ~50 M Hz–200 MHz, leading to confusion over their observed strong contrast in diagnostic US frequencies. By solving the Marmottant model, the dynamics of uncoated and lipid coated NBs and microbubbles (MBs) are studied over the frequency and pressure ranges (6–12 MHz, 0.1–1.2 MPa) generally used in diagnostic US. A novel bifurcation analysis in tandem with the analysis of the frequency component of the scattered pressure are conducted. Results show that despite the increased linear resonance frequency and viscous damping due to the lipid shell, buckling and rupture of the shell enhances the generation of the 2nd and 3rd harmonic resonances at pressures as low as 0.2 MPa, not observed with uncoated NBs. The generation of the harmonic resonances are concomitant with an abrupt increase in the 2nd and 3rd harmonic frequency component of the scattered pressure with their pressure threshold (PT) increasing with decreasing NBs size. For the same gas volume, and above the PT, the maximum non-destructive 2nd and 3rd harmonic powers of NBs can become higher than the 2–4 μm MBs. Similar to the lower subharmonic pressure threshold of MBs, the dynamic variation of the NBs effective surface tension due to buckling and rupture may be the potential reason behind the observed harmonic echogenicity.

TUESDAY AFTERNOON, 14 MAY 2024 ROOM 212, 1:00 P.M. TO 4:40 P.M.

Session 2pBAb

Biomedical Acoustics, Physical Acoustics, Signal Processing in Acoustics, and Structural Acoustics and Vibration: Ultrasound Beamforming and its Applications II

Jian-yu Lu, Chair
Bioengineering, The University of Toledo, 2801 West Bancroft Street, Toledo, OH 43606

Invited Papers

1:00

2pBAb1. A high-resolution ultrafast beamformer for surgical microtransducers. Jeremy Brown (Biomedical and Elec., Dalhousie, 5790 University Ave. rm. 247, Halifax, NS B3L1V7, Canada, j.brown@dal.ca)

In recent years, ultrasound imaging has become an indispensable tool for intra-surgical guidance. Most neuro and spine surgeries, however, are now trending towards minimally invasive approaches. In these surgeries, the entire surgery is performed using endoscopic instruments inserted into a small surgical pathway. Consequently, surgical imaging technologies such as ultrasound, must also adapt to be compatible with these new approaches where they are confined to narrow surgical corridors. We have recently developed a novel, high-resolution, endoscopic ultrasound system specifically for guiding these minimally invasive surgeries. The entire system including the probe, the electronics, and software has been designed and fabricated from the ground up. This imaging platform has recently been approved for a preliminary patient imaging study has already produced promising, first-of-its kind data during brain tumour resection and spinal cord surgeries. This presentation will discuss the challenges associated with developing high frequency (30 MHz) miniaturized micro-arrays and the associated electronic beamformer. The beamformer we have developed is based on parallelized FPGAs, can process data at rates higher than 10Tb/s, and generate frame-rates higher than 1kHz. The beamforming architecture incorporates a hybrid imaging mode that interleaves ultrafast imaging for Doppler, and line-by-line focusing for B-Mode. The Doppler mode is overlayed on the BMode image in real-time.

1:20

2pBAb2. Volumetric beamforming in real-time using commodity hardware. Sebastian K. Præsius (Dept. of Health Technol., Tech. Univ. of Denmark, Ørsteds Plads, Bldg. 349, Rm. 226, Kgs. Lyngby 2800, Denmark, sebka@dtu.dk), Lasse T. Jørgensen, and Jørgen A. Jensen (Dept. of Health Technol., Tech. Univ. of Denmark, Kgs. Lyngby, Denmark)

Ultrasound imaging is widely used in medicine for its safety and affordability. However, it demands large transducer arrays because the image resolution is proportional to the number of elements, *N*, typically around 128 for 2D imaging. Three-dimensional imaging requires N^2 audio channels (≈350 GB/s of data) if the equivalent 2D matrix array is used, which is practically impossible to process. To solve this issue, row-column arrays (RCAs) aggregate rows and columns of elements, reducing data rate and processing demands by a factor of N. A novel dual-stage beamforming algorithm further lowers the beamforming operations by N/2, with negligible impact on the image quality. For $N = 128$, the processing is 8192 times faster than with a matrix array, and it is hypothesized the 3D RCA beamforming can be done in real-time using a commodity graphics card. The beamforming rate of an NVIDIA RTX 4090 GPU was measured for *in-vivo* data from a rat kidney, achieving 1394 full volumes per second, which is over 150 times faster than previous implementations. Combining RCAs with the new beamforming algorithm and GPU processing thus enables volumetric beamforming to be done affordably at the bedside in real-time using a standard scanner and PC.

Contributed Papers

1:40

2pBAb3. Tunable liquid-based lenses for ultrasonic beamforming. Sina Rostami (Phys. and Astronomy, Univ. of MS, 115 Ashley Way, Oxford, MS 38655, srostami@go.olemiss.edu) and Joel Mobley (Phys. and Astronomy, Univ. of MS, University, MS)

Quasi-planar stepped lenses (QSL's) based on phasing have proven to perform closely to conventional refractive lenses for the generation of both focused and limited diffraction beams. However, since QSLs are passive and made from solid materials, each must be designed with a specific frequency and beam type in mind. In this presentation, we report on a tunable lens with a planar aperture that employs liquid channels to perform phase-based beamforming. The desired phase patterns are produced using specific speed-of-sound profiles and can be flexibly configured for a range of frequencies and beam types without modification. In this talk, we report on the results of both experimental and numerical results for this class of lenses and discuss their potential applications.

1:55

2pBAb4. Modulating ultrasound field phase and amplitude using foam gratings as an alternative to acoustic lenses. Luke A. Richards (Univ. of Oxford, Oxford, United Kingdom), Eleanor P. Stride (Univ. of Oxford, Inst. of Biomedical Eng., Oxford OX3 7DQ, United Kingdom, eleanor.stride@eng.ox.ac.uk), and Robin O. Cleveland (Univ. of Oxford, Oxford, United Kingdom)

There is an increasing number of applications for therapeutic ultrasound, including high intensity focused ultrasound surgery, neuromodulation and drug delivery. For many of these applications, accurate focusing of the ultrasound field at tissue depths of several cm is essential and thus patient-specific acoustic lenses have been widely investigated as an economical alternative to phased array transducers. Lenses can be cast or 3D printed from a range of polymers offering good acoustic impedance matching to human tissue. Disadvantages of polymer lenses, however, include their relatively large size and the difficulties involved in eliminating gas bubbles during production. In this study an alternative approach is investigated using a "grating" comprising a sheet of polymer foam, perforated with a series of pores that act as wave guides. The grating is placed in front of a transducer with the pores arranged to modulate the phase and amplitude of the sound field as required. The gratings can be cheaply and rapidly fabricated, are < 5 mm thick and can achieve phase changes of more than 180° were successfully achieved while maintaining a transmission amplitude of more than 50%.

2:10

2pBAb5. Separate emission/reception transducers for 3D ultrafast ultrasound imaging. Alexis Carrion, Ibrahima Touré, Tamara Krpic, Maxime Bilodeau, Patrice Masson (Université de Sherbrooke, Sherbrooke, QC, Canada), and Nicolas Quaegebeur (Université de Sherbrooke, Mech. Eng. Dept., 2500 blvd Universite, Sherbrooke, QC J1K 2R1, Canada, Nicolas.Quaegebeur@USherbrooke.ca)

Recent advancements in ultrafast 3D ultrasound imaging have revolutionized the field of echography, enabling its extension to novel applications such as cerebral dynamics, cardiac electrophysiology and the quantitative imaging of intrinsic mechanical properties of tumors. To facilitate these advancements, two primary transducer strategies are employed in 3D imaging. The first involves dense 2D probes equipped with a large number of elements, typically exceeding 1024. The second strategy uses row-column addressing, which simplifies the electronic control of the probe elements. Despite their effectiveness, these methods entail complexities in design and fabrication. Addressing these challenges, our study introduces an innovative transducer configuration that distinctly separates emission and reception functionalities. This separation not only simplifies the overall transducer design but also significantly reduces the system's complexity. A sparse array of PVDF transducers, which have been laser micro-machined to ensure acoustic transparency, is used at the reception. For the emission aspect, we employ a specialized acoustic concentrator to emulate point-like emission. The paper presents the detailed design requirements, assembly process, and the operational principles of this novel transducer. Furthermore, an experimental validation is conducted using a CIRS 040GSE phantom model. This validation is crucial to demonstrate the practical applicability and reliability of our transducer in real-world medical imaging scenarios.

2:25

2pBAb6. Miniaturized acoustic concentrators for ultrafast 3D ultrasound imaging. Ibrahima Touré, Maxime Bilodeau (Université de Sherbrooke, Sherbrooke, QC, Canada), and Nicolas Quaegebeur (Université de Sherbrooke, Mech. Eng. Dept., 2500 blvd Universite, Sherbrooke, QC J1K 2R1, Canada, Nicolas.Quaegebeur@USherbrooke.ca)

Ultrafast 3D ultrasound imaging is a rapidly growing research field that enables real-time imaging at a high frame rate and in a non-invasive manner. Currently, to achieve ultrafast 3D ultrasound, plane or diverging waves from virtual sources are used. These virtual sources can be considered as focal points during emission, allowing for improved transmission of ultrasonic energy. However, they have limitations in terms of transmitted energy, complex electronics, and timing issues, which impact the quality of the resulting images. To address this, a study was conducted to design and experiment with new miniaturized ultrasound transmitters based on cylindrical waveguides with tapered sections. These transmitters, with a diameter of approximately 100 μm, can replace the virtual sources. A numerical study based on 2D axisymmetric Finite Element Modeling (FEM) is first performed to model the transmission and reflection of longitudinal ultrasonic waves through various cylindrical rods. The results demonstrated that the transmission depends on the diameter ratio, frequency and longitudinal mode order (L0, L1, L2, etc.). An experimental study is then conducted on a stainless steel waveguide instrumented with 6mm piezo discs. The mechanical energy, estimated using Laser Doppler Vibrometer is transmitted through a 150 μm diameter rod, allowing mechananical focusing in the frequency range between 0.5 and 5 MHz. The application of this method to separate emission/reception transducers is then demonstrated for 3D ultrafast imaging.

2:40–2:55 Break

2:55

2pBAb7. Transmit beam design for tissue harmonic volume imaging of muscular tissues with a row-column array. Maryam Satarpour (Dept. of Bioengineering, Univ. of Pittsburgh, Pittsburgh, PA), Zhiyu Sheng (Dept. of Medicine, Univ. of Pittsburgh, Pittsburgh, PA), John M. Cormack (Dept. of Medicine, Univ. of Pittsburgh, Div. of Cardiology, Dept. of Medicine, University of Pittsburgh, Pittsburgh, PA, jmc345@pitt.edu), and Kang Kim (Dept. of Bioengineering, Univ. of Pittsburgh, Pittsburgh, PA)

Row-Column-addressed Arrays (RCAs), which have recently become commercially available, transmit along the rows and receive along columns of a 2D array in order to produce 3D volume images with low channel count. Imaging of the inherently three-dimensional, layered, and fibrous structure of muscular tissues in 3D using an RCA can improve evaluations of muscle function and disease compared to conventional 2D cross-sectional imaging, but volume images obtained using an RCA suffer from low lateral spatial resolution, low contrast, and grating lobe artifacts. Tissue harmonic imaging (THI) employs nonlinear acoustic effects to image at twice the transmit frequency, and can yield increased spatial resolution, increased contrast, and decreased artifacts from grating lobes. Here we investigate transmission considerations for THI in muscular tissues using a commercially available RCA (Vermon RC6gV). The parameter space is explored efficiently with frequency domain simulations of the 3D Westervelt equation. Pulse inversion imaging is applied experimentally using both steered plane wave and focused transmits to evaluate THI with an RCA in tissue phantoms and in muscular tissue. Pulse inversion volume images are compared to conventional images in terms of signal-to-noise, contrast, and ability to identify layers and fibrous structures in muscle in 3D. [Work supported by NIH HEAL Initiative R61AT012282.]

3:10

2pBAb8. Transducer module apodization for reducing bone heating during focused ultrasound uterine fibroid ablation. Sobhan Goudarzi (Physical Sci. Platform, Sunnybrook Res. Inst., 2075 Bayview Ave., Toronto, ON M4N 3M5, Canada, sobhan.goudarzi@sri.utoronto.ca), Ryan M. Jones, Yin H. W. Lee, and Kullervo Hynynen (Physical Sci. Platform, Sunnybrook Res. Inst., Toronto, ON, Canada)

During MR-guided focused ultrasound (MRgFUS) for uterine fibroids, thermoablation of tissue near spine/hips is challenging due to bone heating that can cause patient pain and potentially damage nerves. Here we investigate transducer module apodization for maximizing the focal-to-bone heating ratio ($\Delta T_{;\mathrm{ratio}}$) *in silico* using a 6144-element flat fully populated phased array operating at 0.5 MHz (Arrayus Technologies, Inc.). Acoustic and thermal simulations were performed using anatomies of ten patients who underwent MRgFUS ablation for uterine fibroids with this device as part of a clinical trial (NCT03323905). Transducer modules (64 elements/module) whose beams intersected no-pass regions were identified, their amplitudes were reduced by varying blocking percentage levels, and the resulting temperature field distributions were evaluated across multiple sonications per patient. For all simulated sonications transducer module blocking improved $\Delta T_{;\mathrm{ratio}}$ compared to no blocking. In 42% of sonications, full module blocking maximized $\Delta T_{;\mathrm{ratio}}$, with mean improvements of $97\% \pm 55\%$ and $47\% \pm 36\%$ in hip and spine compared to no blocking, at the cost of increased focal thermal volumes and acoustic power levels. In the remaining sonications, partial module blocking provided increased $\Delta T_{;\mathrm{ratio}}$ values ($39\% \pm 45\%$ in hip, $19\% \pm 15\%$ in spine targets) relative to full blocking. The optimal blocking percentage varied depending on the specific treatment geometry.

3:25

2pBAb9. Investigation of HIFU beam propagation through acoustic holograms in water. Timofey Krit (Inst. for Diagnostic and Interventional Radiology, Univ. Hospital of Cologne, Kerpener Str. 62, Cologne 50937, Germany, Timofey.Krit@uk-koeln.de), Luisa Brecht, Nina Reinhardt, Johannes Lindemeyer, and Holger Grüll (Inst. for Diagnostic and Interventional Radiology, Univ. Hospital of Cologne, Cologne, Germany)

We investigated variations in the HIFU beam of a clinical system as it traverses acoustic holograms—small artificial objects of different types—within a tank that was filled in with the degassed deionized water. The pressure distribution within the HIFU beam was assessed in multiple planes perpendicular to the beam axis. The ultrasound transducer with the focal length of 14 cm inside the Sonalleve table top (Profound Medical, Mississauga, ON, Canada) generated the focusing ultrasonic beam, which propagated through the water. Each hologram was positioned 3 cm below the focal plane, with precise adjustments of positions and angles for each measurement. Pressure measurements were conducted using the needle hydrophone HNA-0400 (ONDA Corp, Sunnyvale, CA, USA) mounted on a 3D positioning system. The motors of the positioning system were controlled, and pressure values were recorded using a GUI generated in LabVIEW. The displacement step created by the motors on each axis was 0.01 mm. The properties of the holograms were reconstructed, considering the experimental data obtained. Our findings demonstrated that the shape, size, elasticity, and placement of each acoustic hologram significantly influenced the observed field pattern. This work was supported by the German Federal Ministry of Education and Research ("MR-HIFU-Pancreas", FKZ:13GW0364D).

3:40

2pBAb10. Experimental and numerical comparison of multiple passive beamformers for separating intra- and extra-canal cavitation activity during transvertebral spinal cord therapy. Andrew P. Frizado (Sunnybrook Res. Inst., 2075 Bayview Ave., Toronto, ON M4N 3M5, Canada, andrew.frizado@mail.utoronto.ca) and Meaghan O'Reilly (Sunnybrook Res. Inst., Toronto, ON, Canada)

Distinguishing intra-canal cavitation activity during transvertebral focused ultrasound sonication of the spinal cord is a challenge due to the strong prefocal cavitation emissions in the spinalis musculature overwhelming emissions originating in the canal. To achieve mapping of all cavitation sources simultaneously, two methods for enhancing detection sensitivity are investigated: (1) multiple dynamic ranges within a reconstructed volume and (2) utilizing alternative beamformers to delay-and-sum (DAS) during map reconstruction. The performance of DAS beamforming is compared to a delay-multiply and-sum beamformer (DMAS), with and without a paired multiplicative compounding method (pDMAS). Experiments and simulations were performed on a 128-element, dual-aperture transvertebral array, through stacks of *ex vivo* human vertebra. Numerically and experimentally obtained point spread functions were compared in 3D, producing voxel-wise cross correlation values of 0.84, 0.89, 0.97 ($N = 1$) for the beamformers listed above, respectively, in water. Experimental, transvertebral localizations of canal sources in isolation produced localization error of 2.8 ± 1.2, 2.9 ± 1.4, 2.7 ± 1.3 mm for a single vertebral target ($N = 30$ sonications), respectively. A large numerical data set investigating the prefocal cavitation problem ($N > 160$) is presented and compared with ($N = 9$) experimental data demonstrating enhancement of intracanal sensitivity in cavitation maps.

3:55

2pBAb11. Quantitative ultrasound in synthetic transmission aperture ultrasound imaging. Yuan Xu (Toronto Metropolitan Univ., 350 Victoria St., Dept. of Phys., Toronto, ON M5B2K3, Canada, yxu@ryerson.ca), Na Zhao, Khalid Abdalla, and Keyan Sheppard (Toronto Metropolitan Univ., Toronto, ON, Canada)

Conventional ultrasound imaging mainly provides qualitative information, and the imaging results depend strongly on the operator of the scanners. Quantitative ultrasound (QUS) imaging can provide specific numbers related to tissue properties that can increase the specificity of image findings, leading to improvements in diagnostic ultrasound. Most of the QUS systems use B-mode imaging. B-mode QUS usually has poor spatial resolution. Some of them need a reference phantom, which makes the clinical implementation slow. In addition, different QUS methods use different data acquisition protocols or imaging modes. So far, no *in vivo* system can extract the attenuation coefficient, speed of sound, and microstructure information of biological tissues at a good spatial resolution from one data set in a clinically applicable device. This paper presents our study on developing QUS in STA (Synthetic Transmit Aperture Ultrasound Imaging). STA is shown to have richer information in the raw data than B-mode imaging. Therefore, it has the potential to improve spatial resolution and accuracy and provide richer feature information in QUS. The quantitative information includes, but is not limited to, attenuation coefficient, speed of sound, and tissue microstructure information from one frame of image data in a clinically applicable device.

4:10

2pBAb12. Low-intensity pulsed ultrasound (LIPUS) mediated on-demand release of cannabidiol from the surface of gold nanoparticles. Anshuman Jakhmola (Toronto Metropolitan Univ., Toronto, ON, Canada), Farshad Moradi Kashkooli (Toronto Metropolitan Univ., 350 Victoria St., Toronto, ON, Toronto, ON M5B 2K3, Canada, fmoradik@torontomu.ca), Tyler Hornsby, Michael Kolios (Toronto Metropolitan Univ., Toronto, ON, Canada), Kevin Rod (Toronto Polyclinic, Toronto, ON, Canada), and Jahangir (Jahan) Tavakkoli (Toronto Metropolitan Univ., Toronto, ON, Canada)

Emerging evidence has suggested that natural drugs like Cannabidiol (CBD) have the potential to selectively destroy cancer cells in the body. In this study, we designed gold nanoparticles (GNPs) loaded with CBD molecules and employed a patented custom-designed handheld 1-MHz low-intensity pulsed ultrasound (LIPUS) device to selectively release CBD from the surface of gold nanoparticles in an *ex vivo* tissue model. A $60 \times 73 \times 25\,\mathrm{mm}^3$*ex vivo* tissue sample was placed in a 3D-printed plastic sample holder, while0.5 ml solution of CBD-loaded GNPs was pipetted into a separate 3D-printed GNP holder. The 3D-printed GNP holder was then inserted into the tissue center, and the combined holders were submerged in a 37 °C acrylic water tank with a water heater for temperature control. The LIPUS device was held at the top of a water tank which directed ultrasound downward towards the region of interest in tissue. Initial results displayed the highest fluorescence intensity at a LIPUS exposure of 4.1 W acoustic power, a 50% duty cycle, and a total exposure time of 5 min. The release of CBD was measured by a dye-based fluorescence assay, and it was found to be significantly higher when applying LIPUS compared to a bulk heating-only treatment using the same temperature profile. It is concluded that both thermal and non-thermal effects caused by LIPUS has a significant impact on CBD release from gold nanoparticles with 42% and 58% for thermal and non-thermal effects of LIPUS, respectively.

4:25

2pBAb13. Real time cavitation monitoring during ultrasound-mediated drug delivery in normothermically perfused human tumour-bearing livers. Michael Gray (Inst. of Biomedical Eng., Univ. of Oxford, Inst. of Biomedical Eng., University of Oxford, Oxford OX3 7DQ, United Kingdom, michael.gray@eng.ox.ac.uk), Brian Lyons, David Johnson (Inst. of Biomedical Eng., Univ. of Oxford, Oxford, United Kingdom), Brannon Nicholls (Medical Sci. Doctoral Training Ctr., John Radcliffe Hospital, Oxford, United Kingdom), Alex Gordon-Weeks (Nuffield Dept. of Surgical Sci., John Radcliffe Hospital, Oxford, United Kingdom), Robert Carlisle, and Constantin Coussios (Inst. of Biomedical Eng., Univ. of Oxford, Headington, Oxford, Oxfordshire, United Kingdom)

In the fight against a broad spectrum of human diseases, cavitation techniques show great promise for overcoming physical barriers that lead to suboptimal uptake of passively administered therapeutics. However, small animal testing of candidate therapies remains a poor predictor of clinical success. Here we demonstrate ultrasound-mediated drug delivery in normal and tumour-bearing human livers infused with protein-based cavitation nuclei (PCaN). Whole and partial human livers were obtained immediately from hepatectomy surgeries and were normothermically sustained using a clinically approved perfusion system (OrganOx Metra). Ultrasound was applied using a 0.5 MHz focused source (Sonic Concepts H107) and was monitored with a calibrated linear array (ATS L7-4) for real time structural and cavitational imaging implemented on an array controller (Verasonics Vantage 256). Specifically, the therapy process was monitored using passive acoustic mapping (PAM) of broadband cavitation emissions, employing a non-adaptive beamformer that deconvolves the array point spread function. Levels of fluorescently labelled drugs incorporated in and co-administered with the PCaN were quantified in blood and tissue samples collected during and following treatment, respectively. This presentation highlights PAM observations of broadband cavitation persistence and drug delivery in untargeted and targeted tissues, including the first ever experiments in tumour-bearing human livers.

Session 2pCA

Computational Acoustics, Animal Bioacoustics, Signal Processing in Acoustics, Biomedical Acoustics, and Architectural Acoustics: Application of Model Reduction Across Acoustics

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

Benjamin M. Goldsberry, Cochair
Applied Research Laboratories at The University of Texas at Austin, 10000 Burnet Road, Austin, TX 78758

Chair's Introduction—1:00

Invited Papers

1:05

2pCA1. Modal reduction for exterior acoustic and vibroacoustic problems. Steffen Marburg (School of Eng. and Design, Tech. Univ. of Munich, Boltzmannstr. 15, Garching 85748, Germany, steffen.marburg@tum.de)

Modal analysis, mode superposition and modal reduction are considered standard approaches for interior acoustic problems as well as for pure vibration problems. Things are different for exterior problems. Only a few methods are known to formulate a linearized eigenvalue problem for unbounded acoustic problems. Even fewer techniques on modal superposition and reduction are found in the literature. This talk will review such techniques which are either based on conjugate Astley-Leis infinite elements or on boundary elements for the unbounded region. While the results appear to be promising, a number of open questions need to be solved in this context. Among others, it is yet unclear which modes are actually required for a modal reduction and how to find a method to just evaluate these modes and the related eigenvalues. The situation is different for vibroacoustic radiation problems with clear resonances. In such cases, a modal reduction is able to substantially accelerate the solution process. Future work might lead to modal reduction using modes from non-linear eigenvalue problems which are known from the numerous papers on the boundary element methods published over the last decade.

1:30

2pCA2. Latent space representation method for structural acoustic assessments. Gregory A. Banyay (Appl. Res. Lab., Penn State Univ., State College, PA) and Andrew S. Wixom (Appl. Res. Lab., Penn State Univ., P.O Box 30, M.S. 3220B, State College, PA 16801, axw274@psu.edu)

When targeting structural acoustic objectives, engineering practitioners face epistemic uncertainties in the selection of optimal geometries and material distributions, particularly during early stages of the design process. Models built for simulating acoustic phenomena generally produce vector-valued output quantities of interest, such as autospectral density and frequency response functions. Given finite compute resources and time we seek computationally parsimonious ways to distill meaningful design information into actionable results from a limited set of model runs, and thus aim to use machine learning to perform model order reduction. Unlike time series data for which recurrent neural networks can learn from prior time steps to inform subsequent steps, frequency-dependent data demands a different machine learning paradigm. We thus evaluate the utility of autoencoders to represent structural acoustic results with a low dimensional latent space to enable such objectives as surrogate modeling for design optimization. We demonstrate the accuracy of autoencoder based methods of constructing a manifold representation for frequency dependent functions of varying modal density and damping, and discuss the predictive capability thereof.

1:55

2pCA3. Blind source extraction using Kronecker product structured mixing model. Zbyněk Koldovský (ITE, Tech. Univ. of Liberec, Studentska 2, Liberec 46117, Czechia, zbynek.koldovsky@tul.cz), Jaroslav Čmejla, and Aamir Farooq (ITE, Tech. Univ. of Liberec, Liberec, Czechia)

Blind Source Extraction (BSE) aims at extracting a source-of-interest (SOI) from a mixture of signals observed through multiple sensors. In this work, we consider a parameter-reduced mixing model suitable for sensor arrays structured in a grid. The parameter vector involving weights with which the sensors receive the SOI is factorized into the Kronecker product of two sub-vectors. In addition to the description of the new model, we present an algorithm designed to identify the mixing parameters based on the assumption of independence of the SOI from the other signals in the mixture. In simulations, the performance of the algorithm is compared with that of the fully parameterized counterpart and is compared with the corresponding Cramer-Rao-induced bounds on the achievable Interference-to-Signal Ratio.

2:20

2pCA4. A hybrid formulation for determining the input power in a stiffened plate over a wide frequency range. Nickolas Vlahopoulos (Univ. of Michigan, 2600 Draper Dr., Ann Arbor, MI 48108, nickvl@umich.edu), Sungmin Lee, and Jonmarcos Diaz (Univ. of Michigan, Ann Arbor, MI)

A hybrid analytical/numerical formulation is presented for determining the input power in a stiffened plate over a wide frequency range. A Component Mode Synthesis (CMS) approach is used for developing the hybrid formulation. The structure is divided into a local substructure in the vicinity of the excitation and non-local substructures representing the remaining system. It is considered that the excitation is applied at a small number of discrete locations, therefore the local substructure is a small portion of the overall system. The computational efficiency originates from using an analytical approach and periodic structure theory for determining the dynamic modes and natural frequencies for the non-local substructures. Finite elements are used for determining the static modes for the non-local substructures and also for both the dynamic and static modes of the local substructure. The CMS matrices for the non-local substructures are condensed to their common interface degrees of freedom with the local substructures. In this manner, the non-local substructures are eventually represented as boundary conditions on the local substructure. Results from the hybrid formulation are compared with results from dense finite element models in order to demonstrate the validity and the efficiency of the hybrid method.

2:45

2pCA5. Modeling high-frequency backscattering from an impedance surface using Kirchhoff approximation. Pei-Tai Chen (Ctr. of Oceanic Eng., National Taiwan Ocean Univ., 2 Pei-Ning Rd., Keelung, Taiwan 886, Taiwan, ptchenline@gmail.com)

The Kirchhoff approximation is based on planar wave impinging on a planar surface, which is approximated for high-frequency wave scattering from a curved object. The Kirchhoff approximation is used to assess backscattering from a curved, rigid surface when exposed to high-frequency incident waves. In this presentation, we expand the rigid surface Kirchhoff approximation to accommodate impedance surfaces. The derivation is performed on a planar wave's incident onto an impedance flat surface, resulting in a reflection coefficient, which in turn determines both surface pressure and surface normal velocity. When these two surface quantities are integrated into a Helmholtz integral formula, it allows for the evaluation of backscattering acoustic field pressures. The BeTTSi model, featuring an impedance surface, serves as an illustration for comparing the current Kirchhoff approximation with the solution obtained by a Fast Multipole Expansion Boundary Element Method (FMBEM). The latter is a numerical solution of the corresponding Helmholtz surface integral equation. The numerical discretization involves nearly a million points, representing the degrees of freedom necessary for modeling the BeTTSi under high-frequency incident waves. The comparison involves these two numerical evaluations across various incident wave directions, encompassing scenarios with single reflection and multiple reflections due to surface geometry.

3:10–3:25 Break

Contributed Papers

3:25

2pCA6. Information geometry approach to model reduction. Jay C. Spendlove (Dept. of Phys. and Astronomy, Brigham Young Univ., C341 ESC, Brigham Young University, Provo, UT 84602, jayclark24@gmail.com), Mark K. Transtrum, and Tracianne B. Neilsen (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Information geometry combines the fields of information theory and differential geometry. In information geometry, a model is interpreted as a Riemannian manifold, referred to as the model manifold, where distance on the model manifold corresponds to statistical distinguishability. Boundaries on the model manifold are reduced-order models where a parameter or parameter combination is taken to an extreme limit and drops out of the model. Therefore, by locating the model manifold boundaries, simpler models with fewer parameters can be obtained. A model reduction algorithm which exploits this insight is the Manifold Boundary Approximation Method, or MBAM, which locates manifold boundaries by calculating geodesics on the surface of the model manifold. This can be done iteratively to find increasingly simple reduced models. We demonstrate this method for model reduction using the Pekeris waveguide transmission loss model. A machine learned surrogate model for the Pekeris waveguide is constructed so that the necessary derivatives for calculating geodesics can be easily obtained using automatic differentiation. [Work supported by the Office of Naval research Grant N00014-21-S-B001.]

3:40

2pCA7. Comparison of substructuring methods with interface reduction. Matthew Luu (Penn State, 446 Bluecourse Dr (Apt907), State College, PA 16803, mbl5743@psu.edu), Jon Young (Appl. Res. Lab, State College, PA), and Andrew S. Wixom (Appl. Res. Lab., Penn State Univ., State College, PA)

For complex dynamical systems, substructuring techniques can be applied to simulate the large system more efficiently by handling it as a collection of smaller components. In order to couple these components, the description of the interface between components is very important. Classic substructuring techniques, such as the Craig-Bampton method, reduce the interior modes of the system, but don't reduce the interface description. This can cause unnecessary degrees of freedom kept in the final assembled model. While there have been a few substructuring methods published that do include some amount of interface reduction, they are much more rare and even more rarely applied in practice. Therefore, the goal of this work is to survey these techniques and compare them with a new, line-based orthogonal polynomial interface reduction technique. These methods will be applied to a set of example problems in order to determine their validity and convergence compared to the full model.

3:55

2pCA8. Accurate and computationally efficient basis function generation using physics informed neural networks. Nathan Cloud (Appl. Res. Labs. at The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX, nathan.cloud@arlut.utexas.edu), Benjamin M. Goldsberry, and Michael R. Haberman (Appl. Res. Labs. at The Univ. of Texas at Austin, Austin, TX)

Basis functions that can accurately represent simulated or measured acoustic pressure fields with a small number of degrees of freedom is of great use across various applications, including finite element methods, model order reduction, and compressive sensing. In a previous work [B. M. Goldsberry, J. Acoust. Soc. Am. 153, A193 (2023)], basis functions were derived for an element in a given mesh using a combination of interpolation functions defined on the boundaries of the element and the Helmholtz-Kirchhoff (HK) integral. This forms a new interpolatory basis set that efficiently and accurately represents the interior of the element. However, the previous analysis was limited to a two-dimensional rectangular element. In this work, physics informed neural networks (PINN) are investigated as a means to generate HK basis functions for general element shapes. PINNs have been previously shown to accurately learn solutions to parameterized partial differential equations. The element geometry parameterization and the boundary interpolation functions are given as inputs to the PINN, and the output of the PINN consists of the physically accurate basis functions within the element. Details on the implementation and training requirements on the PINN to achieve a desired accuracy will be discussed. [Work supported by ONR.]

4:10

2pCA9. An overview of symmetry-based techniques used for sound power application with illustrative case studies. Ian C. Bacon (Phys. & Astronomy, Brigham Young Univ., 333 W 100 S, Provo, UT 84601, icbacon@byu.edu), Micah Shepherd, and Scott D. Sommerfeldt (Phys. & Astronomy, Brigham Young Univ., Provo, UT)

This work investigates the growing application of vibration-based sound power (VBSP) measurements in structural acoustics. The VBSP method involves multiplying a measured velocity vector by a geometry-dependent radiation resistance matrix to compute the frequency-dependent sound power. However, computational challenges arise during matrix construction, even for simple geometries. To address this, inherent symmetries in the radiation resistance matrix for both baffled and unbaffled geometries are exploited. Multi-layered Toeplitz symmetries emerge for baffled conditions, while centrosymmetric symmetries are manifest for unbaffled conditions. These symmetries enable an efficient compression and subsequent reconstruction of the resistance matrix. Simple case studies demonstrating the temporal costs for flat, curved, and arbitrarily curved plates in unbaffled conditions will be presented. [Funding for this work was partially provided by the National Science Foundation (NSF).]

TUESDAY AFTERNOON, 14 MAY 2024 ROOM 204, 1:00 P.M. TO 4:05 P.M.

Session 2pEA

Engineering Acoustics, Biomedical Acoustics and Physical Acoustics: Things That Go Boom: High Amplitude Acoustic Sources

Thomas E. Blanford, Chair
University of New Hampshire, University of New Hampshire, Durham, NH 03824

Invited Papers

1:00

2pEA1. History of high-amplitude underwater sound source research at the University of Texas Applied Research Laboratories. Preston S. Wilson (Walker Dept. Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, 204 East Dean Keeton St., Mail Stop: C2200, Austin, TX 78712-0292, pswilson@mail.utexas.edu), Andrew R. McNeese, Kevin M. Lee, and Robert L. Rogers (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

The Applied Research Laboratories has an extensive history of supporting underwater acoustic research through the development of novel high-amplitude underwater sound sources, such as the Plasma Sound Source (PSS), the Combustive Sound Source (CSS) and the Rupture Induced Underwater Sound Source (RIUSS). These sources generate broadband acoustic pulses capable of long-range propagation and seabed penetration, and can be viewed as alternatives to explosive sources. The PSS is based on the discharge of electrical energy stored in capacitors which results in a plasma bubble. The CSS consists of a submersible combustion chamber, open to the water, which is filled with a combustive mixture that is ignited via spark. Upon ignition, the combustive mixture is converted into high temperature combustion byproducts which expand and ultimately collapse, thereby radiating an acoustic pulse. In this talk, the motivation for alternative impulsive sources is discussed and the PSS is briefly discussed. Next, the early development of CSS, through the end of the 1990's, is described, which led to the basic understanding of the CSS and the parameters required to modify its output. In the following talk further development of CSS, and the invention of RIUSS, are discussed. [Work supported by ONR and NAVO.]

1:20

2pEA2. Further development of high-amplitude underwater sound sources at The University of Texas Applied Research Laboratories. Andrew R. McNeese (Appl. Res. Labs., The Univ. of Texas at Austin, 10,000 Burnet Rd., Austin, TX 78758, mcneese@arlut.utexas.edu), Preston S. Wilson (Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Kevin M. Lee (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Following the early research on the Combustive Sound Source (CSS), described in the previous talk, additional development occurred in the 2000's and beyond, which included work on both the CSS and a new source, the Rupture Induced Underwater Sound Source (RIUSS). Development of the CSS in this era included maximizing the output of the source, investigating the use of CSS in arrays, and towing the source. The CSS was also deployed in surveys conducted during Shallow Water '06 and the Seabed Characterization Experiment. In part due to lessons learned during those surveys, the RIUSS was conceived and developed. The RIUSS functions by placing a rupture disk over an evacuated chamber and mechanically breaking the disk (either by striking on demand or via hydrostatic pressure) at a specified depth to initiate a volume collapse that produces an impulsive acoustic waveform. The development and use of each sound source will be provided along with high-speed underwater video, examples of signatures, and examples of the efficacy of the sources in ocean acoustics research. [Work Supported by ONR and NAVO].

1:40

2pEA3. Controls on the acoustic expression of buried and surface chemical explosions. Daniel Bowman (Sandia National Labs., Albuquerque, NM), Amrit K. Bal (Sandia National Labs., PO Box 969, M.S. 1377, Livermore, CA 94551-0969, akbal@sandia.gov), and Fransiska Dannemann Dugick (Sandia National Labs., Albuquerque, NM)

Surface and buried chemical explosions in the 1-50 ton TNT equivalent range can generate acoustic waves capable of traveling thousands of kilometers. Planned explosive campaigns in this size range have been used to develop acoustics-based yield determination techniques, investigate how acoustic propagation paths vary over short time scales, and examine how ground motion induces sound above the epicenter, among others. Here, we describe several test campaigns involving buried and surface chemical explosions. We discuss the combinations of explosive yield and burial depth that produce laterally propagating infrasound waves, as well as instances where diffuse sound from violent topographic shaking is observed. We also give an overview of a test series specifically aimed at investigating acoustic signal variation at tens to hundreds of kilometers range using co-located surface explosions fired tens of seconds apart. This will provide context for acoustic data interpretation from existing test series and aid in the design of future ones. [SNL is managed and operated by NTESS under DOE NNSA Contract No. DE-NA0003525.]

2:00

2pEA4. Energy density source levels of underwater explosive charges. Ross Chapman (Univ. of Victoria, University of Victoria, 3800 Finnerty Rd., Victoria, BC V8P5C2, Canada, chapman@uvic.ca)

This paper presents experimental measurements of energy density source levels of small explosive charges for high frequencies to 18 kHz. The experiments were carried out using Signals Underwater Sound (SUS) charges deployed at a deep water site in a coastal fiord north of Vancouver BC. The waveform of an underwater explosion is characterized by a high intensity shock wave signal followed by a series of bubble pulses of decreasing intensity. The bubble pulses dominate the SUS energy density spectrum at frequencies below 1 kHz, but at higher frequencies the spectrum is dominated by the characteristics of the decay envelope of the shock wave signal. The explosive material in SUS charges is packed in a thin cylindrical cavity, resulting in a non-uniform decay envelope with secondary pulses when detonated, and a non-spherical distribution of the radiated sound. The impact of the cylindrical charge shape on energy density source level was investigated experimentally. Data were recorded for SUS deployments that generated in-line and broadside sound propagation to the receiver with respect to the axis of the cylindrical charge. Experimental measurements indicate significant differences in the high frequency SUS energy density spectrum depending upon the SUS and receiver geometry.

2:20

2pEA5. Acoustic source characterization for chemical explosions in air. Keehoon Kim (Geophysical Monitoring Program, Lawrence Livermore National Lab., 7000 East Ave., L-103, Livermore, CA 94550, kim84@llnl.gov)

Chemical explosions in the air generate large pressure disturbances. These pressure waves propagate as non-linear shock waves near the source and transition into acoustic waves in the far-field. Since low-frequency acoustic waves propagate long distances without significant loss of energy, acoustic signals induced by explosions are often used to determine explosion energies of the events in terms of an explosion yield. In order to estimate the explosion energy accurately, the relationship between acoustic energy and explosion yield must be understood. However, explosion yields are often measured by non-linear shock waves in the near-field, and it is not clear how much acoustic energy accounts for the explosion energy. In addition, acoustic signals typically have lower-frequency contents than shock waves in the near-field, and hence frequency-dependent explosion energy should be understood to accurately infer explosion yields based on acoustic observations. In this study, we investigate the relationship between acoustic energy and explosion yield based on ground-truth explosion data. A standard acoustic source waveform will be determined by acoustic observations, and frequency-dependent energy will be explored for yield estimation. We will demonstrate that this frequency-dependent acoustic source characterization can improve the accuracy and confidence of explosion yield estimation.

2:40–2:55 Break

2:55

2pEA6. Modeling and development of a narrowband gas-combustion infrasound source. Chad M. Smith (Appl. Res. Lab., The Penn State Univ., State College, PA 16804, chad.smith@psu.edu) and Thomas B. Gabrielson (Penn State Univ., State College, PA)

An invaluable tool in characterization of any receiver, propagation path, or detection system, is a source with known and repeatable signal characteristics. This talk will discuss development of a well-controlled narrowband infrasound source with frequency capabilities over the 0.1 to 10.0 Hz band. Design of a transportable sound source within this band is a difficult engineering challenge. The simple source equation, which will govern any portable infrasound source due to the long wavelengths, shows this fundamental difficulty. As frequency decreases, volume displacement must increase by the squared inverse factor of frequency in order to maintain an equal pressure at equal range. To combat this, the authors have developed a source system that uses gas combustion to displace large volumes in the open atmosphere using the gaseous and highly heated combustion constituents of propane and atmospheric oxygen. Measurements have verified the capability of generating narrowband signals with reasonable signal-to-noise ratio (SNR) over the full band at close range. Signals at 1 Hz have been measured with reasonable SNR to ranges greater than 2 km. Development of the infrasound source prototype, experimental measurements, and modeling of the acoustic pressure output will be discussed.

3:15

2pEA7. Echo sounding with the shock wave generated by the implosion of an instrument pressure housing. Scott Loranger (Ocean Sci., Kongsberg Discovery, 86 Water St., Woods Hole, MA 02543, sloranger@whoi.edu), David R. Barclay (Oceanogr., Dalhousie Univ., Halifax, NS, Canada), and Michael Buckingham (Scripps Inst. of Oceanogr., San Diego, CA)

In December 2014, what was intended to be a passive-acoustic experiment at the Challenger Deep, suddenly became an active-acoustic experiment. Two free-falling, passive-acoustic instrument platforms, each with a suite of hydrophones, a CTD, a sound speed sensor and a glass-sphere pressure housing containing the electronics, were deployed from the R/V Falkor. One of the platforms imploded at a nominal depth of 9000 m. The highly energetic, broadband shock wave created by the implosion was recorded by the surviving platform. The shock wave reflected multiple times from the seafloor and the surface and the arrival times of the multiple reflections were used to obtain a highly constrained estimate of the Challenger Deep: 10 983 $\pm$ 6 m. We will discuss the signal processing methods used and propagated uncertainty in this highly constrained estimate. We will also discuss how this estimate of the depth of the deepest part of the ocean compares with other estimates of the depth of the Challenger Deep, including an update on estimates obtained following the publication of this study.

Contributed Papers

3:35

2pEA8. Dissecting recorded gunshot sounds. Steven D. Beck (Beck Audio Forensics, 14101 Hwy. 290 West, Bldg 1700, Ste. A, Austin, TX 78737, stevendbeck@alumni.rice.edu)

Gunshot audio recordings are composites of multiple sound sources, including blast, ballistic, mechanical, and environmental sounds. The existence of each sound and its distinguishing features depend on the source characteristics, the propagation environment, and the recording system. Controlled recordings show the situational dependence of these sound sources and the recording system. Knowledge of the physics and these dependences is critical in forensic analysis and the ability to dissect, separate, and identify sources in recorded gunshots. Multiple blast sources (muzzle blast, gas jet, mechanical impact, and primer explosion), ballistic sources (shockwave and subsonic flow), and environmental sources (reflections and terminal impact) will be shown from both controlled experiments and uncontrolled forensic recordings.

3:50

2pEA9. Source characterization of afterburning laboratory-scale jet noise with vector acoustic intensity. Michele L. Eggleston (Phys. and Astronomy, Brigham Young Univ., Brigham Young University, Provo, UT 84604, megglest@byu.edu), Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), Logan T. Mathews, Tyce W. Olaveson, Hunter J. Pratt (Phys. and Astronomy, Brigham Young Univ., Provo, UT), Ashwin Kumar M. G. (Mech. Eng., Virginia Tech, Blacksburg, VA), and Joseph W. Meadows (Mech. Eng., Virginia Tech, Blacksburg, VA)

This paper presents an intensity-based characterization of noise sources in a highly heated laboratory-scale jet with a 3-inch exit diameter. Run conditions had nozzle pressure ratios between 2.7 and 3.5 and temperature ratios of $\sim$7. Measurements were performed using a microphone scanning rig containing 8 four-microphone intensity probes. Probes were separated by 1.5 nozzle diameters and the microphones were spaced 0.5 nozzle diameters apart. Each scan mapped out over 3,000 measurement locations to produce an intensity field map. Vector intensity was processed over a 200 Hz to 20 kHz bandwidth using the phase and amplitude gradient estimator (PAGE) method [D. C. Thomas, *et al*., *J. Acoust. Soc. Am.* 137, 3366–3376 (2015)]. Results show the dominant source region as a function of frequency and are compared to the acoustical holography source reconstruction of the T7-A afterburner engine condition [L. T. Mathews and K. L. Gee, AIAA J., in press (2024)]. Additionally, the intensity-derived source locations are connected to the jet's supersonic and subsonic flow regions. [Work supported by ONR Grant No. N00014-21-1-2069].

Session 2pED

Education in Acoustics, Engineering Acoustics, Musical Acoustics, and Physical Acoustics: Acoustics Education: A Potpourri of Classical and Unusual Materials and Demonstrations

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

Invited Papers

1:00

2pED1. Revisiting a classic experiment in violin acoustics as a classroom demonstration or laboratory activity. Andrew C. Morrison (Natural Sci., Joliet Junior College, 1215 Houbolt Dr., Joliet, IL 60431, amorriso@jjc.edu)

In 1982, Carleen Hutchins built a violin to conduct experiments largely to learn about the role of air cavity resonances on the overall sound produced by the violin. This violin was of a standard design except for 65 holes 5 mm in diameter drilled into the ribs. The holes could be selectively plugged with corks for experimentation. Carleen Hutchins was a skilled luthier, and this violin, playfully called "Le Gruyère," was a finely crafted musical instrument. For an introductory acoustics general education laboratory course, a version of Hutchins' Swiss cheese violin was created by drilling holes in the ribs of a factory-made student-model violin. Students in the course have the opportunity to replicate classic acoustics experiments or this violin can be used for class demonstration purposes. In this presentation, the advantages and challenges of implementing this activity as a laboratory or classroom demonstration are discussed.

1:20

2pED2. Flame suppression due to acoustic streaming by using time reversal. Jay M. Cliftmann (Phys. & Astronomy, Brigham Young Univ., BYU, Dept. of Phys. & Astron., N284 ESC, Provo, UT 84602, jc897@byu.edu), Brian E. Anderson (Phys. & Astronomy, Brigham Young Univ., Provo, UT), and Brian D. Patchett (Phys., Utah Valley Univ., Orem, UT)

Using high amplitude time reversal focusing to extinguish small flames is a novel demonstration of transient acoustic streaming. Typically, others have placed a low frequency acoustic source near the flame in order to generate a high enough particle displacement to extinguish it. In this study, we move the sources far from the flame and demonstrate that the time reversal method can focus transient waves to the flame. The peak acoustic overpressure level needed to extinguish a candle flame in free space is 191 dB peak when using a frequency range of 300 to 15000 Hz. This momentary focus causes acoustic streaming at the flame location after the focusing, which extinguishes the flame. By tracking the flame in high-speed video, we show the displacement of the flame due to the passing acoustic wave and subsequently due to the acoustic streaming. We further show that acoustic streaming extinguishes the flame, not the acoustic particle displacement.

1:40

2pED3. Teaching the basics of acoustics using smartphones. Olivier Robin (Université de Sherbrooke, 2500, bd de l'université, Faculté de Génie - Dpt Génie Mécanique, Sherbrooke, QC J1H1L2, Canada, olivier.robin@usherbrooke.ca)

This communication proposes a basis for offering short introductions to the basic principles in acoustics, like adding decibels and the behavior of Helmholtz or quarter-wavelength resonators. The proposed approach combine mobile and experiential learning approaches using smartphone sensors and applications. The objective of these mobile laboratories is to complement, but not replace, traditional laboratories, which allow for more advanced learning in particular topics. However, the proposed series of experiments can be carried out autonomously; students can explore scientific knowledge in real-life situations and without time or place constraints. Indeed, generic or everyday places can be considered, contrary to traditional laboratories that have to be held in dedicated or specific premises.

2:00

2pED4. Pedagogical strategies to enhance learning and awareness of acoustics within our engineering school community. Olivier Doutres (Mech. Eng., ETS (Ecole de technologie superieure), 1100, rue Notre-Dame Ouest, Montreal, QC H3C 1K3, Canada, olivier.doutres@etsmtl.ca), Kévin Rouard (Mech. Eng., ETS (Ecole de technologie superieure), Montreal, QC, Canada), and Maël Lopez (Mech. Eng., ETS (Ecole de technologie superieure), Montreal, QC, Canada)

Acoustics is taught at the École de technologie supérieure (ÉTS, Montreal, Canada) in a single advanced specialization course during the final year of the mechanical engineering bachelor's program. This course aims to equip students with the skills needed to measure and reduce noise based on the theoretical foundations of industrial acoustics and associated experimental techniques. The fact that the science of acoustics is not well-known among engineering students, coupled with the optional nature of this course, results in an average enrollment of only about thirty students each year (across two distinct sessions), a number further reduced since 2020 due to the unfortunate impact of the pandemic. Paradoxically, Quebec lacks engineers trained in this discipline and often recruits them from abroad. This presentation will aim to showcase various strategies and pedagogical tools that have been used and experimented with in recent years (e.g., flipped classroom, in-class experiments, cellphone measurements, community service-oriented semester projects). The goal is to ensure the quality and enjoyment of student learning and to contribute to raising awareness about acoustics and noise-related issues within the ÉTS community.

2:20

2pED5. Educational uses of a microflown p-u probe. Daniel A. Russell (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg, University Park, PA 16802, dar119@psu.edu)

Two of the most useful transducers I have ever purchased for my research lab are a B&K Type 8001 mechanical impedance head and a Microflown p-u matchstick probe acoustic impedance head. This talk will illustrate several educational demonstrations one can perform using a Microflown p-u probe. One demonstration will show the transition from the near-field of a spherical source (where pressure and velocity are in quadrature) to the far-field (where pressure and velocity are in phase). Another demonstration will illustrate the relative phase differences for pressure and velocity upon reflection from an open or closed end of a pipe. A third demonstration will show how a measurement of acoustic impedance can be used to determine the undamped natural frequency of a Helmholtz resonator, which in turn may be used to accurately predict the length correction at the open end of the resonator mouth. Additional demonstrations may be shown if time permits.

2:40–2:55 Break

Contributed Papers

2:55

2pED6. Converting acoustical demonstrations into in-class experiments. Kurt R. Hoffman (Phys., Whitman College, 345 Boyer Ave., Hall of Sci., Walla Walla, WA 99362, hoffman@whitman.edu)

Teaching acoustical principles to non-science students invites extensive use of classroom demonstrations to help clarify concepts. Ideally, a separate laboratory course opens up many more opportunities for active learning for improved student understanding. However, observing and hearing a demonstration is distinct from performing the activity directly. Moreover, the addition of a laboratory course impacts departmental teaching load and may impact enrollment. In this paper I will discuss laboratory and demonstration activities that have been converted into group activities that are implemented as a classroom activity. The opportunity to add these elements to the course hinge on being able to build or purchase a large number of replicas of equipment needed for the planned activities. The two examples I will discuss in detail are the use of two stringed monochords and pvc trumpets. In addition, I will share a simple mechanical demonstration based on vibrating bars to visualize the location dependent resonances of the basilar membrane.

3:10

2pED7. ROC curve measurements using euro-rack modules (and some DIY). Murray S. Korman (Phys., U.S. Naval Acad., Phys. Dept., U.S. Naval Acad., 572 C Holloway Rd., Annapolis, MD 21402, korman@usna.edu) and Kevin L. Mcilhany (Phys., U.S. Naval Acad., Annapolis, MD)

Acoustic signal processing projects encourage DIY electronics as well as hardware designed to expose principles found either in nature or from man-made sources. Searching within the Eurorack format of the modular synthesis community, functional equivalents to more common DIY units are identified. Some signal generators and measurements are still underrepresented commercially and require DIY Eurorack modules to be built. One goal is to use existing Eurorack components where DIY modules are designed as necessary in this format. A Receiver Operating Characteristic curve (ROC) is calculated for three cases using analog energy detection and probability density function (pdf) circuitry. The hypothesis for noise (N) is a fixed Gaussian noise distribution. The three hypotheses for the signal plus noise (S+N) have signals characterized as; a constant sinusoid with random phase, a broadband unknown Gaussian signal, and an unknown narrow band Gaussian signal utilizing a narrow band carrier frequency. Our classroom demonstration involves two loudspeakers, one for noise, the other for signal. The detected microphone's voltage is bandpass filtered, amplified, squared, integrated and then fed into the pdf circuitry where pulses are counted for 4 seconds in each voltage window. Increasing the signal amplitude will increase the detection index.

3:25

2pED8. Design and construction of an in-air sonar for signal processing education. Kyle S. Dalton (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, ksd5377@psu.edu), Robert Nelson, Steven J. Todd, Jonah Warner, Lily R. Hetherington, John A. Case, Mitchell J. Swann (Graduate Program in Acoust., Penn State Univ., State College, PA), and Constantinos S. Kandias (Aerosp. Eng., Penn State Univ., University Park, PA)

Sonar is an acoustic sensing method commonly used in underwater imaging and navigation. Students are often introduced to sonar theory, but few gain hands-on experience due to the cost and logistical difficulties associated with in-water experimentation. Alternatively, a sonar system can be designed for in-air use to mitigate some of the practical and operational challenges of acquiring sonar field data. Inspired by previous in-air sonar projects, the Pennsylvania State University student chapter of the Acoustical Society of America acquired $5,000 to construct an in-air sonar for exploring signal processing and mechanical design concepts. For a cost-effective solution, consumer audio equipment and additive manufacturing were leveraged. This presentation will walk through the sonar design and build process, discuss the project's current status, and present initial results. Plans for project development and potential inclusion into the curriculum will be discussed. Advice will be offered to those who may want to pursue similar projects.

3:40

2pED9. Demonstration of nonlinear tuning curve vibration response using a homemade analog sweep spectrum analyzer. John Paulenich III (Phys., U.S. Naval Acad., 641 Wayward Dr., Annapolis, MD 21401, jpaulenich@gmail.com) and Murray S. Korman (Phys., U.S. Naval Acad., Annapolis, MD)

A super-heterodyne sweep spectrum analyzer has been designed (using analog circuitry), built and tested to cover the 5 to 1000 Hz range. The analyzer employs a linear ramp voltage versus time, driving a voltage-controlled oscillator VCO that generates a sweep from 32.768 kHz to 33.768 kHz in 70 s. An analog AD734 multiplies the VCO signal by a sinusoidal signal (generated by a watch crystal 32.768 kHz oscillator). The "mixer" signal is low-pass filtered and amplified to generate a 5-1000 Hz swept tone using a 2-in. speaker. The accelerometer (mounted on a thin circular clamped acrylic plate) vibration response involves multiplying this signal by the VCO, then filtering this "mixer" signal using a 4-stage watch crystal ladder filter with a 1 Hz bandwidth. This signal is squared and low-pass filtered to generate the time (converted to frequency) vs. mean-squared voltage response on an oscilloscope. In the demonstration, 300 g of 6 mm diameter glass beads are supported by the 11.4 cm diam, 3.2 mm thick plate and upper rigid cylindrical wall column. The nonlinear tuning curve vibration response is recorded for various incremental drive amplitudes to demonstrate that the tuning curve resonant frequency significantly decreases with increasing drive amplitude.

3:55

2pED10. Exploring comb filtering effects in the recording studio: A lesson on room acoustics. Wesley Bulla (Audio Eng., Belmont Univ., Nashville, TN), Lisa LaFountaine (Belmont Univ., 1900 Belmont Blvd, Nashville, TN 37212, Lisa.LaFountaine@bruins.belmont.edu), and Raymond Plasse (Belmont Univ., Nashville, TN)

A common practice during a live music-recording session is to position microphones, so-called, "room mics" to capture the ambient characteristics of the recording space. An in-class, hands-on experiment was conducted at Belmont University's Historic Columbia Studio A in order to measure and observe the effect of comb filtering on the signal captured by an on-axis room microphone placed at various heights from the floor. A total of seven positions were recorded, spanning from just above the floor to twelve feet in the air. Sine sweep measurements and musical samples from a live virtual jazz-quartet were taken at each position. Students were then asked to listen back and discuss their preferences for each microphone height, comparing their observations to the matching frequency response analysis. A majority of students enjoyed the recording with the microphone positioned at the floor corresponding with the least amount of comb filtering. Although listener preference data was not collected, this experiment demonstrated the use of interdisciplinary examples as an educational tool in the field of acoustics.

4:10

2pED11. Building a guitar amplifier: Accessible, engaging, and project-based audio instruction for students of all academic backgrounds. Benjamin R. Thompson (Elec. & Comput. Eng., Univ. of Rochester, 120 Trustee Rd., Rochester, NY 14620, bthomp23@ur.rochester.edu) and Sarah R. Smith (Elec. and Comput. Eng., Univ. of Rochester, Rochester, NY)

A challenge in designing first-year engineering curriculum is meeting the needs of a group of students with diverse academic backgrounds. Educators in the field of audio and acoustical engineering are often tasked with simultaneously preparing majors to be successful in their subsequent coursework while delivering a course that is both accessible to and exciting for *all* students. At the University of Rochester, we offer a survey course that introduces students to fundamental concepts in acoustics, signals, and audio electronics in which we leverage students' existing experiences with sound and music in a project-based environment. Students complete a series of laboratory exercises where they interact with the science of audio by building a guitar amplifier circuit from scratch. Analyzing a schematic, soldering components, solving equations, seeing waveforms, and HEARING the results of their work, allows all students to engage in the content more deeply than any of these activities in isolation. In this presentation, we will share the specifics of our approach, and provide open-source curriculum in the form of lab manuals, supporting files, and sample PCBs.

4:25

2pED12. Teaching concepts of acoustics in air—Part 3, reverberation. William J. Gastmeier (HGC Eng., 12 Roslin Ave. South, Waterloo, ON N2L2G5, Canada, bill@gastmeier.ca)

This paper has been written to participate in the "Education in Acoustics" session at the 2024 joint conference of the Canadian Acoustical Association and the Acoustical Society of America. It is written to build on Parts 1 and 2 of this series and similarly contains materials extracted from 30 years of teaching to Architects at the University of Waterloo and Dalhousie University in Halifax. Part 1 dealt primarily with sound propagation in air and the concepts of longitudinal wave motion, speed, frequency and wavelength and related effects which relate to what we perceive as pitch. Part 2 expanded on those concepts by discussing superposition and the definition and measurement of sound pressure, decibels and the decibel scale and how to manage decibels, all of which we perceive as loudness. Part 3 deals with reverberation, which is arguably the most important aspect of how sound propagates in indoor spaces. It affects both our ability to communicate and how we experience and perceive the quality of the acoustical environment. Practical demonstrations are provided to enhance learning of the concept of Reverberation. These include "hands on" demonstrations including physical experimentation, an audio demonstration and written materials to advance the concept.

Session 2pMUa

Musical Acoustics: Winds Instruments II

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Jonas Braasch, Cochair
*School of Architecture, Rensselaer Polytechnic Institute, School of Architecture,
110 8th Street, Troy, NY 12180*

Invited Papers

1:00

2pMUa1. Electronic wind instruments for mobility-restricted musicians. David S. Whalen (EMPAC, Rensselaer Polytechnic Inst., Troy, NY), Henry Lowengard (AUMI Consortium, Kingston, NY), and Jonas Braasch (School of Architecture, Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, braasj@rpi.edu)

One of the earliest applications of analog music synthesizers was the simulation of orchestral instruments. Electronic wind instrument sounds were played through piano-like keyboards or interfaces with breath controllers that mimicked physical wind instruments. Synthesizers are also often the best alternative for artists with disabilities to play a musical instrument. This talk focuses on methods and experiences the authors were involved with to design electronic alternatives to wind instruments for people who lost control of their arms. The Jamboxx is a USB controller that was specifically designed as a head-only interface using a breath controller to simulate airflow but also operates as a suck/puff device to activate menus and buttons, a slider to manipulate pitch, and a tilt-sensor to control expression parameters. The Jamboxx also addressed musicians seeking a harmonica-type interface to augment their guitar performance. The AUMI instrument is a touchless device based on camera tracking. This talk will address different sound synthesis methods to achieve realistic wind instruments, the ability to integrate these instruments with other off-the-shelf products (e.g., VST plugins), and special requirements for artists with special needs to play these instruments without too much fatigue. [Work supported by the Craig H. Neilsen Foundation.]

1:20

2pMUa2. A theoretical framework for estimating wind instrument bell reflection and transmission functions. Tamara Smyth (Music, UC San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0021, trsmyth@ucsd.edu)

This work examines the bell reflection/transmission by using a combination of a piecewise waveguide model and an open-end reflection (along with the assumed amplitude-complementary transmission) function, modeled as a first-order shelf filter. The wind instrument bell is first modeled by deriving the chain scattering matrix for a piecewise cylindrical model, constructed using reflection coefficient vectors that have a useful structure of being interleaved with zeros (due to the round-trip delay of two samples in each section). The matrix is then used to yield the instrument (or instrument bell) transfer function which is shown to have both poles and zeros if the open-end reflection/transmission is made frequency dependent. If the open-end is modeled as a first-order filter and incorporated into the instrument model using matrix convolution, filter coefficients can be estimated from values in the reflection coefficient vectors that would otherwise be zero.

1:40

2pMUa3. Resonators in free reed instruments: Good, bad, and unusual effects. James P. Cottingham (Phys., Coe College, 1220 First Ave. NE, Cedar Rapids, IA 52402, jcotting@coe.edu)

Western free reed instruments such as the accordion and the harmonica do not typically use pipe resonators. Yet even without pipes there are two sorts of resonators employed. In the case of mouth blown instruments, free-reed or not, the resonances of the players vocal tract are quite significant. Less often recognized as resonators are the reed chambers necessary to provide a secure mounting for the reed and to properly direct the airstream. The acoustical properties of these may or may not be beneficial. Because of their different reed construction, the Asian free-reed mouth organs require pipe resonators in to function. This paper will present examples of ways in which reed chamber and pipe resonances can enhance or interfere with tone production, as well as some interesting special effects produced by intentional mismatches between reed and resonator.

2:00

2pMUa4. The effects of vowel, pitch, and dynamics on the emotional characteristics of the Mezzo-Soprano, Countertenor, and Baritone Voices. Bing Yen Chang (Comput. Sci. and Eng., Hong Kong Univ. of Sci. and Technol., Hong Kong, Hong Kong), Hiu Ting Chan (Comput. Sci. and Eng., Hong Kong Univ. of Sci. and Technol., Hong Kong, Hong Kong), Man Hei Law (Comput. Sci. and Eng., Hong Kong Univ. of Sci. and Technol., TClear Water Bay, Kowloon, Hong Kong, mhlawaa@connect.ust.hk), and Andrew B. Horner (Comput. Sci. and Eng., Hong Kong Univ. of Sci. and Technol., Hong Kong, Hong Kong)

Previous research on the soprano, alto, tenor, and bass voices have shown that their emotional characteristics change with different vowel, pitch, and dynamics. This work continues the investigation with the mezzo-soprano, countertenor, and baritone voices. Listening tests were conducted whereby listeners gave absolute judgements on the voice tones over ten emotional categories, with the data analyzed using logistic regression. High-arousal categories were stronger for loud tones, whereas low-arousal categories were stronger for soft tones. The categories Happy, Heroic, Romantic, and Comic had an upward trend across the pitch range, whereas Calm, Mysterious, Shy, and Sad had a downward trend. Angry and Scary had different trends among the voices. For most categories, all five vowels were mostly similar in terms of emotional expressiveness, with exceptional cases for the baritone voice. Overall, pitch had the strongest effect and was almost twice as strong as dynamics and vowel, with dynamics slightly stronger than vowel. Vowel O had the largest strength-of-expressiveness overall, closely followed by vowels U and A, and finally vowels I and E last. These results give a quantified preliminary perspective on how vowel, pitch, and dynamics shape emotional expression in the voices across a wide range of voice types.

Contributed Papers

2:20

2pMUa5. Study on the source of overtone series in a harmonica note. Kuiliang Li (Tufts Univ., 640 BOSTON Ave., Unit 509, Medford, MA 02155-1330, likuiliang123456@gmail.com) and Chris Rogers (Tufts Univ., Medford, MA)

The harmonica reed is a cantilever beam and therefore does not produce integer harmonics. High-speed camera video of the reed and vibrometer measurements both demonstrate this expected behavior at the reed tip when being played. However, the acoustical recording shows an integer overtone series with large amplitudes (<10 dB from the fundamental) all the way past 22 kHz. To understand this phenomenon, we eliminated resonant cavities and other possible acoustic feedback systems by plucking the reed with only the reed plate attached. The acoustical and vibrometer data showed the same discrepancy in this case, but only when the reed enters the reed slot in the reed plate. The overtones don't show up when the reed is vibrating completely away from the reed slot. In this paper, we will present these data and show how this phenomenon could be a result of vorticity produced by the sharp edge of the plate and the small air layer gap between plate and reed.

2:35–2:50 Break

2:50

2pMUa6. Abstract withdrawn.

3:05

2pMUa7. One-dimensional acoustic modeling of the şimşal considering the external Interaction of the open finger holes. Diako Kaboodi (Faculty of Graduate and Postdoctoral Studies, Laval Univ., 2345, allée des Bibliothèques, QC, QC G1V 0A6, Canada, diako.kaboodi.1@ulaval.ca), Denis Laurendeau (Faculty of Sci. and Eng., Laval Univ., Québec, QC, Canada), Gary Scavone (Music Res., McGill Univ., Montreal, QC, Canada), and Aaron Liu-Rosenbaum (Faculty of Music, Laval Univ., Québec, QC, Canada)

A one-dimensional model in Python was developed to analyze the acoustic behavior of the Kurdish wind instrument, the şimşal. This model was developed using the Transmission Matrix Method (TMM), which was refined by taking into account the external interactions between the open tone holes. This research focuses on the acoustic resonator part of the şimşal with the lattice of finger holes in different cross-fingering patterns. The results were validated by measurements using an impedance probe, CapteurZ, which was developed by LAUM[1] and CTTM[2] The results are in good agreement with the measurement and despite the possible sources of error, which are discussed, the deviations are within an acceptable range. [1] The Acoustics Laboratory of the University of Le Mans [2] Le Mans Technology Transfer Center.

3:20

2pMUa8. Acoustic impedance measurement head design and evaluation. Champ C. Darabundit (Music Technol., McGill Univ., 550 Rue Sherbrooke O, Montréal, QC H3A 1B9, Canada, champ.darabundit@mail.mcgill.ca) and Gary Scavone (Music Res., McGill Univ., Montreal, QC, Canada)

Acoustic input impedance measurements are a useful tool for characterizing the behavior of a wind instrument. Methods for measuring acoustic input impedances include the two-microphone three-calibration (TMTC) technique (Gibiat and Laloë, 1990) and the multi-microphone method (Jang and Ih, 1998). We present and evaluate the design of a new impedance measurement head based on the multi-microphone method. We will discuss practical aspects such as impedance head calibration, measurement stimulus, and leak detection.

3:35

2pMUa9. Numerical modeling of a saxophone and comparison with experimental measurements. Marie L. Jeanneteau (Res. and development, Henri Selmer Paris, ICA, 3 rue Caroline Aigle, Toulouse, Occitanie 31400, France, marie.jeanneteau@insa-toulouse.fr), Paul Oumaziz, Jean-Charles Passieux, Vincent Gibiat (Metrology Identification Control and Monitoring, Clément Ader Inst., Toulouse, Occitanie, France), and Jonathan Cottier (Res. and development, Henri Selmer Paris, Paris, France)

Saxophones making has long relied on craftsmanship, associated with an empirical knowledge of their acoustic functioning. The design process is now mainly based on the study of the input impedance, accessible experimentally or computationally. The numerical approaches are currently limited either by the accuracy (analytical models) or by the computation time (numerical resolution of PDEs). The challenge is to propose a numerical method combining both, to access the acoustic pressure and velocity fields in the instrument. Our objective was to develop a high-performance parallel numerical tool based on FEM to model accurately (a few cents on the resonances) the acoustic behavior of the resonator under playing conditions. The computation time was reduced by an order of magnitude compared to the brute force approach, using different modeling strategies specific to wind instruments: First, the modeling of the visco-thermal losses at the walls; then, the reduction of the size of the linear systems and finally, the reuse and model reduction to limit the effect of the multiple resolutions imposed by the wide frequency range of interest. The model is validated by comparison with experimental measurements. An impedance measurement system allowing acquisitions under controlled flow has been developed to reproduce realistic playing conditions.

3:50

2pMUa10. Sounding saxophones like flutes. Jonas Braasch (School of Architecture, Rensselaer Polytechnic Inst., School of Architecture, 110 8th St., Troy, NY 12180, braasj@rpi.edu)

The saxophone can be modified with alternative non-single-reed tone generators, including double and free reeds and brass-instrument mouthpieces. It can also be played as a rim flute on the neck, which is a known extended technique for the saxophone. Flutes differ fundamentally from the other tone-generator types as they do not act like a valve, and their flutes are so-called open-open resonator instruments, while reeds and the lips in brass instruments effectively close one end of the resonator. As a practical consequence, all open-close hole key combinations result in fundamentally different pitches known from the regular saxophone. The fact that the saxophone is conically shaped, while the flute is typically cylindrical or inverse conical, further complicates the matter. Approaches to playing the saxophone as flute using alternative fingering combinations will be discussed along with measures of pitch accuracy, timbral, and level balance. While the achievable range aligns with many orchestral wind instruments, complex cross-fingerings make it difficult to play fast chromatic lines. The large bore of the saxophone gives the flute sound a dark character, more like a Native American flute or shakuhachi than a Western concert flute or recorder.

2p TUE. PM

Session 2pMUb

Musical Acoustics: Native American Flute Concert

Gary Scavone, Cochair
Music Research, McGill University, 555 Sherbrooke Street West, Montreal H3A 1E3, Canada

Jonas Braasch, Cochair
*School of Architecture, Rensselaer Polytechnic Institute, School of Architecture,
110 8th Street, Troy, NY 12180*

Chair's Introduction—4:30

Invited Paper

4:35

2pMUb1. A concert by Theresa "Bear" Fox. Theresa. Fox (Mohawk Nation - Akwesasne, Akwesasne, ON, Canada), Jonas Braasch (School of Architecture, Rensselaer Polytechnic Inst., School of Architecture, 110 8th St., Troy, NY 12180, braasj@rpi.edu), and Paula Conlon (School of Music, Univ. of Oklahoma, Ottawa, ON, Canada)

Theresa "Bear" Fox is a prolific singer and songwriter from the Mohawk Nation at Akwesasne. During this concert, she will present songs from her eight CD albums highlighting Akwesasne culture. The ancestral land of the Akwesasne community of about 12,000 people spreads across both banks of the St. Lawrence River and intersects Quebec, Ontario, and New York State. Bear Fox's music is deeply rooted in her living indigenous tradition. Following Akwesasne culture, Bear Fox's songs often have a specific function. Some songs are healing songs, and others are provided as gifts to family members. Bear Fox writes her lyrics in both Mohawk language and English, which will give the audience the opportunity to listen to the authentic sound of the Akwesasne music tradition and directly experience the meaning of words in the larger context of Native American songs. During the concert, Bear Fox will explain how each song has a story that is grown from a planted seed. Songs are often written to honor Mother Earth, mothers, water, wind, Grandmother Moon, and our newborns. Bear Fox believes in the positive message that can come from songs, which is a central part of the Mohawk culture.

Session 2pNSa

Noise, Computational Acoustics and Psychological and Physiological Acoustics: Advanced Air Mobility Noise: Noise from New Air Transportation in Urban and Underserved Communities II

Matthew Boucher, Chair
NASA Langley Research Center, 2 N. Dryden St., M/S 463, Hampton, VA 23681

Chair's Introduction—1:00

Invited Papers

1:05

2pNSa1. Reduced-order equivalent source modeling of aeroacoustic sources. Christian Dreier (Inst. for Hearing Technol. and Acoust., RWTH Aachen Univ., Kopernikusstrasse 5, Aachen 52074, Germany, christian.dreier@akustik.rwth-aachen.de), Xenia Vogt, Wolfgang Schröder (Inst. of Aerodynamics, RWTH Aachen Univ., Aachen, Germany), and Michael Vorlaender (Inst. for Hearing Technol. and Acoust., RWTH Aachen Univ., Aachen, Germany)

The acoustic far-field prediction of high-fidelity aerodynamic simulation data is still not accurately possible when accounting for the surface reflections from an aircraft's fuselage or wings. By presenting a general method for order-reduced equivalent source modeling of aeroacoustic sound sources, this work is bridging the gap between high-fidelity aerodynamic simulations and its integration to auralization frameworks. At the example of a numerically simulated jet, this study shows the computation of an equivalent source model with minimal complexity by means of spherical harmonic (SH) coefficients. The minimal source extension, in which all acoustic sources of a flow field are confined, is computed by using spherical Hankel extrapolation of sound pressure data from virtual concentric microphone arrays. The result of the SH transform shows that the dominant energy contribution can be associated to nine elementary sound sources. The resulting equivalent source model of jet noise provides a convenient date interface between large-scale computational fluid dynamics and acoustics simulations.

1:25

2pNSa2. Acoustic measurements of full-scale electric vertical takeoff and landing aircraft. Shane V. Lympany (Blue Ridge Res. and Consulting, 29 N Market St., Ste. 700, Asheville, NC 28801, shane.lympany@blueridgeresearch.com) and Juliet A. Page (Blue Ridge Res. and Consulting, Asheville, NC)

The Advanced Air Mobility industry is progressing rapidly towards the goal of revolutionizing transportation by connecting urban and underserved communities with electric vertical takeoff and landing (eVTOL) aircraft. To achieve widespread use, eVTOL aircraft must generate acceptable noise in surrounding communities. Many eVTOL aircraft currently under development have novel configurations of rotors, propellers, and wings that may produce rotor-rotor and rotor-airframe interaction noise at certain flight conditions. Acoustic measurements of full-scale eVTOL aircraft are critical to understand the complex noise-generation mechanisms and to validate aeroacoustic models for these novel aircraft designs. In this presentation, we present acoustic measurements of Hexa, an 18-rotor, single-passenger eVTOL aircraft developed by LIFT Aircraft, Inc. We apply acoustic source characterization techniques using simultaneous acoustic measurements and vehicle telemetry data to provide a better understanding of the noise-generation mechanisms. One such analysis technique is the Vold-Kalman filter, which we apply to auralize the tonal waveforms of individual rotors. Finally, we discuss plans for upcoming acoustic measurements of a full-scale, multi-passenger eVTOL aircraft. These measurements and analyses provide the data and tools required to better understand the noise-generation mechanisms and to validate aeroacoustic models for full-scale eVTOL aircraft

1:45

2pNSa3. Onboard sound emission measurements of unmanned aerial vehicles for psychoacoustic experiments. Jonas Jaeggi (Lab. for Acoust. / Noise Control, Empa, Swiss Federal Labs. for Material Sci. and Technol., Ueberlandstrasse 129, Dübendorf 8600, Switzerland, jonas.jaeggi@empa.ch), Jonas Meister, and Reto Pieren (Lab. for Acoust. / Noise Control, Empa, Swiss Federal Labs. for Material Sci. and Technol., Dübendorf, Switzerland)

With the prospect of small UAVs being more commonly used for tasks like surveillance or parcel transport within inhabited areas, demand for noise regulations arises. The noise assessment of UAV operations can be investigated through laboratory listening experiments using auralized sound stimuli. Sound recordings of source signals for dynamic flight operations have to be performed outdoors where measurements with stationary microphones impose uncertainties in backpropagation and suffer from bad signal-to-noise ratio due to the weak source and the presence of ambient sounds. To overcome these difficulties, this contribution presents an approach for acquiring UAV source signals using onboard microphones. A lightweight setup for time synchronous recordings of sound using MEMS microphones, the UAV position and the rotational speed of each rotor was developed and applied to two quadcopters of 0.9 and 6.3 kg mass.

Source directivity and propagation effects were added to the source signals to obtain clean sound pressure signals at virtual listener positions. The resulting stimuli were successfully used in a listening experiment on short-term noise annoyance.

2:05

2pNSa4. Sound from unmanned aerial vehicles in ambient acoustic environments: Influences of context, vehicle characteristics and flight operations on human perception and response. Michael J. Lotinga (Acoust. Res. Ctr., Univ. of Salford, Newton Building-Salford, Salford M5 4BR, United Kingdom, m.j.lotinga@edu.salford.ac.uk), Marc C. Green (Acoust. Res. Ctr., Univ. of Salford, Salford, United Kingdom), and Antonio J. Torija Martinez (Acoust. Res. Ctr., Univ. of Salford, Manchester, United Kingdom)

The potential opportunities for unmanned aerial vehicles (UAVs) to offer wide societal benefits are accompanied by risks of introducing novel sources of community noise impact. Accordingly, it is important to develop a greater understanding of the subjective perception and response to UAV sound. Suitable evidence-based strategies for managing the potential risks to public health and wellbeing can then be devised in the form of flight path optimisation tools, to support initial technology deployment efforts. The 'REFMAP' project aims to develop these tools, which will include a component enabling noise constraints to be considered. In support of this objective, a laboratory listening experiment has been undertaken to study psychoacoustic aspects of UAV sound exposure. The experiment design incorporated influences of contextual auditory and soundscape factors, by embedding spatially rendered UAV sound events within ambisonic recordings of urban acoustic environments. The UAV renderings comprised varying flight operations and numbers of flight events. The experiment was focussed on determining both perceptual noticeability and affective responses to UAV sound, including noise annoyance. Initial results from the experimental data analysis are discussed, in the context of future planned work within the REFMAP research project.

2:25

2pNSa5. A psychoacoustic test on the effect of masking on annoyance to urban air mobility vehicle noise. Matthew Boucher (Structural Acoust. Branch, NASA Langley Res. Ctr., 2 N. Dryden St., M/S 463, Hampton, VA 23681, matthew.a.boucher@nasa.gov), Andrew Christian (Structural Acoust. Branch, NASA Langley Res. Ctr., Hampton, VA), Tyler Tracy (Structural Acoust. Branch, NASA Langley Res. Ctr., Cambridge, MA), Siddhartha Krishnamurthy, Kevin Shepherd (Structural Acoust. Branch, NASA Langley Res. Ctr., Hampton, VA), Durand Begault (Human Systems Integration Div., NASA Ames Res. Ctr., Moffett Field, CA), and Stephen A. Rizzi (NASA Langley Res. Ctr., Hampton, VA)

Urban Air Mobility (UAM) vehicles have a large range of designs and configurations that lead to new noise characteristics and potentially different perceptual responses when compared to traditional aircraft. In addition, UAM vehicles are expected to operate around and within densely populated regions where the presence of ambient background noise is often present. Strategically, this can be leveraged to inform vehicle design and operations to partially or completely mask UAM noise, allowing for mitigation of negative responses and an increased number of allowed operations. A psychoacoustic test was conducted to investigate how masking effects can influence the annoyance response to a low frequency harmonic tone complex (80-320 Hz). To do this, five test subjects compared their annoyance response to the low frequency tonal noise with a higher frequency broadband noise (10 dB down bandwidth between 300 and 2000 Hz), with and without a masking noise present. Detection thresholds were also measured for both sounds to help fit a model to the data. Although the effect of masking on annoyance is complex, results indicate that for some individuals, masking leads to a lower annoyance than the sound level alone would predict.

Session 2pNSb

Noise and Architectural Acoustics: Soundscape—Focus on Applications II

Brigitte Schulte-Fortkamp, Cochair
HEAD Genuit Foundation, Ebert Straße 30 a, Herzogenrath, 52134, Germany

Bennett M. Brooks, Cochair
Brooks Acoustics Corporation, 49 N. Federal Highway, #121, Pompano Beach, FL 33062

Contributed Papers

1:00

2pNSb1. Harmony in confinement: Exploring acoustic and thermal comfort in the Sandiaoling eco-friendly tunnel. Chan N. Truong (Dept. of Architecture, National Taiwan Univ. of Sci. and Tech., 43 Keelung Rd. Sec. 4, Taipei 106, Taiwan, M11213805@mail.ntust.edu.tw), Clarissa Averina, Gabriela Niederberge, Juliana Manuela Muet, Nikita Grace Manullang, Stijn Zeger van Brug, Roxana Ghadiri, Ni Made Putri Indriyani, Phoa Angela Grace Wibowo, Khaing Thinzar, Tuan Sanh Diep, Shiang-I Juan, and Lucky S. Tsaih (Dept. of Architecture, National Taiwan Univ. of Sci. and Tech., Taipei, Taiwan)

The Sandiaoling Eco-friendly Tunnel in New Taipei City, Taiwan, a renowned tourist destination for foot and bicycle travel, features intentionally designed elements such as illumination, sustainable materials, and low-carbon construction. In contrast to open public spaces, the tunnel's conditions are confined by a closed atmosphere, limited direction, and unclear vision. However, the consistent temperature and intermittent natural sounds within the tunnel could contribute to a sense of comfort. To evaluate acoustic comfort at the site, the study utilizes the soundscape study method outlined in ISO 12913. This involves documenting sound sources, paths, and levels, along with a questionnaire to understand visitor preferences. Additionally, infrared thermography images, relative humidity, and wind speed for each measured position are documented and analyzed to explore potential correlations between acoustic and thermal comfort. Preliminary results indicate that the pleasantness level at the site during measurements is not necessarily affected by thermal conditions. Visual and environmental freshness in the tunnel appear to be dominant factors for pleasantness despite the constant occurrence of manmade sounds. A comprehensive understanding of the relationship between thermal conditions and the acoustic environment would necessitate year-round data collection, providing valuable insights into the factors influencing the overall comfort experience for visitors.

1:15

2pNSb2. Unveiling the impact of auditory salience and sources identification on perceived pleasantness in environmental soundscapes. Nicolas Misdariis (Sound Percept. and Design Group, Ircam STMS Lab, 1, Pl. Igor Stravinksy, Paris F-75004, France, nicolas.misdariis@ircam.fr), Baptiste Bouvier (Sound Percept. and Design group, Ircam STMS Lab, Paris, France), Catherine Marquis-Favre (Univ Lyon, ENTPE, Ecole Centrale de Lyon, CNRS, LTDS, UMR5513, Vaulx-en-Velin, France), and Patrick Susini (Sound Percept. and Design group, Ircam STMS Lab, Paris, France)

On the basis of initial results related to modulations of auditory salience—the ability to capture attention—by timbre attributes (especially, brightness and roughness), the present study investigates auditory salience as a predictor of perceived pleasantness in environmental sound scenes. A new paradigm was set up to measure continuous pleasantness while observing the impact of specific events, in a corpus of 11 various soundscapes. Specific events were found to affect perceived pleasantness, in a source-dependent direction in line with the literature. In addition, causality analyses examined the impact of temporal salience predictions on pleasantness ratings, along with other indices (equivalent sound level, loudness, or timbre attributes). They showed that salience was the best predictor of pleasantness, ahead of loudness, level or brightness and roughness. Nonetheless, the last two turned out to play, as well, a significant role in the pleasantness percept. Therefore, auditory salience is confirmed as a relevant indicator for assessing the perception of soundscapes, and the identification of sound sources that compose them is also confirmed to be crucial in this assessment.

1:30

2pNSb3. Towards soundscape management of protected natural areas using the ISO 12913: A field study. Tin Oberman (Inst. for Environ. Design and Eng., Univ. College London, Central House, 14 Upper Woburn, London, United Kingdom, t.oberman@ucl.ac.uk), Simone Torresin (Dept. of Civil Environ. and Mech. Eng., Univ. of Trento, Trento, Italy), Arianna Latini (Dept. of Construction, Civil Eng. and Architecture, Università Politecnica delle Marche, Ancona, Italy), Giacomo Gozzi (Silenzi in Quota, Trento, Italy), Francesco Aletta (Univ. College London, London, United Kingdom), and Jian Kang (Inst. for Environ. Design and Eng., Univ. College London, London, United Kingdom)

Human perception of soundscapes in protected natural areas like national parks is crucial for their protection. At popular scenic spots, visitors themselves often contribute to noise pollution. Decibel-based systems (such as LAeq or Lden) do not fully explain human reactions to this phenomenon, necessitating a more holistic approach to allow for an effective management strategy. A mixed-methods soundscape approach based on the ISO 12913 series, developed mostly in urban soundscape studies, was tested in four protected natural ex-urban areas in the Dolomites (Italy) and Cairngorms (United Kingdom). During five soundwalks (7-12 km long), conducted by adopting the Method A of ISO/TS 12913-2, a total of 443 questionnaire responses were gathered across 28 evaluation points, alongside corresponding binaural measurements. A range of acoustic environments as quiet as LAeq = 31 dBA and as loud as LAeq = 76 dBA were observed, eliciting perceptions ranging from very calm to chaotic. A Linear Mixed-Effects Model was computed to analyse the impact of sound source dominance, psycho-acoustic and environmental acoustic indices on perception. Presence of human sounds proved to be a major factor driving the perception of chaotic soundscapes.

1:45

2pNSb4. Relationship between greenery and the health of Madrid's citizens. Guillermo Rey-Gozalo (Física Aplicada, Universidad de Extremadura, Av. Universidad s/n, Cáceres 10003, Spain, guille@unex.es), Juan Miguel Barrigón Morillas, David Montes González, Rosendo Vílchez-Gómez (Física Aplicada, Universidad de Extremadura, Cáceres, Spain), Carlos Iglesias-Merchán (Universidad Politécnica de Madrid, Madrid, Spain), Silvia Merino-de-Miguel (Universidad Politécnica de Madrid, Madrid, Spain), Pierre Aumond (Université Gustave Eiffel, Nantes, France), Laura Muñoz-Bermejo (Universidad de Extremadura, Mérida, Spain), José Manuel Pérez Pintor (Universidad de Extremadura, Cáceres, Spain), and Celia Moreno González (Física Aplicada, Universidad de Extremadura, Cáceres, Spain)

Several studies have demonstrated the relationship between the presence of green spaces and a decreased likelihood of experiencing health issues or exposure to environmental pollutants, including noise. The increase in green areas will be one of the interventions in future cities to reduce pollution and enhance the quality of life for their citizens. However, it is crucial to identify the variables within green spaces that have the most significant impact on well-being and health. Different green areas in the city of Madrid were analyzed in this study, and simultaneously with the recording of environmental variables, their users were surveyed. The study's hypothesis aimed to determine whether the use of green areas has a greater impact on health than their mere presence. The quality of green areas is an influential factor in these relationships; therefore, the quality of the acoustic environment was considered. The results indicate that the acoustic environment is a significant factor in the development of various activities and the overall perception of the park. The visibility of greenery is a variable with psychological benefits, but the use of green spaces with good environmental quality also has a direct relationship with health status.

2:00

2pNSb5. A soundscape perspective for biophilia hypothesis: Theoretical considerations and conceptual approach. Enkela Alimadhi (Interior Architecture and Environ. Design, Bilkent Univ., Faculty of Art, Design and Architecture, Ankara 06800, Turkey, enkelaalimadhi@bilkent.edu.tr) and Semiha Yilmazer (Interior Architecture and Environ. Design, Bilkent Univ., Ankara, Turkey)

This study proposes a conceptual framework to consider the integration of the soundscape perception into the Biophilia Hypothesis paradigm and to further investigate the positive psychological outcomes such as affection or cognition indicators, and physiological outcomes or neurophysiological outcome indicators based on brain signal analysis on building occupants. The biophilia hypothesis claims that a natural content environment has positive psychological and physiological outcomes identified as restorative. The established soundscape framework, which is still under exploration in indoor environments, will help frame the rationale of this study to understand the soundscape perception and consider how to integrate it into the biophilic theoretical claims. In this line of thought, the conceptual framework emphasizes the significance of not just the visual realm, as suggested by the Biophilia hypothesis paradigm through empirical, experimental, and theoretical claims. Additionally, incorporating sound can contribute to a more holistic approach. To do so, based on an literature review and also on our very preliminary study this conceptual framework is an extension of the soundscape theoretical framework. It proposes the integration of complementary theories such as Attention Restoration, Stress-Recovery Theory, Biophilia hypothesis, and the soundscape framework to be further implemented for healthy designed environments.

2:15–2:55
Panel Discussion

Session 2pNSc

Noise and ASA Committee on Standards: Honoring the Life and Legacy of Tony F. W. Embleton

Bennett M. Brooks, Cochair
Brooks Acoustics Corporation, 49 N. Federal Highway, #121, Pompano Beach, FL 33062

Nicholas Sylvestre-Williams, Cochair
Aercoustics Engineering Ltd., 165-50 Ronson Drive, Toronto, M9W 1B3, Canada

Chair's Introduction—3:05

Invited Papers

3:10

2pNSc1. Tony Embleton—Distinguished researcher, successful manager, consummate acoustician. Bennett M. Brooks (Brooks Acoust. Corp., 49 N. Federal Hwy., #121, Pompano Beach, FL 33062, bbrooks@brooksacoustics.com)

Tony F. W. Embleton was a distinguished researcher in acoustics. His early career focused on practical noise control projects and experimental non-linear acoustics. Later, Tony made important contributions to the understanding of sound propagation outdoors. Tony spent much of his career at the National Research Council of Canada (NRCC). His professional homes were the ASA and the CAA. Tony received numerous awards from the ASA for his work, including the R. Bruce Lindsay Award in 1964, the ASA Silver Medal in Noise in 1986, and the ASA Gold Medal in 2002 "For fundamental contributions to understanding outdoor sound propagation and noise control and for leadership in the Society." Tony recognized the value of professional organizations and served the ASA as Vice-President (1977–78) and President (1980–81). He also understood the importance of practical applications in acoustics serving as ASA Standards Director (1993–97), navigating the organization through a tough period, and growing the program toward a solid future. Most importantly, Tony was known for his kind and generous spirit, and his outreach and inspiration to newer colleagues. Many of us are the grateful beneficiaries of his gentle enthusiasm.

3:40

2pNSc2. Tony F. W. Embleton in the Acoustics Section, Physics Division, of the National Research Council of Canada (NRCC). Anthony J. Brammer (Medicine, Univ. of Connecticut Health, Farmington, CT, brammer@uchc.edu), Gilles Daigle (Gatineau, QC, Canada), Michael Stinson (Carlsbad Springs, ON, Canada), and Floyd Toole (Ottawa, ON, Canada)

Tony completed his PhD in three years at Imperial College (London) studying under Dr. R.W.B. Stephens. He then joined George Thiessen and Edgar Shaw at NRCC to form the core of what became arguably the most influential and productive research group in acoustics in Canada from the 1960s to the 1980s. An important activity was service to industry, a role Tony embraced by collaborating with industry associations, participating in committees and directing his research activities. Working with George Thiessen, he succeeded in reducing the noise of couch rolls, a major source of noise in paper making, by randomizing the pattern of holes through which air was sucked to dry the paper. Other successful noise control projects included staggered stator blades for gas turbine engines and mufflers for rock drills. Tony made seminal contributions to outdoor sound propagation and refined condenser microphone calibration. In addition to his research and outreach, he found time for professional service to the ASA and CAA, including serving as founding editor of what is now Canadian Acoustics, and to develop standards and mentor younger scientists, He is remembered for his cheerfulness and willingness to provide advice and engage in conversation on any and all topics.

4:10

2pNSc3. The 1952 Ph.D. dissertation of T.F.W. Embleton on nonlinear acoustic propagation and reflection. Victor W. Sparrow (Grad. Prog. Acoust., Penn State, 201 Appl. Sci. Bldg., University Park, PA 16802, vws1@psu.edu)

In his doctoral research Tony Embleton studied nonlinear acoustics experimentally using techniques available in the Physics Department of Imperial College, University of London. Originally developed by V. Timbrell, the interferometric equipment was substantially improved by Tony Embleton, enabling his Ph.D. findings. Studying propagation in a tube and reflection by both closed and open ends, Tony Embleton was able to substantially confirm the theory of the time, developed by Stokes, Earnshaw, Rankine, Rayleigh, and Taylor, many pieces confirmed for the first time. He also found discrepancies regarding the attenuation coefficients regarding the first and second half cycle of each pulse as well as a shortening of a pulse at reflection from a closed end. These findings are still worthy of further investigation. The results of reflection of pulses from the open end of a tube, forming a pressure release boundary condition, still are very interesting to the present author. A pulse propagating toward the open end steepens, but the reflected wave "unsteepens" which is a unique nonlinear acoustic phenomenon. [Work supported by the Penn State College of Engineering and its United Technologies Corporation Professorship.]

4:40

2pNSc4. Tony Embleton and the ASA Standards Program. Bennett M. Brooks (Brooks Acoust. Corp., 49 N. Federal Hwy., #121, Pompano Beach, FL 33062, bbrooks@brooksacoustics.com) and Stephen J. Lind (LindAcoustics LLC, Onalaska, WI)

Tony F. W. Embleton had a distinguished career as a researcher in acoustics, and as a leader in the ASA and CAA. Yet, he was keenly aware of the importance of standardization to practical applications in acoustics. Tony served as ASA Standards Director from 1993 to 1997, when the organization was going through a tough period, even facing abolishment by the ASA Executive Council. Tony reinvigorated the ASA Standards Program, including increasing its connection to the Technical Committees, as well as its administrative operations [T. F. W. Embleton, "Standards and the Acoustical Society," J. Acoust. Soc. Am. 98, 691–693 (1995); S. Blaeser and C. J. Struck, "A history of ASA standards," J. Acoust. Soc. Am. 145 (1), (2019)]. Tony also was instrumental in founding CAA's Standards program. Tony's legacy of engagement with the ASA technical and administrative leadership is still evident today. An overview of how Tony's ideas remain a guiding force in the current ASA Standards Program is given.

5:10–5:30
Panel Discussion

TUESDAY AFTERNOON, 14 MAY 2024 — ROOM 202, 1:00 P.M. TO 5:30 P.M.

Session 2pPA

Physical Acoustics, Structural Acoustics and Vibration and Engineering Acoustics: Resonant Ultrasound Spectroscopy For Characterizing Material Property and Structure

Christopher M. Kube, Cochair
Engineering Science and Mechanics, The Pennsylvania State University, 212 Earth and Engineering Sciences Bldg, University Park, PA 16802

Matthew Cherry, Cochair
Air Force Research Laboratory, 2230 10th St., Fairborn, OH 45433

Rasheed Adebisi, Cochair
UDRI, University of Dayton, 141 Firwood Drive, Shroyer Park Center, Dayton, OH 45419

Jeff Rosin, Cochair

Invited Papers

1:00

2pPA1. Resonant ultrasound spectroscopy for textured materials. Julian D. Maynard (Phys., The Penn State Univ., 104 Davey Lab, Box 231, University Park, PA 16802, maynard@phys.psu.edu)

Resonant ultrasound spectroscopy (RUS) is a method in which the least-squares fitting of a sample's measured resonance frequencies is used to determine the sample material's elastic constants. Difficulties arise when RUS is applied to textured materials, which are composed of crystallites that are neither completely aligned nor randomly oriented; such materials would have effective elastic tensors with 21 independent elastic constants, and this would overburden the conventional RUS analysis process. In any case, the elastic nature of some textured materials may be represented as the weighted average of the elastic tensors of the constituent crystalline material rotated in the directions of the individual crystallites. The collection of weights is the orientation distribution function (ODF). Now the least-squares fitting of a sample's measured resonance frequencies may be used to determine not the sample's elastic constants but instead its ODF. This is not trivial since the weights must be positive and sum to one, but it may be done with a simple modification of the conventional RUS software.

1:20

2pPA2. Characterization of effective elastic constants and anisotropy directions for Wire Arc Additive Manufactured steel samples using RUS. Florian Le Bourdais (DIN, Commissariat à l'Énergie Atomique (CEA), CEA Ctr. de Saclay, Gif sur Yvette 91191, France, florian.lebourdais@cea.fr), Audrey Gardahaut, and Nicolas Leymarie (DIN, Commissariat à l'Énergie Atomique (CEA), Gif sur Yvette, France)

In recent years, Resonant Ultrasound Spectroscopy (RUS) has been extensively applied to objects produced by additive manufacturing to characterize elastic material properties, detect defects or geometrical deviations. In this talk, we analyze samples that were produced using the wire-arc additive manufacturing (WAAM) process using different grades of steel wires. Resonance spectra were obtained and allowed to classify samples as either elastically isotropic or anisotropic. Detailed investigation on anisotropic samples (produced with 316L wire) under an orthotropy hypothesis showed that the samples were markedly softer along the layer deposition direction. Subsequent investigation using EBSD confirmed the results obtained with RUS. They also allowed to quantitatively model the elastic constants using the Voigt-Reuss-Hill averaging theory, which were in good agreement with the ones obtained using the RUS inverse problem.

1:40

2pPA3. Resonant ultrasound spectroscopy experimental design and optimization. Joshua Ward (Univ. of Dayton Res. Inst., 300 College park, Dayton, OH 45469, joshua.ward@udri.udayton.edu), Mark Obstalecki, and Matthew Cherry (Air Force Res. Lab., Fairborn, OH)

Resonant Ultrasound Spectroscopy (RUS) is a nondestructive material characterization technique that utilizes solid body resonances to estimate elastic moduli. This method is advantageous over traditional mechanical testing due to the significantly lower number of required samples to estimate the same number of moduli. The goal of this project is to develop a high-throughput RUS system capable of measuring experimental resonances and mode shapes in a fraction of the time of traditional RUS measurements. A new system was designed which leverages data/image analysis algorithms to aid in sample alignment and data collection. A standardized sample geometry was also analyzed by performing a geometric sensitivity study and optimizing the aspect ratio of lengths of a parallelepiped. The results show that the optimized sample geometry is consistent between materials with varying elastic properties. In order to validate the design of the new RUS system a comparative study to an existing system was conducted. The results show that the newly designed system produces comparable resonant frequency and mode shape measurements.

Contributed Papers

2:00

2pPA4. Modeling resonant ultrasound of synthetic polycrystalline microstructures produced by Dream3D. Christopher M. Kube (Eng. Sci. and Mech., The Penn State Univ., 212 Earth and Eng. Sci. Bldg, University Park, PA 16802, kube@psu.edu)

Resonant ultrasound spectroscopy (RUS) is established for determining elastic constants of homogeneous solids. Often applied to polycrystalline materials like metals, RUS presumes homogeneity based on the microstructure's scale. This presumption is challenged with large grain sizes and higher-order modes, complicating homogeneity testing in real samples due to the need for 3D metallurgical analysis. An alternative is RUS modeling on digitized synthetic microstructures. This presentation investigates the RUS variational model with Rayleigh-Ritz solutions on microstructures created using Dream3D. It starts with a proof for solutions that minimize the Lagrangian in heterogeneous solids. Subsequently, Rayleigh-Ritz volume integrals are numerically evaluated for synthetic polycrystalline microstructures, as spatial heterogeneity precludes analytical solutions. No homogeneity assumption is made. The 3D integrals, requiring advanced GPU processing for a detailed mesh, lead to forward model results. These will be compared against common homogenization schemes, highlighting new insights when using RUS for material characterization of polycrystalline materials.

2:15

2pPA5. Constrained anisotropy using spectral structures of elastic tangent tensors. Jim W. Colovos (Vibrant Corp., 8440 Washington St. NE, STE B, Albuquerque, NM 87113, jcolovos@vibrantndt.com)

Geomaterials and biomaterials that are assumed to have symmetry about a single preferred direction have five independent transversely isotropic elastic constants. Spectral decompositions of the anisotropic elastic tangent stiffness tensor have led to an elegant framework for a variety of symmetry classes known to be of importance for characterizing material property and structure. The fourth-order eigenprojectors can be combined to form the isotropic and anisotropic basis tensors with various spatial symmetries and computationally motivated constraints. We consider the mathematically motivated decoupling of tensorially linear and non-linear functions of a texture tensor. Neglecting the non-linear components, as often done for cracked-rock models, imposes a constraint that the lateral shear modulus depends on the remaining elastic constants. We also consider the mathematically motivated decoupling of purely spherical and purely deviatoric components, often used in the field of biomechanics. When the mixed spherical-deviatoric components are neglected, the axial and lateral Poisson's ratios are constrained to become dependent on the two tensile moduli and a new independent bulk modulus type parameter. The described constraints reduce the number of independent elastic constants from five to four. The potential for using similar techniques to investigate elastic constants is discussed for resonant ultrasound spectroscopy, specifically additively manufactured materials, with the goal of increasing robustness to local minima.

Invited Papers

2:30

2pPA6. Abstract withdrawn.

2:50

2pPA7. Enabling resonant ultrasound spectroscopy in extreme environments. Christopher A. Mizzi (National High Magnetic Field Lab., Los Alamos National Lab., Los Alamos, NM) and Boris Maiorov (National High Magnetic Field Lab., Los Alamos National Lab., Los Alamos Nat. Lab, MS E536, Los Alamos, NM 87545, maiorov@lanl.gov)

Elastic moduli are fundamental thermodynamic susceptibilities that connect to thermodynamics, electronic structure, and mechanic properties. Thus, determining the changes of elastic moduli as a function of time or temperature is a powerful tool to study several thermodynamic and materials phenomena. Resonant Ultrasound Spectroscopy (RUS) determines elastic constants with high accuracy and precision from a single measurement of the mechanical resonances of a sample. Conventionally, the quantitative extraction of elastic moduli with RUS assumes free boundary conditions but lead to unstable positioning of the sample making it incompatible with extreme environments like high magnetic fields. In practice, even holding the sample produces contact forces that deviate from free-boundary conditions shifting resonant frequencies. In this talk, we show that, we can reduce the free-boundary conditions (by gluing/adhering the sample to the transducer), while still being able to obtain the full elastic tensor by a simple model of the sample-transducer interaction. This means elastic constants can be determined to within the uncertainty of conventional RUS, but with significant improvements in sample stability and control of sample orientation. We show the results of studying of several magnetic and structural phase-transitions using this new method.

3:10–3:30 Break

3:30

2pPA8. Phonon dispersion at long and short wavelength as seen by ultrasound and neutron spectroscopy. Raphael Hermann (Mater. Sci. and Technol. Div., Oak Ridge National Lab., 1 Bethel Valley Rd., Oak Ridge, TN 37830-8050, hermannrp@ornl.gov)

Lattice excitations are fundamental energy carriers responsible for thermal transport in every solid. Whereas ultrasound spectroscopy typically probes these excitations in the kHz to MHz "sound" regime corresponding to mm to μm wavelengths, inelastic scattering of neutrons or x-rays probe the THz regime and Å wavelength. We will discuss a few examples of materials with peculiar sound and phonon properties, such as chiral α-$TeO_{;2}$ and the $AgSbTe_{;2}$-PbTe (LAST) solid solution. Analysis of resonant ultrasound data of α-$TeO_{;2}$ requires the use of code that supports chiral space groups, such a RUSCal; intriguingly sound is extremely anisotropic, and in some directions the speed of sound is highly frequency dependent in the 0–1 THz range. LAST exhibits nanoscale inhomogeneities where the speed of sound is different in the matrix and the nanoinclusions. Work supported by the US Department of Energy (DOE), Office of Basic Energy Science (BES), and by Laboratory Directed Research at Oak Ridge National Laboratory; this research used resources at the Spallation Neutron Source a facility supported by DOE, BES, Scientific User Facilities Division. J. Torres, A. Flores-Bettancourt, M. Manley, B. Winn, G. Yumnam, I. Sergeev, P. Bauer-Pereira and A. Jafari are gratefully acknowledged for their collaboration.

Contributed Paper

3:50

2pPA9. Using resonant ultrasound spectroscopy to characterize thermally conditioned high explosive materials. Jordan Lum (Lawrence Livermore National Lab., 7000 East Ave., Livermore, CA 94550, lum21@llnl.gov), David M. Stobbe, Paul Mirkarimi, William Shaw, Henry Reinstein (Lawrence Livermore National Lab., Livermore, CA), Rebecca Lindsey (Chemical Eng., Univ. of Michigan, Ann Arbor, MI), and Richard Gee (Lawrence Livermore National Lab., Livermore, CA)

The ability to nondestructively quantify changes in mechanical properties of granular high explosive materials due to thermal conditioning is of importance for a myriad of civil and defense applications and could lead to better understanding of environmental aging-related effects for explosive material performance and safety. In this study, we report the first demonstration of using resonant ultrasound spectroscopy (RUS) to quantify the bulk elastic properties of granular high explosive materials at different bulk pressing densities and degree of thermal conditioning. Monitoring elastic property changes in granular explosive pressings has not yet been demonstrated using RUS, which is an appealing nondestructive characterization tool since it requires only dry point contact with the explosive material and can be applied to explosives with small form factors. Here, explosive material was studied in the form of binderized plastic bonded explosive pressings and unbinderized neat pressings, both of which are used commercially. Elastic stiffness coefficients calculated from the RUS measurements on the binderized and neat pressings show a significant increase for the post-conditioned samples compared to the pre-conditioned samples. This trend of increasing elastic properties with thermal conditioning was consistent for different density pressings and different thermal exposure conditions. [This work was performed by LLNL under Contract No. DE-AC52-07NA27344 and document release number LLNL-ABS-858868.]

Invited Papers

4:05

2pPA10. Microcavity-enhanced photoacoustic vibrational spectroscopy of single particles. Yun-Feng Xiao (School of Phys., Peking Univ., No. 5, Yiheyuan Rd., Haidian District, Beijing 100871, China, yfxiao@pku.edu.cn)

Confinement and manipulation of photons using microcavities have triggered intense research interest in both fundamental and applied photonics for more than two decades. Prominent examples are ultrahigh-Q whispering gallery microcavities which confine photons using continuous total internal reflection along a curved and smooth surface. The long photon lifetime, strong field confinement, and in-plane emission characteristics make them promising candidates for enhancing light-matter interactions on a chip. In this talk, I will focus on single-particle photoacoustic vibrational spectroscopy using optical microcavities.

4:25

2pPA11. Laser-based resonant ultrasound spectroscopy for monitoring secondary phases in metastable Ti-alloys: From growth kinetics to high-throughput characterization. Hanus Seiner (Inst. of Thermomechanics, Czech Acad. of Sci., Dolejskova 5, Prague 18200, Czechia, hseiner@it.cas.cz), Petr Sedlak, Michaela Janovska, and Jitka Nejezchlebova (Inst. of Thermomechanics, Czech Acad. of Sci., Prague, Czechia)

Metastable beta-Ti alloys undergo complex microstructural changes when annealed for extended times (hours, days, weeks) at temperatures below beta-transus. Contact-less RUS turns out to be a perfect tool for monitoring these processes, as well as for analyzing the properties of the alloys after the heat treatment. In the talk, we will illustrate this ability with three examples of recent advancements in this field. In particular, we will show that RUS is capable of analyzing (i) growth kinetics [1], from fast non-equilibrium phenomena to extremely slow processes; (ii) cubic symmetry conservation in single crystals with particles of various secondary phases [2]; and (iii) phase composition-elasticity relationships in series of oligocrystalline samples [3]. [1] Nejezchlebová *et al*. Elastic constants of β-Ti15Mo (2019) J Alloys Compounds, 792, pp. 960–967. [2] Olejňák *et al*. An ultrasound-based evaluation of cubic symmetry preservation and homogeneity in elastic behavior of $\beta+\omega$ and $\beta+\alpha$ Ti-alloys (2023) Materials and Design, 236, art. no. 112474. [3] Preisler *et al*., High-throughput characterization of elastic moduli of Ti-Nb-Zr-O biomedical alloys fabricated by field-assisted sintering technique (2023) J Alloys Compounds, 932, art. no. 167656.

2p TUE. PM

Contributed Papers

4:45

2pPA12. Convergence properties of eigenfrequencies in RUS. Farhad Farzbod (Mech. Eng., Univ. of MS, 1764 University Circle, Rm. 203, University, MS 38677, farzbod@olemiss.edu) and Casey Holycross (Aerosp. Systems Directorate (AFRL/RQTI), Air Force Res. Lab., Dayton, OH)

An in-depth exploration of the asymptotic behavior of resonant frequencies in Resonant Ultrasound Spectroscopy (RUS), a non-destructive method for material property evaluation, is presented in this work. This work delves into the asymptotic analysis using Legendre polynomials for cuboid samples and examines the impact of increasing elements on eigenfrequencies. In this examination, we noted various elements concerning the computational boundaries and the influence of diagonal components on the asymptotes. The presence of a boundary for the asymptotes indicates limited information beyond a certain point. Additionally, how elastic constants influence these eigenfrequencies is discussed.

5:00

2pPA13. Inverse problems in resonant ultrasound spectroscopy based on spectra perturbation. Petr Sedlak (Dept. of Ultrasonic Methods, Inst. of Thermomechanics, Czech Acad. of Sci., Prague, Czechia, psedlak@it.cas.cz), Hanus Seiner, and Michaela Janovska (Dept. of Ultrasonic Methods, Inst. of Thermomechanics of the CAS, Prague, Czechia)

In RUS, the fit between the experimental resonant spectrum and the result of the inversion procedure is usually worse than the inaccuracy of the experimental measurement of the resonant frequencies. Thus, it can be problematic to extract any valuable information from slight perturbation of experimental spectra, where the experimental frequency shift falls well below the accuracy of this fit. However, the inverse procedure can be reformulated and instead of fitting the resonant frequencies itself, it can be based on the direct fitting of the frequency perturbations. In this contribution, we show two examples of the use of this approach (i) quantitative RUS evaluation of the elasticity of micrometric and submicrometric surface layers from frequency differences measured before and after layer deposition [1] and (ii) *in situ* characterization of the thermal degradation of the bonding layer in a bimetallic system [2]. [1] Thomasová M. *et al*. Young's moduli of sputter-deposited NiTi films determined by resonant ultrasound spectroscopy: Austenite, R-phase, and martensite (2015) Scripta Materialia, 101, pp. 24–27. [2] Janovská M. et al Characterization of bonding quality of a cold-sprayed deposit by laser resonant ultrasound spectroscopy (2020) Ultrasonics, 106, art. no. 106140.

5:15

2pPA14. Measuring the internal damping of thin metal beams in flexure. Micah Shepherd (Brigham Young Univ., N249 ESC, Provo, UT 84602, mrs74@byu.edu), Peter K. Jensen (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), and Joshua T. Mills (Brigham Young Univ., Provo, UT)

Since the internal damping of most metals is a relatively small quantity, precise measurement requires that both acoustic radiation damping and boundary condition damping be either eliminated or highly minimized. In this talk, an experiment setup will be described which can measure the internal damping of metal beams in flexure. To perform a measurement, a beam specimen is placed on wire supports within a vacuum chamber and excited by an autonomous force hammer located within the chamber. The wire supports are located exactly at the nodal lines of the beam mode of interest, so that the frictional losses created by the wire supports are minimized. Once excited, the beam deflection is measured with a single point laser vibrometer. The signal decay measured by the vibrometer is used to estimate damping of the mode of interest using either the Hilbert transform or the half-power method. The wire supports are then moved, and the procedure is repeated for the next mode of interest. Results for several metal specimens will be presented and compared to Zener's thermoelastic model, which describes the losses created by thermal currents for a beam in flexure, to demonstrate the accuracy of the measurement method.

Session 2pPP

Psychological and Physiological Acoustics: The Importance of Binaural Listening: Speech Intelligibility, Localization, and Virtual Environments

Axel Ahrens, Cochair
Technical University of Denmark, Ørsteds Plads 352, Kgs. Lyngby, 2800, Denmark

Mark A. Stellmack, Cochair
Psychology, University of Minnesota, 75 East River Parkway, Minneapolis, MN 55455

Chair's Introduction—1:00

Contributed Papers

1:05

2pPP1. Using vocoded speech to study temporal weighting in spatial release from masking. G. Christopher Stecker (Ctr. for Hearing Res., Boys Town National Res. Hospital, 555 N 30th St., Omaha, NE 68131, cstecker@spatialhearing.org) and Brittany T. Williams (Boys Town National Res. Hospital, Omaha, NE)

Several aspects of binaural and spatial hearing appear to be dominated by moments of rising envelope fluctuations ("onsets"), although those phenomena are limited to events occurring less than a few hundred times per second. Thus, for example, the localization of a brief tone is dominated by the binaural information available in the overall sound onset. Whereas for a modulated sound each envelope period may contribute equally, assuming the modulation rate is not too high. For speech signals, syllabic modulations provide multiple onset-like events which may occur synchronously or asynchronously across frequency bands. Although previous studies have shown that rising envelope fluctuations play an outsized role in localization and lateralization, it is unknown whether other aspects of spatial hearing–such as spatial release from masking (SRM)–are similarly dominated by rising envelopes. Here, we used a Gabor-click-train vocoder to transform speech sounds and manipulate the spatial content of rising- versus falling-envelope segments of the speech in each frequency band. Introducing binaural differences between target and masker speech allows the assessment of localization and/or SRM across conditions in which the cues are limited to rising- or falling-envelope segments, or available throughout the signal. [Work supported by NIH R01DC016643, T32000013.]

1:20

2pPP2. A spatial digit task for assessing binaural function in individuals with hearing loss. Douglas Brungart (Walter Reed National Military Medical Ctr., Bethesda, MD, douglas.s.brungart.civ@health.mil), Alyssa Davidson, Kelli Clark (Walter Reed National Military Medical Ctr., Bethesda, MD), and Trevor T. Perry (National Ctr. for Rehabilitative Auditory Res., Bethesda, MD)

Spatial hearing is crucial in occupational tasks, yet until 2019, US Army fitness-for-duty requirements only considered hearing thresholds in the better ear. The regulation change introduced the Military Operational Hearing Test (MOHT) for those with worse-ear thresholds exceeding 40 dB HL at 0.5 or 1 kHz, or 60 dB HL at 2 kHz. This led to the development of the Spatial Digit Test (SDT) to assess binaural function in individuals with poor hearing in one ear but sufficiently good hearing in the other ear to pass a speech-in-noise test. The SDT involves digit pairs presented with Interaural Time Delays (ITDs) of $+/-800\,\mu$s. Two studies were conducted: a validation study with over 200 individuals, and a verification study with more than 130 undergoing MOHT exams. Results show that individuals with normal auditory thresholds perform very well on the SDT, with about 95% correctly identifying at least 8 digits and 75% correctly identifying all 10. In contrast, only about 50% of individuals with monaural losses who took the SDT as part of the MOHT identified 8 or more digits. Results of the SDT were correlated both with subjective hearing complaints and with the Binaural Masking Level Difference. In conclusion, the SDT proves valuable for identifying individuals struggling using ITD cues to segregate and localize simultaneously presented speech signals.

1:35

2pPP3. The effect of eye-gaze direction on speech intelligibility. Tongthai Taotong, Torsten Dau (Tech. Univ. of Denmark, Kgs. Lyngby, Denmark), and Axel Ahrens (Tech. Univ. of Denmark, Ørsteds Plads 352, Kgs. Lyngby 2800, Denmark, aahr@dtu.dk)

In many daily situations we move our eyes to capture different visual information. Previous studies have shown interactions between the eye-gaze direction and auditory perception. The eye-gaze direction has been shown to modulate activity to acoustic stimuli in the auditory cortex of monkeys. In humans, decreased thresholds of binaural lateralization cues and improved accuracy of understanding series of digits in the presence of other speech has been shown. Here, we investigated the effect of the eye-gaze direction on speech perception of single-digit and 4-digit streams while directing the gaze at different directions containing either speech or non-speech stimuli. The speech interferers were also digits but a different gender than the target speech. The non-speech interferers consisted of speech-modulated speech shaped noise. The results showed improved speech intelligibility when the gaze direction and the auditory target direction were matching, in comparison to conditions where the gaze is pointed at a non-target direction. No differences were found comparing single-digit and 4-digit streams. Presenting speech or non-speech stimuli at the gaze direction did not affect the speech intelligibility results. These findings further clarify the influence of the gaze direction on speech perception.

1:50

2pPP4. Relation between spatial release from informational masking and localization discrimination. Benjamin H. Zobel (Speech, Lang., and Hearing Sci., Univ. of Massachusetts Amherst, 358 N Pleasant St., Rm 201A, Amherst, MA 01002, bzobel@umass.edu), Patrick Zurek (Sensimetrics Corp., Woburn, MA), Emily Buss (Otolaryngology/Head and Neck Surgery, Univ. of North Carolina at Chapel Hill, Chapel Hill, NC), and Richard Freyman (Speech, Lang., and Hearing Sci., Univ. of Massachusetts Amherst, Amherst, MA)

Spatial release from masking (SRM) occurs when a signal of interest (target) and an interfering sound (masker) come from different locations in

space. SRM and sound localization are assumed to share common mechanisms, but it is still unclear whether SRM actually *depends* on hearing the target and masker in different places, even in situations dominated by informational masking. The current study used a spatial rhythmic release from masking (RMR) task [J. Middlebrooks and Z. Onsan, *JASA*, 132, 3896–3911, 2012]. The target and masker were trains of non-simultaneous noise bursts; hearing the target temporal pattern relies on segregation from the masker, which is supported by spatial separation. Using a headphone simulation of space (HRTFs), the current study replicated key aspects of Middlebrooks and Onsan's (2012) finding that RMR spatial thresholds were poorer for high-frequency than low-frequency bursts. We investigated whether this low/high frequency difference, and others created with degraded localization cues, were associated with variations in basic localization discrimination tasks using the same RMR stimuli. Data collection is ongoing, but preliminary results suggest a covariation of spatial release in the RMR task and sensitivity to change in spatial angle. [Work supported by NIDCD R01-01625.]

2:05

2pPP5. Front-back bias and the 1000-Hz interaural intensity anomaly. William Hartmann (Michigan State Univ., 749 Beech St., East Lansing, MI 48823, wmh@msu.edu)

Free field measurements were made of the level differences, as measured in an ear canal, for a noise source in front of the head compared to a source in back. The measurements were made on a Kemar manikin for 19 lateral angles over the complete 180-degree range for front-back comparisons. There were 7 octave bands, 125–8000 Hz, and two horizontal planes. Front-back level differences were generally positive except for the dramatic case of 1000 Hz, where the differences were negative over an azimuth range of 120 degrees. When the head-related elevation of the sources was increased from zero to 10 degrees, the range of the negative front-back differences grew to 140 degrees. These physical results are consistent with the strong tendency for listeners to identify a one-third octave noise band in a narrow range near 1000 Hz as a source in back. This tendency may be the origin of the often-observed Grantham effect, wherein the difference limen for the interaural level difference is larger near 1000 Hz than at higher or lower frequencies. The connection is that sources in back of the listener, whatever their frequencies, are not localized nearly as well as sources in front.

2:20

2pPP6. Localization of real and tangent-law panned phantom sound sources in the frontal horizontal plane. Mark A. Stellmack (Psych., Univ. of Minnesota, 75 East River Parkway, Minneapolis, MN 55455, stell006@umn.edu), Stanley Sheft (Dept. of Commun. Disord. and Sci., Rush Univ. Medical Ctr., Chicago, IL), and Ewan A. Macpherson (National Ctr. for Audiol., Western Univ., London, ON, Canada)

Auditory source separations of as little as 1 degree are detectable. However, presenting auditory stimuli at small separations presents technical challenges, with loudspeaker separation limited by transducer diameter. An alternative procedure is to utilize phantom sources, with the perceived position of a single source determined by the relative output levels of two spatially separated loudspeakers. Therefore, it is important to determine whether real and phantom sources can be localized with the same precision. In the present experiment, listeners localized real (individual) sources and phantom sources computed using a tangent-law model giving the same nominal azimuthal angles as the real sources. Listeners used a laser pointer to indicate perceived source location. Infrared cameras detected pointer position with responses stored in terms of azimuth. Signals were broadband or narrowband (300-700 Hz and 3800–4200 Hz) noise, 100 or 500 ms in duration. Generally, phantom sources were localized with less precision than real sources, and high-frequency signals were localized with less precision than broadband or low-frequency signals, with no effect of duration. Results show that phantom sources are localized with sufficient accuracy and precision to be useful in assessing auditory spatial acuity, but they are not localized with the same precision as real sources.

2:35–2:50 Break

2:50

2pPP7. Conversation behavior assessment to estimate communication difficulty. Stefan Klockgether (R&D, Sonova AG, Laubisrütistrasse 28, Stäfa, Zürich 8712, Switzerland, stefan.klockgether@sonova.com), Laurent S. Simon (R&D, Sonova AG, Staefa, ZH, Switzerland), and R. Peter Derleth (R&D, Sonova AG, Stäfa, Zürich, Switzerland)

The communication behavior of humans adapts to the needs of different communication situations. Humans have developed several strategies to successfully communicate in challenging acoustic environments. Some of these strategies can be consciously controlled while in a conversation, but others happen sub-consciously, or as a mixture of both. The applied strategies come with an increased effort, a deviation from common behavior, or the overcoming of personal comfort zones and are usually ceased as soon as the situation allows. This study's aim is to monitor participant's communication behavior in easy and challenging acoustic situations and use this data as a measure for the experienced communication difficulty. The Sonova Real Life Lab allows to investigate natural conversation situations in a controlled acoustic environment. Participants can move around and interact with each other across a 25m^2 stage, while their head positions and orientations are tracked with a motion capturing system. The voices of the participants are recorded using wireless headset microphones. This allows to estimate individual vocal effort and analyze backchanneling and turn taking behavior. This contribution will present and discuss data collected with participants engaged in real one-to-one conversations, while the acoustic background was systematically altered.

3:05

2pPP8. Exploring attentive listening in noise through the just-follow conversation task. William M. Whitmer (Hearing Sci. - Scottish Section, Univ. of Nottingham, Level 3, New Lister Bldg., Glasgow Royal Infirmary, Glasgow G31 2ER, United Kingdom, bill.whitmer@nottingham.ac.uk), David McShefferty (Hearing Sci. - Scottish Section, Univ. of Nottingham, Glasgow, United Kingdom), and Karolina Smeds (ORCA Europe, WS Audiol., Stockholm, Sweden)

Listening to a conversation demands comprehension, attention and prediction. To better understand the impact of these demands, as well as how hearing aids might alleviate them, a conversational listening test should be realistic, repeatable and relatable (i.e., have interpretable units). A test that potentially satisfies these needs is the just-follow conversation (JFC) task, where the listener adjusts the signal level to where they can understand, with effort, the gist of what is being said in a background of noise. Fifty-four participants sat in the centre of a circular loudspeaker array and adjusted the overall level of one monologue, one dialogue, two monologues or two dialogues presented in the front hemifield to where they could just follow the speech four times per trial. Signals were presented in surrounding fixed-level café and same-spectrum noise backgrounds. Bilateral hearing-aid users adjusted aided and unaided; non-users repeated each condition to evaluate reliability. Results showed an increase in JFC for dialogues re monologues. Individual JFC SNRs correlated with SSQ12 speech subscale scores as well as pure-tone threshold averages. JFC reliability was comparable to more objective speech understanding measures, but it may not be suitable to capture perceived conversational benefits for more subtle changes in hearing-aid processing. [Work supported by the UK Medical Research Council Grant No. MR/X003620/1 & WS Audiology.]

3:20

2pPP9. Relationship between objective measures and listener intelligibility of speech processed by source-separation algorithms. Karen L. Payton (Speech Technol. & Appl. Res., 4 Militia Dr., Lexington, MA 02421, kpayton@umassd.edu), Richard Goldhor (Speech Technol. & Appl. Res., Lexington, MA), Behdad Dousti (Commun. Sci. and Disord., Univ. of Cincinnati, Cincinnati, OH), Sarah Dugan (Rehabilitation, Exercise, and Nutrition Sci., Univ. of Cincinnati, Dayton, OH), Suzanne Boyce (Commun. Sci. and Disord., Univ. of Cincinnati, Cincinnati, OH), and Joel MacAuslan (Speech Technol. & Appl. Res., Lexington, MA)

A major determinant of success for separating a speech signal from a noisy environment is the intelligibility of the extracted speech signal. Intelligibility is best measured as the fraction of words correctly recognized by

listeners, but computational measures are often preferred because they are less labor-intensive than collecting listener judgements. We compare listener intelligibility data to three acoustically derived measures: (a) SNRs estimated from the processed mixtures, (b) coherence and (c) speech-based Speech Transmission Index (sSTI). Sentences were recorded against restaurant babble, white Gaussian noise, and nonstationary noise by four microphones at different SNRs ranging from $+4\,\text{dB}$ to $-8\,\text{dB}$. Processing conditions included (1) the original mixture; (2) the mixture processed by a critically determined 4-channel blind source separation (BSS) algorithm; (3) the mixture processed by a 2-channel, underdetermined, BSS algorithm; (4) the residual after subtracting noise estimates determined using a least-mean squared (LMS) algorithm; (5) an estimate of speech extracted using an LMS algorithm to remove the two noises and then 2-channel BSS to separate the sentences from the babble; and (6) pristine speech recorded with no noise. The computational measures are compared to gold-standard intelligibility results from listening tests.

TUESDAY AFTERNOON, 14 MAY 2024 ROOM 206, 1:00 P.M. TO 4:10 P.M.

Session 2pSA

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

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

Alexey Titovich, Cochair
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Abigail D. Willson, Cochair
Acoustics, Penn State University, PO Box 30, Mail Stop 3220B, State College, PA 16804

Chair's Introduction—1:00

Invited Paper

1:05

2pSA1. Higher-order Non-Hermitian skin effect in an acoustic lattice. Yun Jing (Graduate Program in Acoust., Pennsylvania, 201 Appl. Sci. Bldg., State College, PA 16802, jing.yun@psu.edu), Jiaxin Zhong (Graduate Program in Acoust., Pennsylvania, State College, PA), Wladimir Benalcazar (Phys., Emory Univ., Atlanta, GA), Kevin Kim (Graduate Program in Acoust., Pennsylvania, University Park, PA), and Mourad Oudich (Graduate Program in Acoust., Pennsylvania, State College, PA)

Non-Hermitian physical systems offer a distinct band topology that gives rise to the interesting phenomenon known as the non-Hermitian skin effect (NHSE). However, the exploration of higher-order NHSE in classical wave systems by leveraging non-reciprocity has been largely unexplored. In this study, we introduce a novel approach to experimentally realize the higher-order NHSE in a two-dimensional non-reciprocal breathing acoustic Kagome lattice. This lattice is composed of acoustic cavities with non-reciprocal coupling achieved through electrically controlled active acoustic elements. We successfully observed second-order corner modes using a 3×3 lattice, which manifest as localized pressure distributions at specific frequencies in both parallelogram and triangular topological acoustic lattices. We delve into the underlying mechanisms and their implications, thereby paving the way for a deeper understanding of higher-order NHSE and potentially innovative applications rooted in the acoustic non-reciprocity.

1:25

2pSA2. Breathers for nonlinear energy management in acoustic waveguides. Mohammad A. Bukhari (Mech. Eng., Wayne State Univ., 5050 Anthony Wayne Dr., 2140, Detroit, MI 48202, bukhari@wayne.edu), Oumar Barry (Mech. Eng., Virginia Tech, Blacksburg, VA), and Alexander F. Vakakis (Mech. Sci. and Eng., Univ. of Illinois Urbana-Champaign, Urbana, IL)

Recent investigations of nonlinear metamaterials have revealed interesting wave propagation phenomena with no counterpart in linear systems. One of the appealing phenomena is traveling discrete breathers. Breathers are traveling oscillatory wavepackets with spatially localized envelopes and energy-dependent speeds. These nonlinear waves were reported in lattices with purely (or nearly purely) nonlinear neighbor interactions (possessing zero speed of sound as defined in the classical acoustics sense, and, hence, designated as sonic vacua). However, the effect of local resonators on breather propagation is elusive to date. In this work, we aim to fill this gap by studying 1D grounded strongly nonlinear lattices with linear local resonators (SNLRMs) under impulsive force excitation. Numerical simulations have demonstrated that SNLRMs support the generation of new families of breathers. These families resulted from the opening of a new optical propagation zone (PZ) in the spectrum, which is associated with the linear dynamics of the local resonators. These families are characterized by the local resonator parameters and can support multiple fast frequencies or a single frequency that belongs to either the optical or acoustical PZs. Further aspects of the nonlinear acoustics are also demonstrated using the complexification averaging method. The newly reported breathers open the venue to employ SNLRMs in passive nonlinear energy management applications. Examples are presented through acoustics waveguides and frequency filtering.

1:40

2pSA3. Using a leaky-wave antenna as an acoustic prism for speech-separation. Abigail D. Willson (Appl. Res. Lab., Penn State Univ., PO Box 30, M.S. 3220B, State College, PA 16804, adw5@psu.edu), Andrew S. Wixom, and Amanda Hanford (Appl. Res. Lab., Penn State Univ., State College, PA)

Leaky-wave antennas (LWA) are an acoustic metamaterial comprised of an array of unit cells. For a 1D line array, the output ranges from 0° to 180° relative to the direction of the axis of the array. The direction of the output is dependent on the frequency of the input; a single frequency input will produce a highly directive output beam at a specific angle. Much like an optic prism, which can be used to split a complex optic wave into its components, a LWA can be used to split a complex input audio wave into its components. This work explores the capabilities of LWA to separate transient signals like human speech into different output directions. This is accomplished by way of an equivalent circuit model of the LWA that enables simulations in both the time and frequency domains. The frequency domain performance of the model is validated against previously published results, and then the transient response is shown to recover these as well once steady-state is reached

1:55

2pSA4. Vibroacoustic metamaterial for enhanced Sound Transmission Loss with variable frequency range designed for serial production. Klara Chojnacka (AGH Univ. of Sci. and Technol., Mickiewicza 30, Cracow 30-059, Poland, klara.chojnacka@agh.edu.pl), Aleksander Kras (Silencions Sp. z o.o., Wroclaw, Poland), and Tadeusz Kamisinski (AGH Univ. of Sci. and Technol., Cracow, Poland)

Recently, vibroacoustic metamaterials have been broadly investigated, especially for noise and vibration mitigation. By creating a band gap effect in flexural wave propagation in a base element, metamaterials improve its vibroacoustic parameters in low and mid frequency range while preserving low mass and structure dimensions. While the foundational assumptions and lab validations highlight the advantages of these metamaterials, the practical application of these structures in real-world scenarios remains a challenge. The main aspect that is widely investigated is achieving maximum effectiveness while maintaining the simplicity of geometry to optimize mass production costs. This work presents a vibroacoustic metamaterial design with geometry adapted to serial production using injection molding. The proposed geometry allows for adjusting the effective frequency range of the structure after the production process. Due to the distinctive configuration of the elements and the grouping of unit cells, the design facilitates the generation of broadband and multi-band structures as well. Numerical simulations were employed to assess the impact of the proposed metamaterial on Sound Transmission Loss and other vibroacoustic parameters. Experimental validation of prototype effectiveness was conducted through Sound Reduction Index measurements in a diffuse field.

2:10

2pSA5. General acoustic Willis media modeled as conventional materials with embedded sources. Mehmet U. Demir (Mech. Eng., Univ. of Michigan, G.G. Brown Lab., 2350 Hayward, Ann Arbor, MI 48109, udemir@umich.edu) and Bogdan-Ioan Popa (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

Willis coupling vectors provide additional degrees of freedom to manipulate acoustic waves unavailable in conventional materials, but the additional material properties make it difficult to simulate the interaction between sound and Willis media especially when inhomogeneous media with arbitrary geometries are involved. To address this challenge, we establish the equivalence between the wave equation inside a three-dimensional, inhomogeneous, and anisotropic Willis medium to the well-known acoustic wave equation in conventional materials with embedded continuous distributions of monopole and dipole sources. We validate the equivalence in numerical simulations showing that conventional materials with embedded sources replace physical Willis metamaterials anisotropic in the Willis and mass density tensors without modifying the scattered sound distribution. Moreover, the equivalence offers insights into the physics of Willis materials. For example, it shows that various combinations of Willis coupling vectors can generate identical sound scattering for any excitation. It also shows whether effective material parameters extracted from single Willis cell simulations remain valid in bulk metamaterials obtained by replicating periodically that cell. These results suggest that the equivalence model can advance the design of Willis metamaterials and provide a tool to better understand the physics of Willis media.

2:25–2:40 Break

2:40

2pSA6. Study on vibration and wave propagation characteristics of thin-walled metamaterial structure with embedded hexagonal Acoustic black hole lattice. Chinna B. Singepogu (Aerosp. Eng., Indian Inst. of Sci. Bangalore, Indian Inst. of Sci. Bangalore, Bangalore 560012, India, singepogub@iisc.ac.in), Dinesh K. Harursampath (Aerosp. Eng., Indian Inst. of Sci. Bangalore, Bangalore, India), and Ramesh G. Burela (Shiv Nadar Univ., Noida, India)

An acoustic black hole is an inhomogeneity of structure that can absorb the total incident wave energy by trapping the wave and reducing the group velocity of the propagating wave. ABH can be achieved by varying the thickness to zero, which is impossible, so the power law profile is the ideal thickness variation. The dispersion characteristics of hexagonal acoustic black hole lattices were studied using band gap analysis for the irreducible Brillouin zone. The study also used the geometric nonlinearity for ABH. Later, modal analysis of the plate with hexagonal black hole arrangement is done for harmonic excitation cases for various boundary conditions. All the numerical finite element simulations for the above analysis are done using commercially available software, COMSOL Multiphysics. Keywords: Acoustic black hole, Irreducible Brillouin zone, Modal analysis, COMSOL Multiphysics.

2:55

2pSA7. Numerical extraction of three-dimensional acoustic polarizabilities for acoustically small scatterers. A. J. Lawrence (Walker Dept. of Mech. Eng. and Appl. Res. Labs., Univ. of Texas at Austin, 204 E. Dean Keeton St., Stop C2200, Austin, TX 78712-1591, ajlawrence@utexas.edu), Samuel P. Wallen, and Michael R. Haberman (Walker Dept. of Mech. Eng. and Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX)

Polarizability is a convenient descriptor for scattering from low-*ka* inhomogeneities in both electromagnetism, where it originated, and in acoustics, where it is increasingly used in the design of metamaterial elements. In two or three dimensions, the polarizability is represented by a block matrix that couples the local pressure and particle velocity fields to the lowest-order components of the multipole expansion, usually truncated to dipole order. A fully dense acoustic polarizability matrix has components that couple spatially uniform pressure and velocity oscillations to dipole and monopole scattering, respectively. This unique coupling leads to acoustic bianisotropy, also known as Willis coupling, for a collection of scatterers. The design of Willis materials necessitates computationally efficient methods to extract all components of the polarizability matrix and, at present, only two-dimensional analytical extraction methods are found in the literature. In this work, we present an algorithm to extract all components of the three-dimensional polarizability matrix from a scatterer with arbitrary geometry and composition. The method presented here is numerically efficient and yields results that are in agreement with an analytically obtained benchmark polarizability, providing a useful tool for acoustic metamaterial design.

3:10

2pSA8. Abstract withdrawn.

3:25

2pSA9. Design and analysis of a compact sound absorber made of multiple parallel Helmholtz resonators. Zacharie Laly (Dept. of Mech. Eng., Sherbrooke Univ., 2500, boul. de l'Université, Sherbrooke, QC J1K 2R1, Canada, zacharie.laly@usherbrooke.ca), Chris Mechefske (Dept. of Mech. and Mater. Eng., Queen's Univ., Kingston, ON, Canada), Sebastian Ghinet, and Tenon Charly Kone (Aerosp., National Res. Council Canada, Ottawa, ON, Canada)

In this paper, a compact sound absorber material is proposed and studied for noise attenuation at multiple frequencies using finite element method. The global cylindrical cavity of the material is partitioned into twelve sub-cavities where four are located in the center and one extended neck is connected to each sub-cavity. Each sub-cavity with the associated neck represents a Helmholtz resonator and thus the proposed material design is made of twelve parallel Helmholtz resonators. The sound absorption coefficient and the transmission loss present twelve resonant peaks at different frequencies where the surface impedance is close to the air impedance. The resonant frequencies can be adjusted by the geometrical parameters of the necks and the sub-cavities volumes. The proposed sound absorber can be used in multiple engineering applications to attenuate noise simultaneously at twelve different frequencies.

3:40

2pSA10. Sound absorption analysis of a metamaterial based on parallel dual Helmholtz resonators. Zacharie Laly (Dept. of Mech. Eng., Sherbrooke Univ., 2500, boul. de l'Université, Sherbrooke, QC J1K 2R1, Canada, zacharie.laly@usherbrooke.ca), Chris Mechefske (Dept. of Mech. and Mater. Eng., Queen's Univ., Kingston, ON, Canada), Sebastian Ghinet, and Tenon Charly Kone (Aerosp., National Res. Council Canada, Ottawa, ON, Canada)

In this paper, a sound absorbing material consisting of four parallel dual Helmholtz resonators is proposed and its sound absorption coefficient is studied using finite element method. The dual Helmholtz resonator is made of neck-cavity-neck-cavity where each neck extends into each cavity. The sound absorption coefficient of the dual resonator presents two resonant peaks. It is demonstrated that when the radius of the first or the second neck increases, the two resonant frequencies of the sound absorption increase while they decrease when the length of the first or the second neck increases. The proposed material design, which combines four parallel dual Helmholtz resonators, presents eight sound absorption peaks and these eight resonant frequencies can be tuned to specific frequencies by adjusting the parameters of the necks. It is a compact sound absorber, which can help to attenuate the noise simultaneously at eight different frequencies in several engineering applications.

3:55

2pSA11. Acoustic properties of natural fiber reinforced composite micro-perforated panel designed using 3D printing. Umberto Berardi (Toronto Metropolitan Univ., 350 Victoria St., Toronto, ON M5B 2K3, Canada, uberardi@ryerson.ca)

The present study investigated the acoustic performance of biodegradable MPP absorbers made of natural fiber-reinforced composites (NFRC) using 3D printing. The novelty of this current research lies in the recent development of a methodology that aids industry professionals in optimizing the production of MPP (Micro Perforated Panel) at a competitive cost. This is achieved by addressing and eliminating issues commonly faced in traditional manufacturing processes, such as manual preparation and pressing. The FDM technique was used to fabricate test samples utilizing the PLA/corkwood composite. Using an impedance tube device with two microphones, the acoustic absorption coefficients of MPPs with different perforation diameters, thicknesses, and perforation rates were measured. Maa's analytical model was used to predict the acoustic absorption performance. Moreover, considering the average sound absorption and total cost of fabricating the samples, RSM-CCD was employed to optimize these samples. In the end, the parallel arrangement of MPP double layer and the combination of MPP with kenaf porous material were tested to improve the sound absorption performance. The results showed that the average sound absorption coefficient of the NFRC-MPP sound absorber is 25% more than that of conventional MPP sound absorbers.

Session 2pSCa

Speech Communication: VowelFest: Honoring the Past and Celebrating the Present II

Ewa Jacewicz, Chair
Speech and Hearing Science, The Ohio State University, 1070 Carmack Road, 110 Pressey Hall, Columbus, OH 43210

Invited Papers

1:00

2pSCa1. Acoustic vowel distances within and across dialects. Cynthia G. Clopper (Ohio State Univ., 1712 Neil Ave., Columbus, OH 43210, clopper.1@osu.edu)

Regional dialects of American English exhibit variation in the phonetic realization of vowel categories. This variation leads to different degrees of phonetic vowel similarity within and across dialects. This vowel similarity has been captured by measures of formant distance and overlap in the acoustic space, typically with static formant estimates from a single timepoint in the vowel. In contrast, dialect similarity has been assessed using measures of acoustic distance over time. In the current study, dynamic measures of acoustic distance, including dynamic time warping, root mean-square distance, and generalized additive mixed effect models, were used to assess within-dialect vowel category distances and to compare these distances across dialects. The results were generally consistent across the different distance metrics, confirming their respective ability to capture acoustic vowel distance. However, the magnitude of the distances varied across dialects and vowel variables. Whereas some dialect-specific variables showed the expected patterns of within- and across-dialect similarity based on previous descriptive work, other variables did not. These mixed findings have implications for our understanding of regional dialect variation in American English, its effect on acoustic vowel similarity, and the relationship between acoustic vowel similarity and perceptual ambiguity of vowel contrasts within and across dialects.

1:20

2pSCa2. Regional vowel patterns as shown by discrete cosine transforms. Erik R. Thomas (English, North Carolina State Univ., Dept. of English, Box 8105, Raleigh, NC 27696-8105, erthomas@ncsu.edu), Jeff Mielke (English, North Carolina State Univ., Raleigh, NC), and Kirk A. Hazen (CVS Health, Warrenton, NC)

This project presents a new approach to analyzing geographical patterning in vowel variation that combines discrete cosine transforms (DCTs) with cluster analysis. It offers a means of reducing bias from analyses of geographical patterning in vowel variation. DCT0 and DCT1 capture the overall position of vowels in the vowel envelope and DCT2 adds information about curvature. To mitigate anomalies, these metrics are based on numerous tokens and measurement points. Cluster analysis can then be applied to the DCT data to indicate which speakers are most similar without predetermined groupings. The procedure suggests how vowel realizations are correlated with geographical divisions, if at all, within the area covered by a dialect survey. Here, we apply DCTs and hierarchical clustering to a corpus of speakers born 1970 or later and covering eastern Ohio, West Virginia, and western North Carolina. The results align only partially with isophones from earlier dialect surveys. Analyses of individual vowel phonemes typically exhibit considerable intermixture of forms; clearer geographic patterns emerge primarily when multiple vowels in named chain shifts are considered together. Recent dialect leveling appears to play a role in the paucity of distinguishable regional patterns.

Contributed Papers

1:40

2pSCa3. Neurophysiological correlates of the perceptual magnet effect in speech perception. Yang Zhang (Speech-Language-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr SE, Minneapolis, MN 55455, zhang470@umn.edu)

This study investigated the Perceptual Magnet Effect (PME) in speech perception using behavioral and Event-Related Potentials (ERP) measures. There were two primary research questions: (1) whether category goodness rating influences speech discrimination with reduced sensitivity near the prototype and increased sensitivity in the vicinity of a poor exemplar, (2) whether the PME is domain-specific to speech sounds. Twenty adult native English speakers participated in identification, goodness rating, and discrimination tasks. The stimuli were synthesized vowels for /a/ and their nonspeech analogs by systematically varying the first two formants. ERP oddball conditions included four conditions presented in a counter-balanced order with the prototypical /a/ as the standard stimuli and its variants as deviants, and a non-prototypical /a/ as the standard and its variants as deviants, as well as the nonspeech matches. Results indicated that participants rated vowel category goodness based on the F2/F1 ratio for the /a/ sounds. Mismatch negativity amplitudes in the two speech conditions aligned with PME predictions, but similar patterns were also observed in the nonspeech conditions. The findings collectively demonstrated neurophysiological evidence for the perceptual organization of within-category variations in line with PME, which may transfer to auditory processing of similar acoustic patterns in nonspeech.

1:55

2pSCa4. Implicit and explicit responses to infant sounds: A cross-sectional study among parents and non-parents. M. Fernanda Alonso Arteche (School of Commun. Sci. & Disord., McGill Univ., 1295 Rue des Carrieres, apt 402, Montreal, QC H2S 0E1, Canada, maria.alonsoarteche@mail.mcgill.ca), Leatisha Ramloll (School of Commun. Sci. & Disord., McGill Univ., Montreal, QC, Canada), Lucie MENARD (Linguist, UQAM, Montréal, QC, Canada), and Linda Polka (School of Commun. Sci. & Disord., McGill Univ., Montreal, QC, Canada)

This research investigates the infant schema in auditory perception by examining how different demographics, including males, females, parents, and non-parents, respond implicitly and explicitly to baby vocalizations in comparison to adult, cat, and kitten sounds. Utilizing a single category implicit association task (SC-IAT) and a detailed questionnaire, we analyzed participants' responses to synthesized vowel sounds from infants, adults, cats, and kittens. The questionnaire focused on participants' liking and perception of cuteness for these sounds. Findings reveal a universal positive implicit preference for baby vocalizations across all groups ($p = 0.01$), without a similar effect for other sound sources. In contrast, explicit responses varied significantly. While all groups showed a preference for the sounds of cats, babies, and kittens over adults, only mothers demonstrated a statistically significant explicit preference for infant sounds over those of cats and kittens. This study highlights the discrepancy between unconscious and conscious attitudes towards infant sounds. It underscores that while an implicit affinity for baby vocalizations is widespread, explicit preferences, particularly in terms of cuteness and likeability, are markedly stronger in mothers. These insights contribute to our understanding of the auditory dimension of the infant schema, emphasizing the role of gender and parental status in shaping responses.

TUESDAY AFTERNOON, 14 MAY 2024 ROOM 214, 2:30 P.M. TO 5:30 P.M.

Session 2pSCb

Speech Communication: VowelFest III (Poster Session)

Ewa Jacewicz, Chair
Speech and Hearing Science, The Ohio State University, 1070 Carmack Road, 110 Pressey Hall, Columbus, OH 43210

All posters will be on display from 2:30 p.m. to 5:30 p.m. Authors of odd-numbered papers will be at their posters from 2:30 p.m. to 4:00 p.m. and authors of even-numbered papers will be at their posters from 4:00 p.m. to 5:30 p.m.

Contributed Papers

2pSCb1. Losing ground?: Towards a description of palatal lateral variation in Breton. Kaitlyn Owens (Indiana Univ. Bloomington, 355 North Eagleson Ave., GA 3151, Bloomington, IN 47405, kaitowen@iu.edu)

Traditional Breton descriptions and grammars attest that [j] is an often-pronounced variant of /ʎ/ (Hemon, 1995), however, little is known about what factors influence this phoneme's realization. This study aims to elucidate what linguistic and social factors affect Breton /ʎ/ pronunciations. Given that Breton new speakers are often militant in promoting Breton revitalization (Jones, 1995) and their pronunciations, unlike those of traditional speakers, are strongly influenced by French (Hornsby, 2015), we predicted new speakers will produce [j] more than traditional speakers. We elicited 125 tokens that contain /ʎ/ from eight Breton-speaking participants in a wordlist task—the only categorical [j]-producer was the sole traditional Breton speaker to participate. Focusing on new speakers, we use mixed-effects regression and find older speakers produce [ʎ] more often than younger speakers ($p < 0.0001$), and gender is not a significant predictor. For linguistic factors, word position does not play a significant role, but [j] is more frequent in intervocalic tokens when the following vowel is palatal ($p = 0.0249$). Although /ʎ/ variation does not exhibit effects of prestige, our results suggest a potential change in progress whereby [ʎ] is being lost in favor of [j] and is particularly gaining momentum intervocalically when the following vowel is palatal.

2pSCb2. Acoustic variability underlying pathological voice quality. Jody Kreiman (Head and Neck Surgery, UCLA, 1000 Veteran Ave., 31-19 Rehab Ctr., Los Angeles, CA 90095-1794, jkreiman@ucla.edu)

Studies have shown that a few acoustic parameters (variability in spectral energy in the higher frequencies, plus formant dispersion) account for the most variability in normal human voices, regardless of sex, age, or language spoken. These parameters also characterize vocal variation in other animals and carry biologically important information about the speaker. They further allow us to conceive of voices as existing in a simple, low-dimensional "quality space" that could facilitate quick judgments of similarity and difference. How well does this model fit pathological voices, given that one hallmark of voice disorders is increased variability? In a preliminary study of sustained vowel phonation in pathologic voices [*J. Acoust. Soc. Am.* 150, A191] we found that the variability introduced by pathology did not replace or obscure universal factors. Instead, measures of variability and instability carried greater significance for the quality of pathologic voices compared to normal voices. Expanding on this finding, this paper employs principal component analysis to derive the factors characterizing vocal variability from acoustic profiles of read speech for pathological voices. Results may provide insights into treatment strategies aimed at reducing the perceived pathology in disordered voices.

2pSCb3. Revisiting the sociophonetics of sexuality in spontaneous speech. Amber Galvano (Linguist, UC Berkeley, Dwinelle Hall #2650, Berkeley, CA 94704, amber_galvano@berkeley.edu)

This study surveys sociophonetic variation in sibilant and vowel production for 44 Bay Area English speakers. I focus on (i) whether sexual orientation is an independent predictor, or interacts with gender, and (ii) what the patterning of bi + (bisexual, pansexual, queer, etc.) and non-binary speakers adds to previous claims (e.g., Gaudio 1994, Pierrehumbert 2004, Willis 2023). All participants completed a Map Task in pairs; 37 participated in a sociolinguistic interview. Recordings were auto-transcribed, forced-aligned, and hand-checked. For /s ʃ z ʒ/, tokens under 0.05 sec, in /str/ clusters, and/or adjacent to another sibilant were excluded. Before getting COG and skew, 0.02 sec were subtracted from token edges and voicing was Hann band-filtered. For all vowels and sibilants, mean and midpoint f0 were taken, and F1-F4 for vowels (using the median of values at midpoint plus six surrounding timepoints). Results show sexual orientation alone is not a robust predictor of /s/ frontedness. Both gender and sexuality interacted with duration for /s/ frontedness and vowel f0; gender and sexuality also interacted to influence /ʃ/ frontedness and /z/ devoicing, as well as vowel space shape. These results imply that, when bi + and non-binary identities are incorporated, the phonetic indexation of sexual identity is not easily generalizable within orientation labels, at least for informal speech.

2pSCb4. Speech adaptation in conversation: Effects of segment cue-weighting strategy for non-native speakers. Han Zhang (Dept. of Linguist, Simon Fraser Univ., 8888 University Dr., Burnaby, BC V5A 1S6, Canada, han_zhang7@sfu.ca), Morgan Glover, Fenqi Wang, Xizi Deng (Dept. of Linguist, Simon Fraser Univ., Burnaby, BC, Canada), Dawn Behne (Dept. of Psych., Norwegian Univ. of Sci. and Technol., Trondheim, Norway), Allard Jongman (Dept. of Linguist, Univ. of Kansas, Lawrence, KS), Joan Sereno (Dept. of Linguist, Univ. of Kansas, Kansas City, KS), and Yue Wang (Dept. of Linguist, Simon Fraser Univ., Burnaby, BC, Canada)

Research has investigated how speakers make phonetic adaptations by adjusting their speech sounds to either converge to or diverge from their interlocutor's. Less is known about the dynamic adaptive strategies non-native speakers use to improve intelligibility when interacting with native speakers, particularly how their strategies change over the course of spontaneous conversations. The current study examines phonetic adaptations in English words contrasting tense and lax vowels (e.g., sheep-ship) in unscripted conversations between non-native Japanese English and native English speakers during an engaging computer game task. Japanese speakers have previously been found to rely on temporal cues (vowel length) for tensity distinctions due to their L1 cue-weighting strategy rather than spectral cues (vowel quality) that are predominantly used in English. Acoustic analyses are conducted to examine changes in non-native vowel productions before, during and after the conversation task. We predict that, in an attempt to overcome miscommunication, non-native speakers may initially lengthen the tense vowels to distinguish them from their lax counterparts. As the conversation progresses, non-native speakers may adapt to a more native-like cue-weighting pattern by shifting to spectral distinctions. Results are discussed in terms of adaptations to cue-weighting strategies for intelligibility gains by interlocutors of different linguistic backgrounds.

2pSCb5. Examining the maintenance of dialect features in colloquial urban Armenian speech via variationist analysis of vowels in Gavar̦, Armenia. Emma L. Portugal (Linguist, Univ. of Michigan, Lorch Hall, 611 Tappan Ave., Ann Arbor, MI 48109, esantelm@umich.edu)

This study examines the maintenance of previously described local dialect vowels in the regional city of Gavar̦, Armenia. The dialect vowel system contains additional phonemic and allophonic distinctions not found in supralocal Armenian varieties. As such, adherence to this system was conceptualized through the presence or absence of various vowel mergers. F1 and F2 were measured from picture and word list data collected from sociolinguistic interviews with 31 participants. Mergers were assessed for each participant via Pillai scores and Euclidean distances between clusters of words where different vowels were predicted to appear based on previous descriptions of the dialect. Countering previous claims about dialect leveling in regional cities, some participants were found to maintain some aspects of the dialect vowel system. Fixed effects linear models assessed the effect of demographic predictors (self-reported gender, birth year, education level) on maintenance of dialect vowels. There was an effect of gender, such that men were found to maintain some dialect vowels to a greater degree than women. Participants' commentary suggested that the most salient aspects of the dialect vowel system are the same ones in relation to which this demographic variation was uncovered, indicating a relationship between salience and maintenance of dialect vowels.

2pSCb6. Effects of formant peak flattening on vowel perception: A larger-scale web-based experiment. Filip Nenadić (Dept. of Linguist, Univ. of Potsdam, 3-28 Assiniboia Hall, Edmonton, AB T6G 2E7, Canada, nenadic@ualberta.ca), Dejan Sredojević (Dept. of Serbian Lang. and Linguist, Faculty of Philosophy, Univ. of Novi Sad, Novi Sad, Serbia), and Michael Kiefte (School of Commun. Sci. and Disord., Faculty of Health, Dalhousie Univ., Halifax, NS, Canada)

Ito *et al*. reported that suppressing the first or the second formant in Japanese synthesized vowels did not importantly change listener perception. However, quantitative analyses of data collected in the English (Nenadić *et al*., 2020) and the Serbian language (Nenadić *et al*., 2023) have shown that formant suppression increases response entropy between participants (i.e., decreases agreement in vowel identity). Previous studies never tested more than fifteen participants, but it was noted that certain listeners gave different responses to a manipulated synthesized vowel in comparison to its original version more often than others. We tested 118 native monolingual speakers of Serbian language (87% female, 13% male; age 18 to 44, $M = 21.07$, $SD = 4.71$) in an online replication of the experiment. The results again show that listeners agree less about vowel identity for stimuli with suppressed formants. Certain listeners again tended to change their response to the manipulated version of the vowel more often than others. Importantly, this tendency did not correlate with their mean response latency. We discuss possible causes for these findings, including differences in listener strategies and time requirements of a more careful processing of the incoming signal.

2pSCb7. Creaky voice across language and gender: A study of Canadian English-French bilingual speech. Jeanne Brown (Linguist, McGill Univ., 1085 Ave. du Docteur-Penfield #104, Montreal, QC H3A 1A7, Canada, jeanne.brown@mail.mcgill.ca) and Morgan Sonderegger (Linguist, McGill Univ., Montreal, QC, Canada)

This study addresses how non-contrastive creaky voice varies among languages and across speakers (as a function of gender). Spontaneous speech from 9 English-French bilingual speakers born and raised in Ontario/Québec was collected from publicly available online data sources, amounting to roughly 5 min of speech per speaker-language pair and 13 992 vowels total. This corpus will reach 40 speakers by the conference. Acoustic analysis consisted of pitch tracking in Praat, providing a proportion of unreliable f0 tracks for each vowel as well as one spectral slope measure (H1*-H2*) and two Harmonics-to-Noise Ratios (CPP and HNR05) as acoustic correlates of creaky voice. Statistical significance was tested using mixed-effects regression models, with fixed effects of language, gender, and utterance position, and maximal by-word and by-speaker random effects. The main results for gender show that men's vowels have more unreliable f0 tracks, lower H1*-H2*, lower HNR05 and somewhat lower CPP, suggesting that male speakers are creakier overall. Regarding language, English displays more unreliable pitch tracking compared to French, providing some evidence for language-dependent vocal settings. Other acoustic correlates of creak, however, do not show consistent cross-linguistic differences.

2pSCb8. Metrical segmentation across dialects. Natasha Warner (Linguist, Univ. of Arizona, Dept. of Linguist, Box 210025, Tucson, AZ 85721-0025, nwarner@email.arizona.edu), Ki Woong Moon (Linguist., Univ. of Arizona, Tucson, AZ), Seongjin Park (Linguist., Univ. of Arizona, Princeton, NJ), James M. McQueen (Donders Inst. for Brain, Cognition and Behavior, Radboud Univ., Nijmegen, Netherlands), and Mohammed K. Albusairi (Linguist., Univ. of Arizona, Tucson, AZ)

Norris, McQueen & Cutler (1995) tested the Metrical Segmentation Strategy (MSS; Cutler & Norris, 1988) as part of the spoken-word recognition model Shortlist, using British English stimuli and listeners. We replicate their study using American English listeners, who we exposed to one of two sets of stimuli. One group heard a new set of stimuli recorded in American English, while the other was exposed to the original British English recordings. Norris *et al*. used a word-spotting task: listeners had to spot words within speech (e.g. "stamp" in [stæmpɪdʒ]). Target words were CVCC (like "champ") or CVC (like "done"), and were followed by a full vowel (e.g. /tʃæmpoʊʃ/) or a reduced vowel (e.g. /tʃæmpəʃ/). The original study found different behavior for CVCC versus CVC targets, with the results suggesting that listeners hypothesize a word onset at the start of a full-vowel strong syllable (the MSS). The results for the current study with American stimuli partially replicate the original findings, showing even more consistent support for the MSS. The results with British stimuli also support the MSS, but with higher error rates. The results indicate that the MSS has a very strong effect even in the difficult setting of cross-dialectal perception.

2pSCb9. An acoustic analysis of French contrast-specific clear-speech productions. Morgan Robertson (Linguist, Univ. of Kansas, 1541 Lilac Ln., Lawrence, KS 66045, morgan.robertson@ku.edu) and Allard Jongman (Linguist, Univ. of Kansas, Lawrence, KS)

Clear speech provides insight into acoustic correlates speakers alter when attempting to increase intelligibility or enhance phonemic contrasts. The present study consists of an acoustic analysis of clearly spoken French voiced and voiceless stops, nasal and oral vowels, and front rounded and unrounded vowels. Productions were elicited using a simulated interaction between French-speaking participants and a computer program, in which participants produced casual tokens and the program 'guessed' with controlled responses. When the program incorrectly responded with "???" (indicating 'What did you say?') or a specific competitor (e.g., response *reins* [ʁɛ̃] 'kidney' to target *reine* [ʁɛn] 'queen'), participants reproduced the word more clearly. The program's responses probed at differences in speakers' productions when responding to specific competitors versus globally attempting to increase intelligibility. Preliminary results show a general increase in prevoicing for clearly produced voiced stops. Speakers also adjust coarticulatory nasality to disambiguate CVN targets from CṼ competitors. Finally, expansion is exhibited between front rounded and unrounded vowel pairs in the F2/F3 vowel space from casual to clear; however, it is largest when the program's response is "???". Overall, findings suggest that speakers modify their speech differently in response to a specific competitor versus the general "???" response.

2pSCb10. The correlation between acoustic and articulatory variation in Laurentian French high vowels. Beth MacLeod (School of Linguist & Lang. Studies, Carleton Univ., 1125 Colonel By Dr., Ottawa, ON K1S 5B6, Canada, Beth.Macleod@carleton.ca), Suzy Ahn, and Phillip Burness (Dept. of Linguist., Univ. of Ottawa, Ottawa, ON, Canada)

This study explores the relationship between acoustic and articulatory variation in Laurentian French (LF) high vowels. LF /i/, /y/, /u/ undergo laxing before a consonant other than a voiced fricative. While several studies have characterized LF vowel laxing acoustically, limited work has described it articulatorily. This study investigates the alignment between ultrasound tongue imaging data and acoustic realization in LF vowels. Using data from Burness *et al*. (2022b), seven native LF speakers participated in a study collecting ultrasound tongue imaging and audio while producing 52 French words with target vowels /i/, /y/, /u/. We chose the ultrasound frame for each token that aligns with the vowel midpoint. From these, we obtained x (tongue backness) and y (tongue height) values. Linear mixed-effect models were employed to assess the relationship between acoustic parameters (F1, F2) and articulatory measures. Findings reveal a varied but generally related pattern between acoustics and articulation. F1 relates significantly to tongue height for /u/ and /y/, while F2 aligns significantly with tongue backness across all three vowels. However, substantial unaccounted variation suggests factors like vocal tract physiology, the non-linearity between articulation and acoustics, and specific choice of articulatory measures might contribute to this variability. Future research will explore these factors to better comprehend this intricate link between acoustics and articulation.

2pSCb11. Stories and words online regional dialect corpus collection. Kevin D. Lilley, Jory P. Ross, Marie Bissell (Ohio State Univ., Columbus, OH), and Cynthia G. Clopper (Ohio State Univ., 1712 Neil Ave., Columbus, OH 43210, clopper.1@osu.edu)

Online data collection allows for access to diverse populations. In the current study, we used online recruitment and data collection methods to obtain a corpus of read short stories, real English words, and nonwords from adult talkers representing three authentic regional dialects of American English and one novel accent. The authentic dialects are New England, Northern, and Southern American English and are each represented by 8–10 talkers, ranging in age from 22 to 75 years old. The novel accent was produced by five Spanish-English bilinguals with training in linguistics, who were asked to produce Spanish /o/ in an otherwise English segmental context. The four target varieties each contain one vowel pair of interest, in which the vowels within the pair are relatively more ambiguous than in the other varieties. Each talker produced one familiar short story (e.g., Goldilocks and the Three Bears) with 40 tokens of each vowel within the target pair for their dialect, as well as a set of real words and nonwords that represent both the target vowel pair for their dialect and the other three vowel pairs for comparison across dialects. All corpus materials are available to the scholarly community.

2pSCb12. Harmony in transition: Exploring the perception-production relationship in sound change. Felix Kpogo (Dept. of Linguist, Boston Univ., 621 Commonwealth Ave., Rm. 116, Boston, MA 02215, fkpog001@bu.edu)

This study investigates the perception-production dynamics of [æ] and [e] in an ongoing merger in Asante Twi's Advanced Tongue Root (ATR) harmony system. Traditional descriptions state that [-ATR] /a/ is realized as [+ATR] [æ] before [+ATR] vowels /i, u, o/. Yet, recent acoustic evidence points to an ongoing sound change in different Twi-speaking communities: in contrast to traditional (rural/suburban) Twi speakers, urban speakers raise and merge [æ] with phonemic /e/ before /i, u/, a change that is more advanced in men and younger speakers (Kpogo, 2023). The present study aims to ascertain whether merger in production correlates with perceptual merger among the same participants in Kpogo (2023). Data from a forced-choice identification task suggests that the presence of a perceptual merger is associated with the participant's locality. Traditional participants consistently identified [æ] and [e] correctly, suggesting separate mental representations for these vowels. Conversely, urban participants often perceived [æ] as [e], suggesting less distinct representations influenced by their own production. Concerning the perception-production relationship, evidence of a link was observed across localities but, crucially, not within each locality. This study contributes to theoretical discussions on the perception-production relationship during sound change, hinting at a co-evolving link between the two modalities.

2pSCb13. Vowel space expansion following alcohol intoxication. Arian Shamei (Linguist, UBC, 2613 West Mal, Vancouver, BC V6T 1Z4, Canada, arianshamei@gmail.com), Xinglei Liu, Rima Seiilova (Tenvos Res. Labs, Sacramento, CA), and Bryan Gick (Linguist, UBC, Vancouver, BC, Canada)

Alcohol intoxication is characterized by hypermetric movements (overshoot) in manual fine motor skills [Phillips *et al*., (2009). Hum. Movement Sci,. 28(5), 619–632]. It remains unknown whether hypermetric movements manifest in speech following alcohol intoxication. Recent work on a small population ($n = 15$) revealed vowel space area (VSA) expansion following intoxication [Chang *et al*., 2023. Proc. ICPhS '23]. To validate these findings, we made use of the publicly available Alcohol Language Corpus [Schiel *et al*., 2012. Lang. res. and eval. 46, 503–521]. VSAs were compared across 162 speakers (85 male, 77 female) while sober ($n = 162$) and intoxicated (blood alcohol level $> 0.08\%$, $n = 97$) using an unsupervised method based on cluster center detection of normalized vowel formants [Sandoval *et al*. JASA. 134(5), EL477-EL483]. Substantial VSA expansion was observed for males (10.8%), while a smaller expansion was observed for females (3.6%). These results support recent observations of VSA expansion and suggest hypermetric tongue movement following alcohol intoxication. Further evaluations are ongoing and employ hierarchical linear modeling to compare the effects of specific blood-alcohol concentrations and speech tasks (free speech, tongue-twisters, repetition) on VSA expansion. [Research funded by Tenvos Incorporated for the development of commercial speaker state-detection algorithms.]

2pSCb14. Patterning of laxing spreading in Quebec French. Philippe Aigner-Therrien (Linguist, Carleton Univ., Ottawa, ON K2C3L5, Canada, philippeaignertherri@cmail.carleton.ca)

Quebec French laxing is a rather well-known phonological process, starting from the tense vowels [i], [y], and [u] being rendered lax in some closed syllables. This process can also trigger laxing of high vowels in preceding syllables as well, though the nature of this process remains a source of debate, whether it is a spreading process or vowel harmony. In addition to the nature of the process itself, the patterning that occurs for the laxing spreading is one that has been long discussed, particularly when it comes to words with more than two syllables. Poliquin (2006) lists three different possible patterns that the laxing may take. The goal of this thesis was to not only confirm the presence of the spreading patterns found in Poliquin(2006), but also to determine to what degree they occur within native speakers of Quebec French. While the results seem to confirm the presence of laxing spreading, the frequency at which they occur seems to be rather low and vary not only from speaker to speaker, but also within the same speaker.

2pSCb15. A descriptive study of vowels in Dialects of Pakistani English. Mumtaz Yaqub (The Dept of English Lang. and Appl. Linguist, AIOU, Islamabad-Pakistan, Dept. of Linguist, University of Arizona, Tucson, AZ 85719, mumtazyaqub@arizona.edu) and Muhammad K. Khan (The Dept of English Lang. and Appl. Linguist, AIOU, Islamabad-Pakistan, College Park, MD)

The widespread use of English as a lingua franca has created many varieties of Global English. Pakistani English is an under-documented variety. As Pakistan is a multilingual country, it has been a difficult task for the linguists to document the dialects of Pakistani English and to determine how these dialects are influenced by regional languages (e.g. Pashto, Punjabi, Urdu etc.). Previous studies focused on Pakistani English of only a few locations (influenced by only a few regional languages). A broad-based study is required to document the phonetics of Pakistani English. The current study investigates two questions about the Pakistani English vowel system: how much variety there is, and whether the dialects of Pakistani English are influenced by regional languages. Speech data was collected from 208 undergrad students from thirteen cities representing thirteen regional languages of Pakistan (e.g. Lahore, Karachi, Peshawar etc.). Participants read thirty-two words expected to contain monophthongs in initial, medial and final position. Formant measurements are measured automatically in Praat. Preliminary analyses show variation in formant values of vowels in regional dialects; The results will allow for documentation of the vowel space of Pakistani English and will provide a broad representation of Pakistani English dialects.

2pSCb16. American English diphthong transitions and the acoustic vowel space: A pilot comparison of two tasks. Christina Kuo (James Madison Univ., 235 Martin Luther King Jr. Way, MSC 4304 James Madison University, Harrisonburg, VA 22807, kuocx@jmu.edu)

The purpose of the present study is to evaluate the potential relationship(s) between diphthong transitions and the acoustic vowel space parameters in two speaking tasks. Diphthongs are associated with acoustic changes that reflect changes in vocal tract configurations, affording unique opportunities for investigating the articulatory-acoustic link. The acoustic vowel space, a representation of monophthong vowels in a two-dimensional plane defined by the first and second formant frequencies (F1 and F2), has been widely used to interpret articulatory-acoustic consequences and to index articulatory integrity. Nevertheless, limited is known as to how diphthongs and monophthong vowels account for the acoustic working space together [Lee *et al*., JASA 136(4), 1880–1894 (2014)]. To add to this inquiry, transitions of American English diphthongs /ei/, /ɑɪ/, /ɑʊ/, /oʊ/, and /ɔi/, defined by the primary F2 transition, are examined with respect to the quadrilateral vowel space anchored by /i/, /æ/, /ɑ/, and /u/. The speech sounds were obtained from four male speakers in two tasks, sentence reading and connected speech, that have been shown to elicit vowel space changes. It is hypothesized that greater diphthong transitions are associated with parameters corresponding to a larger vowel space. Findings will be discussed within the framework of the acoustic theory.

2pSCb17. Perceptual adaptation to unfamiliar dialect variation. Kevin D. Lilley (Linguist, The Ohio State Univ., 1712 Neil Ave., Columbus, OH 43210, lilley.35@osu.edu) and Cynthia G. Clopper (Ohio State Univ., Columbus, OH)

Regional varieties of a language often differ in the phonetic realization of shared phonological categories, resulting in perceptual ambiguity. In perceptual tasks, listeners with more exposure to regional dialect variation perform differently than listeners with less exposure, reflecting underlying differences in lexical competition. In this study, we examined perceptual adaptation to one authentic dialect of American English and one novel dialect to test for adaptation to unfamiliar variation. Our materials contained contrasts that are perceptually ambiguous in Southern American English and in the novel dialect we created, in which linguistically trained bilinguals substituted English /ow/ with Spanish /o/. Participants heard a familiarization passage spoken in a Southern or Novel dialect and then completed an auditory lexical decision task. Adaptation was assessed by differences in accuracy and response times for each vowel contrast, talker dialect, and familiarization dialect. Results revealed lower accuracy and slower response times for both target contrasts when pronounced in their respective dialects. These effects reflect the difficulty in processing perceptually ambiguous stimuli. Familiarization dialect had no effect on accuracy or response times; future analyses will include quantification of listener regional dialect exposure to better assess its influence on processing novel variation.

2pSCb18. Monophthongization of Diphthongs in Southern American English: A perception study. Ratree Wayland (Linguist, Univ. of Florida, 4131 Turlington Hall, P.O. Box 115454, Gainesville, FL 32611-5454, ratree@ufl.edu), Harys Dalvi (Comput. Sci., Brown Univ., Providence, RI), and Rachel Meyer (Linguist, Univ. of Florida, Gainesville, FL)

The more prevalent monophthongization of /ai/ compared to /au/ in certain dialects, particularly in Southern American English, can be attributed to several factors. Historically, /ai/ has been more susceptible to dialectal variation and change. Phonetically, the tongue movement required for /ai/ (from a low to a high front position) may be more easily simplified or reduced than the movement for /au/ (from a low to a high back position). Sociolinguistic factors might also play a role, with certain vowel changes becoming markers of regional or social identity, leading to their widespread adoption in speech. This study examines the monophthongization of the diphthongs /ai/ to /a/ and /au/ to /a/ in Southern American English. Two 11-step continua, [a-ai] and [a-au], were presented to listeners for identification. The hypothesis is that a higher incidence of /a/ will be perceived in the /a-ai/ continuum than in the /a-au/ continuum among Southern dialect speakers compared to non-Southern dialect speakers. The expected results suggest that a significant tongue position shift in the front region of the oral cavity (for the /i/ in /ai/) leads to a smaller perceptual shift than a similar shift in the back region (for the /u/ in /au/).

2pSCb19. Neural network-based measure of voice quality in Parkinson's disease. Ratree Wayland (Linguist, Univ. of Florida, 4131 Turlington Hall, P.O. Box 115454, Gainesville, FL 32611-5454, ratree@ufl.edu), Kevin Tang (English Lang. and Linguist, Heinrich-Heine-Universität Düsseldorf, Düsseldorf, Germany), Sophia Vellozzi (Comput. Sci., Univ. of Florida, Gainesville, FL), Rachel Meyer (Linguist, Univ. of Florida, Gainesville, FL), and Rahul Sengupta (Comput. Sci., Univ. of Florida, Gainesville, FL)

Parkinson's Disease (PD) significantly affects speech and voice. The Parkinsonian voice is often described as breathy, rough, hoarse, tremulous, abnormally pitched, having reduced pitch range, and unusually quiet. Changes in voice quality result from altered neurological controls of the muscles in the respiratory and phonatory systems, impacting breath support and vocal fold vibration. This study investigates the effects of PD on voice quality (e.g., breathiness) in vowel production among native Spanish speakers. The degree of breathiness is estimated from posterior probabilities calculated by recurrent neural networks trained to recognize spread glottis phonological features in Gujarati, a language contrasting breathy and modal voicing in vowels and between voiced aspirated (breathy voice) and voiceless aspirated versus plain voiced stops in consonants. It is hypothesized that vowels produced by PD patients will exhibit a higher degree of breathiness than those produced by normal controls, with degrees of breathiness potentially varying as a function of disease progression.

2pSCb20. Effects of exposure to dialect-specific allophonic variation on perceptual similarity ratings of vowels. Marie Bissell (Ohio State Univ., 4229 Avent Ferry Rd., Apt 3, Raleigh, NC 27606, marie.bissell@gmail.com) and Cynthia G. Clopper (Ohio State Univ., Columbus, OH)

Dialect exposure affects listeners' lexical processing. In this study, we examined whether listeners' exposure to dialect-specific allophonic variation in two linguistic variables in American English (nasal split /æ/ system and pre-voiceless /ɑɪ/ raising) affects their perceptual similarity ratings of matching versus mismatching allophones. Listeners from the U.S. North were assumed to have more exposure to pre-voiceless /ɑɪ/ raising and less exposure to a nasal split /æ/ system, while listeners from the U.S. Midland were assumed to have more exposure to a nasal split /æ/ system and less exposure to pre-voiceless /ɑɪ/ raising. We analyzed Northern and Midland listeners' perceptual similarity ratings on a 1–5 scale for three types of auditory vowel pairs: match (e.g., *ban-bang* or *bike-bite*), allophonic mismatch (e.g., *ban-bat* or *bike-bide*), and phonemic mismatch (e.g., *ban-bike*). We predict that listeners with more exposure to dialect-specific allophonic variation should rate allophonic mismatches as more different than allophonic matches compared to listeners with less exposure. Preliminary results suggest that listeners rate allophonic matches as more similar than allophonic mismatches, and both of these types are rated as more similar than phonemic mismatches, consistent with existing literature. There was no effect of dialect exposure to allophonic variation on perceptual similarity ratings.

2pSCb21. The effect of aging on vowel production in Western Canadian English. Brooklyn Jones (Northern Arizona Univ., 208 E Pine Knoll Dr., Flagstaff, AZ 86011, bj587@nau.edu) and Benjamin V. Tucker (Commun. Sci. and Disord., Northern Arizona Univ., Flagstaff, AZ)

Aging influences many aspects of speech production. For example, a speaker's fundamental frequency or formant space has been reported to change as they age. However, this research has largely focused on careful or prepared speech, with a few notable exceptions in the longitudinal research literature. There is limited knowledge of the relationship between aging and vowel reduction in spontaneously produced speech. It is important to investigate how spontaneous everyday speech is influenced by the aging process. In the present investigation, we work with a corpus of Western Canadian English containing spontaneous conversational speech produced by 18 younger (17–31) and 13 older speakers (64–79). The data was force-aligned and then acoustic measures were extracted from the aligned data. In this presentation, we report on the differences in formant space for stressed and unstressed vowels. The results of this study will be used to better inform our understanding of the role of vowel centralization and aging in spontaneous speech. They will further support the understanding of reduction in spontaneous speech.

2pSCb22. Binaural fusion and vowel perception in cochlear implant users. Kayla Pierre (Commun. Sci. and Disord., Penn State Univ., 308H Ford Bldg., University Park, PA 16802, kaylapierre0124@gmail.com) and Lina A. Reiss (Otolaryngol. & Biomedical Eng., Oregon Health & Sci. Univ., Portland, OR)

Excess binaural fusion can be a problem, especially in noisy environments when speech from multiple talkers are fused together. Previously, we showed that hearing aid users experience abnormally broad binaural fusion that leads to fusion and averaging of dichotic vowels of fundamental frequency (F0) differing by up to 1 octave (Reiss and Molis 2021). This study investigated whether cochlear implant (CI) listeners also experience abnormal fusion and averaging of double vowels in the free field. Six adult CI users (4 females) and four NH adults (1 female) were tested. Stimuli were synthetic vowels/æ/, /a/, /u/ and /i/, at $F0 = 106.9$, 151.2, and 201.8 Hz. Two different vowels were presented simultaneously from two loudspeakers at $\pm 60^\circ$, with same or different F0. Subjects were instructed to select which one or two vowels they perceived. When $\Delta F0 = 0$, NH and CI users often fused and identified only one vowel. As $\Delta F0$ increased, NH listeners increased identification of two vowels, while CI users continued to fuse vowels. The findings show that in realistic free field listening conditions, CI users can experience excess fusion of multiple vowels of differing F0s, presenting additional challenges in noisy listening environments. [Work supported by NIH R01DC013307 and ASA SUREIA program.]

2pSCb23. Quantifying vowel category distinctness using Bayesian modelling. Irene Smith (Linguist, McGill Univ., 1085 ave. du Docteur-Penfield, Montreal, QC H3A1A7, Canada, irene.smith@mail.mcgill.ca) and Morgan Sonderegger (Linguist, McGill Univ., Montreal, QC, Canada)

Phonetic and sociolinguistic studies of vowel merger require a measure of acoustic distinctness between vowel categories. Three desiderata for such a metric are that it be multivariate, in the sense that it account for correlations between dimensions (e.g., F1 and F2), control for other factors affecting vowel formants (e.g. surrounding consonants), and work for unbalanced data (common in naturalistic data). Previous work (Nycz and Hall-Lew, 2013; Kelley & Tucker, 2020) has considered a variety of measures, including variants of Euclidean distance, Pillai score, and Bhattacharyya affinity, but none meet all three criteria. We present a new method for quantifying vowel merger that meets all desiderata and can be applied to different metrics: we fit a Bayesian mixed-effects linear model to jointly predict F1 and F2, then compute any desired metric—here, Euclidean distance, Pillai score, and Bhattacharyya affinity—from the posterior. We evaluate each metric, and the overall method, to describe PIN-PEN across a range of English dialect corpora. We find that controlling for covariates and unbalanced data adds substantial signal, but that multivariate modelling does not perform substantially better than univariate modelling. Additionally, we argue that Bhattacharyya affinity has particularly desirable properties (e.g., allowing for heteroskedastic data: Johnson, 2015).

2pSCb24. Vowel quantity/quality redux: Revisiting vowel contrasts in Hakha Lai monophthongs. Kelly H. Berkson (Linguist, Indiana Univ., Bloomington, 1020 E. Kirkwood Ave., Ballantine Hall 540, Bloomington, IN 47405-2201, kberkson@indiana.edu), Grayson Ziegler, Amanda Bohnert (Linguist., Indiana Univ., Bloomington, IN), and Kenneth Van Bik (Dept. of English, Comparative Lit., and Linguist, California State Univ., Fullerton, Fullerton, CA)

Previous scholars have documented a phonemic length distinction in the monophthongal vowels of Hakha Lai, a Tibeto-Burman language spoken in Chin State in western Myanmar (Melnik, 1997; Peterson, 2003; Maddieson, 2004). In a previous pilot study of data from two college-aged speakers (the first instrumental acoustic analysis of Hakha Chin vowel quality), our findings revealed stark individual differences: one speaker showed robust quantity and minor quality differences, while for the other, neither duration nor quality played key roles; rather, her system had merged. Maddieson (2004) proposed that the length associated with long vowels in Hakha Lai is mainly realized through lengthening of sonorant codas; we did not find this to be the case for either speaker. However, those pilot data were limited both in terms of sample size and age range. Therefore, to expand on our previous findings, we now present analysis of data from fourteen talkers, including

both a group of nine college aged talkers (6 women) and five older talkers (2 women, ages 46 and 77; 3 men, ages 48, 60, and 62). Duration and quality measures are reported for all monophthongs in all possible syllable shapes.

2pSCb25. Vocalic contrasts in Zotung. Amanda Bohnert (Linguist, Indiana Univ., Bloomington, IN), Kelly H. Berkson (Linguist, Indiana Univ., Bloomington, 1020 E. Kirkwood Ave., Ballantine Hall 540, Bloomington, IN 47405-2201, kberkson@indiana.edu), Kenneth Van Bik (Dept. of English, Comparative Lit., and Linguist., Cal State Fullerton, Fullerton, CA), and Grayson Ziegler (Linguist., Indiana Univ., Bloomington, IN)

Though very few Maraic languages from the South Central branch of Tibeto-Burman are represented in the existing body of phonetics literature, they are of interest in that they appear to contain typologically unique dense high vowel systems. Zotung, spoken by approximately 100 000 people in Chin State in western Burma and in diaspora communities, is one such language (Eberhard *et al*., 2022). The only linguistic scholarship on Zotung (Shintani, 2015) is an extensive wordlist with brief preliminary comments on the syllable shape, tone, and vowel inventories; however, acoustic analyses are absent. To augment Shintani (2015) and further elucidate the vowel inventory of Zotung, we present data from an older male speaker deeply involved with Zotung translation, preservation, and literacy efforts, who hails originally from Tingsi on the far southern border of the Zotung speaking area. Based on analysis of 553 tokens collected via a wordlist, we substantiate some of the findings reported by Shintani (2015) and document several important differences. Shintani reported 8 monophthongal vowel qualities; we find 10: [i, y, e, ø, æ, ə, a, u, o, ɔ]. Shintani also reported an expansive diphthong inventory; our data support this claim, with 9 total vowels that can be categorized as diphthongs.

2pSCb26. An ultrasound investigation of Hnaring Lutuv (high?) vowels. Grayson Ziegler, Amanda Bohnert (Linguist, Indiana Univ., Bloomington, IN), Kelly H. Berkson (Linguist, Indiana Univ., Bloomington, 1020 E. Kirkwood Ave., Ballantine Hall 540, Bloomington, IN 47405-2201, kberkson@indiana.edu), and Sui Hnem Par (Linguist, Indiana Univ., Bloomington, IN)

Lutuv (also known as Lautu) is an under-documented Chin language from the Tibeto-Burman language family spoken by 18,000 people, both in Chin State in western Burma and in diaspora communities worldwide, including approximately 1000 people in the Indianapolis Chin refugee community. Lutuv utilizes a typologically rare six-way contrast in the higher part of the vowel space (i y ɨ ʉ ɯ u, see Bohnert *et al*. 2022), with an additional four high diphthongized vowels (ie̯ yə̯ ɯə̯ uo̯). Previous work has also identified that the high central vowels (/ɨ ʉ/) are poorly disambiguated—acoustically, they show considerable overlap with both each other and the high back vowels, and in terms of lip posture, they do not display the characteristics of a typical rounding contrast (Bohnert & Berkson 2023). The present work utilizes 3D ultrasonography to provide detailed lingual articulatory data of Lutuv vowels with special attention paid to the high central vowels, adding a new dimension to the existing acoustic and articulatory data. Real-time images of tongue position and motion provide new insights into the complex articulatory gestures involved in the production of these sounds and constitute the first ultrasound investigation into this underdocumented language.

2pSCb27. Acoustic features of running speech by Chinese college student: Effects of local dialect and second language. Yunzi Wan (Dept. of English, Chengdu Inst., Sichuan Int.. Studies Univ., School of English Studies, Chengdu Inst. Sichuan Int.. Studies University, Chegdu, Sichuan 611844, China, 463445683@qq.com), Wei Hu (Dept. of Psych., Tianjin Normal Univ., Tianjin, China), and Chang Liu (Univ. of Texas at Austin, Austin, TX)

The goal of this study was to investigate the acoustic features of running speech recorded by college students in China in three languages: standard Mandarin, Mandarin dialect (Sichuan dialect), and English (e.g., the second language). Acoustic analyses were focused on spectral features (e.g., vowel space and spectral tilt), temporal features (e.g., temporal envelope), and speaking rate. Preliminary data analysis indicated that for Chinese college students, the speaking rates of their native languages: standard Mandarin and Sichuan dialect were comparable and significantly faster than English. In addition, the vowel space (e.g., /a-u-i/) was the largest for standard Mandarin and the smallest for Sichuan dialect with English in between. Temporal properties of speech like the dominant temporal modulation frequency and temporal modulation depths will be compared and discussed across the three languages.

2pSCb28. Exploring infant talker bias: Insights from remote speech perception testing. M. Fernanda Alonso Arteche (School of Commun. Sci. and Disord., McGill Univ., 1295 Rue des Carrieres, apt 402, Montreal, QC H2S 0E1, Canada, maria.alonsoarteche@mail.mcgill.ca), Nicola Phillips, Samin Moradi, Lulan Shen, Marianne Chen-Ouellet, Leatisha Ramloll, Sumana Abraham, Lei Zeng (School of Commun. Sci. and Disord., McGill Univ., Montreal, QC, Canada), Lucie MENARD (Linguist, UQAM, Montréal, QC, Canada), and Linda Polka (School of Commun. Sci. and Disord., McGill Univ., Montreal, QC, Canada)

Lab studies show that infants (4- to 7-month-olds) prefer to listen to vowels with infant-like f0 and formant frequencies over those of an adult female (Masapollo *et al*., 2015; Polka *et al*., 2021). This *Infant Talker Bias* may facilitate infants' mapping of articulatory gestures to acoustic correlates. In this study, 4- to 12-month-olds completed a listening preference task on the *Lookit* online testing platform. Across eight trials, we presented synthesized infant and adult vowel sounds (/i/ and /a/) paired with a simple animation and recorded the infant's response via the webcam. Infant looking time and vocalization to each vowel type were coded offline. Preliminary analyses ($n = 91$) show that listening time increased with age ($p < 0.05$), and all infants listened longer to infant vowels than to adult vowels ($p < 0.01$). Preliminary analyses ($n = 62$) also show an increase in infant vocalizations with age ($p = 0.00$) and a trend towards more and longer vocalizations in response to the adult vowels. These findings replicate and extend the infant talker bias to new vowel stimuli and to older infants, and support the use of remote testing in infant speech perception studies.

Session 2pSP

Signal Processing in Acoustics, Biomedical Acoustics, Physical Acoustics, and Underwater Acoustics: Denoising and Enhancing Acoustic Signals

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Chair's Introduction—3:50

Contributed Papers

3:55

2pSP1. Balancing ensemble averaging and signal stationarity when processing acoustic data recorded during a rocket launch. Carson F. Cunningham (Phys., Brigham Young Univ., Brigham Young University, Provo, UT 84602, carsonfcunningham@gmail.com), Micah Shepherd (Brigham Young Univ., Provo, UT), and Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

There are many important details to consider when collecting acoustical data during a rocket launch including system noise floor, dynamic range and transducer sensitivity. In this talk, the choice of signal processing parameters will also be shown to be important. Since the pressures are created by turbulence, ensemble averaging is key to reducing random variation. As such, the use of a Hanning window with 50% overlap would be customary for performing this type of processing. However, as the rocket lifts off, the source location moves, and the data can only be considered stationary for short blocks of time. Therefore, the ability to average is limited by non-stationarity. To balance these competing phenomena, an adjustable windowing function such as the Tukey window may be more appropriate than Hanning. Noise data collected at two different launches will be used to compare of the Hanning window with 50% overlap to that using of the Tukey window with a taper of $\alpha = 0.25$, which filters 12.5% of the signal at each extremity, with 87.5% overlap. The use of both window types will be compared when estimating auto spectral densities and integrated overall levels. The effect of blocksize will also be determined.

4:10

2pSP2. A lucky covariance estimator using cumulative coherence. Daniel J. Brooker (U. S. Naval Res. Lab., Washingtron, DC) and Geoffrey F. Edelmann (U. S. Naval Res. Lab., 4555 Overlook Ave. SW, Code 7145, Washington, DC 20375, edelmann@nrl.navy.mil)

This talk demonstrates a technique to improve covariance estimation using the principles of lucky signal processing and the cumulative coherence. Lucky processing, popularized in astro-photography, is a technique that increases signal quality by selectively keeping only a small fraction from a pool of potential snapshots. Cumulative coherence, a measure of how well a set of vectors is described by its subsets, provides the measure of "data quality" that enables the lucky processing. This approach was applied to covariance estimation on an acoustic array by taking a fixed duration sample of data and creating a dense set of snapshots with higher than usual overlap. From these densely sampled snapshots, the "luckiest" ones were found using cumulative coherence, and the covariance was averaged as normal. It was found that the lucky covariance estimate was successful at adaptive matched field processing and produced a less ambiguous processor output than the conventional estimator. The lucky covariance estimate had a higher estimated signal-to-noise ratio, especially when the source was at longer ranges from the array. [This work is supported by the Office of Naval Research.]

4:25

2pSP3. Reciprocal sensitivity kernels for receiver positioning. Alexis Bottero (DGA, Av. De la tour Royale, Toulon 83000, France, alexis.bottero@gmail.com)

In their classical formulation, sensitivity kernels aim to evaluate the points where an infinitesimal variation in the properties of the medium would most affect a pressure (or displacement) measurement at a fixed location. These kernels feature typical cigar shapes, with characteristically low sensitivity on the direct path. In this presentation, the same approach is applied to the reciprocal problem, where we want to evaluate, at a constant source position, which points are the most (or least) sensitive to a given model change. These reciprocal kernels exhibit flared shapes centered on the areas where the model is perturbed. Applications to receiver positioning in acoustic are discussed.

4:40

2pSP4. Comparison of conventional and adaptive acoustic beamforming algorithms using a tetrahedral microphone array in noisy environments. Megan B. Ewers (Elec. and Comput. Eng., Portland State Univ., 1900 SW 4th Ave., Ste. 160, Portland, OR 97201, megan.ewers1@gmail.com), Adina Edwards, and Martin Siderius (Elec. and Comput. Eng., Portland State Univ., Portland, OR)

In situ acoustic measurements are often plagued by interfering sound sources that occur within the measurement environment. Both adaptive and conventional beamforming algorithms, when applied to the outputs of a microphone array arranged in a tetrahedral geometry, are able to capture sound sources in desired directions and reject sound from unwanted directions. Adaptive algorithms may be able to measure a desired sound source with greater spatial precision, but require more calculations and, therefore, computational power. A conventional frequency-domain phase-shift algorithm and a modified adaptive frequency-domain Minimum Variance Distortionless Response (MVDR) algorithm were applied to simulated and recorded signals from a tetrahedral array of omnidirectional microphones.

The algorithms are described mathematically and demonstrated on both deterministic and real-world sound data, to quantitatively validate and compare their performance and to provide listening examples of their outputs in a variety of acoustically replicated environments. [Work supported by Portland State University.]

4:55

2pSP5. Efficient methods for iterative ultrasound image reconstruction using L1 and L2 norm regularization. Marko Jakovljevic (Radiology, Massachusetts General Hospital, 101 Merrimac St., Boston, MA 02114, mjakovljevic@mgh.harvard.edu), Ettore Biondi (Div. of Geological and Planetary Sci., California Inst. of Technol., Pasadena, CA), and Anthony E. Samir (Radiology, Massachusetts General Hospital, Boston, MA)

Ultrasound image reconstruction can be posed as an inverse problem, with image pixels as parameters to a model of wave propagation that are estimated from raw ultrasound channel data. Such framework allows one to make assumptions about channel signals and image properties in form of regularization that can be used to improve reconstruction accuracy in the presence of electronic and acoustic noise. A simple model assumes spherical wave propagation from point sources in a homogeneous medium, similar to delay-and-sum (DAS) beamforming; solving such problems numerically can require many iterations and can result in blurred images when traditional L2 norm regularization is used. We use Fast Iterative Shrinkage Thersholding Algorithm (FISTA) to reduce the number of iterations to less than 10, and to apply L1-norm regularization to the data, which improves edge sharpness and image contrast. We demonstrate the concept using FIELD II simulated ultrasound signals from point, speckle, and anechoic targets with controlled levels of electronic and speckle noise. The FISTA-reconstructed point targets show reduced sidelobe levels by 15 dB compared to the traditional DAS image, and the reduction in full-width at half maximum by a factor of 2. We also implement iterative reconstruction in omega-k frequency domain, which allows factoring the forward and adjoint operators at each spatial frequency and paves the way for an intuitive and more memory efficient, multi-threaded implementation of the method.

TUESDAY AFTERNOON, 14 MAY 2024 — ROOM 215, 2:15 P.M. TO 5:15 P.M.

Session 2pUW

Underwater Acoustics: Geoacoustics of Marine Sediments

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Contributed Papers

2:15

2pUW1. Observations of spatial coherence from a multibeam subbottom profiler. Laura Brownstead (Acoust., Penn State, 201 Appl. Sci. Bldg., University Park, PA 16802, lgb5113@psu.edu) and Daniel C. Brown (Acoust., Penn State, State College, PA)

Sonar data from the Kongsberg SBP-29 (with a 64-channel receive array) aboard R/V Sally Ride is analyzed to judge the value of spatial coherence of signals reflected from the ocean floor. The multibeam sub-bottom profiler collected five days of sonar test data off the coast of Southern California over a variety of sediment types and marine geologies. The spatial coherence of measured signals is sensitive to different bottom types and to sediment layering deep below the ocean floor. Navigational data also permits comparison of sonar data to existing bathymetric maps of the ensonified floor. The flexibility of the transmit and receive arrays permit detailed study of potentially informative seafloor features and landmarks. The findings of primary interest are interpreted in the context of a bivariate Normal surface fit to characterize the spatial coherence of scattered signals and future work is reported based on statistical analysis of the fitted data.

2pUW2. Temporal change of seafloor scattering and its dependence on environmental parameters in shallow-water sandy sites. Jenna Hare (Ctr. for Coastal & Ocean Mapping, Univ. of New Hampshire, 33-1173 Wellington St., Halifax, NS B3H3A2, Canada, jenna.hare@unh.edu), Anthony P. Lyons, and Gabriel R. Venegas (Ctr. for Acoust. Res. and Education and Dept. of Civil and Environ. Eng., Univ. of New Hampshire, Durham, NH)

In the ocean, the performance of active sonar systems used for object detection and seafloor characterization can be affected when the acoustic properties of the seafloor change due to near-bottom hydrodynamics and biological activity. Determining the dominant environmental mechanisms and corresponding time scales that regulate seafloor scattering will increase our understanding of the performance of these remote-sensing applications. To this end, a high-frequency active acoustic system (operating at 38 kHz, 70 kHz and 200 kHz), a wave-sensing CTD, and a stereo camera were deployed on the seafloor in a series of experiments lasting from two weeks to five months. Seafloor scattering measurements were obtained in two shallow water locations in New Hampshire, USA: a wave-dominated site and a tidal current dominated site. Daily and weekly trends in mean scattered levels and the mechanisms causing their temporal variability are discussed. The temporal change in scattering as a function of angle is compared to the small-slope approximation model where seafloor roughness estimates were obtained using stereophotogrammetry.

2:45

2pUW3. Deep sediment characterization from very low-frequency features from merchant ships. Alexandra M. Hopps-McDaniel (Phys., Knobles Sci. and Anal., PO Box 2185, Laramie, WY 82073, mcdaniel.alexh@gmail.com), David P. Knobles (Phys., Knobles Sci. and Anal., Austin, TX), Tracianne B. Neilsen (Phys. and Astronomy, Brigham Young Univ., Provo, UT), Preston S. Wilson (Walker Dept. of Mech. Eng. and Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX), William Hodgkiss (Scripps Inst. of Oceanogr., San Diego, CA), and Jason D. Sagers (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

The very low-frequency noise from merchant ships provides a good wideband source to study the deep layers of the seabed. The nested striations which characterize ship spectrograms contain unique acoustic features where the waveguide invariant (β) becomes infinite. This occurs at frequencies between 20 and 80 Hz where pairs of modal group velocities are equal. The goal of this project was to identify the $\beta = \infty$ frequencies in ship noise spectrograms and use these frequencies to perform statistical inference for the deep layer sound speeds and thicknesses. The Seabed Characterization Experiment of 2022 on the New England continental shelf had three vertical line arrays strategically placed between two shipping lanes. The average water depth was 75 meters with less than one meter bathymetry change between the arrays. The results of this study are based primarily on five ships. There was a gradual shift in the $\beta = \infty$ frequencies between the three arrays, suggesting a gradual change in the deep sediment layers. [Work supported by Office of Naval Research.]

3:00

2pUW4. The effect of salinity on the rigidity and settling behavior of reconstituted water-saturated kaolinite sediments. Alicia Casacchia (Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, 204 E. Dean Keeton St., Austin, TX 78712-1591, acasacchia@utexas.edu), Megan Ballard, Kevin M. Lee (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Preston S. Wilson (Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Nicholas P. Chotiros (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

In sandy sediments, the salinity of the pore water only affects the sound speed of the mixture by way of affecting the sound speed of the pore water. With clay mineral and saltwater mixtures, the intergranular forces that depend on the salinity of the pore water affect the sediment acoustic properties beyond simply affecting the sound speed of the pore water. These dependencies were investigated via image analyses of prepared samples of two kaolinite clay types (Flat DS and RSA) via analyses of their settling dynamics, porosity, and rigidity as a function of pore water salinity. There exists an observable dependence on salinity to mixture porosity and rate of settling. In addition, three regimes of slip-stick dynamics are shown to exist at different salinity ranges: (a) an immediate transition to liquid-like behavior; (b) a viscoelastic transition to liquid-like behavior with long term creep; and (c) a delayed transition to viscoelastic behavior with long term creep. [Work supported by ONR.]

3:15–3:30 Break

3:30

2pUW5. The crucial role of granular packing structure in marine sediment acoustics. Abe Clark (Phys., Naval Postgrad. School, Monterey, CA), Derek R. Olson (Oceanogr., Naval Postgrad. School, 833 Dyer Rd., 313b Spanagel Hall, Monterey, CA 93943, olson.derek.r@gmail.com), and Andrew Swartz (Phys., Naval Postgrad. School, Monterey, CA)

Motivated by the acoustic properties of marine sediments, we study via discrete-element method simulations the frequency dependence of dispersion and attenuation in model marine sediments. By numerically solving for the motion of each grain in a packing, rather than solving a continuum wave equation, we find that the granular packing structure, which is not explicitly considered in existing sediment-acoustics models, plays a crucial role in determining how attenuation and wave speed vary with frequency. Prior work has typically postulated a wave equation that is motivated by grain-scale forces, including viscous effects from the interstitial fluid, without explicitly considering the disordered packing structure of the grains. However, the packing structure is known to produce complex, nonlinear behavior (e.g., during shear or quasistatic compression), and how it affects dispersion and attenuation is not known. We consider linearized forces between particles (i.e., springs and dashpots), and demonstrate that the disordered packing structure leads to emergent scaling laws that do not agree with PDE-based approaches. Our results demonstrate that the granular packing structure must be explicitly considered when constructing theories for the acoustic properties of marine sediments.

3:45

2pUW6. Model manifold of transmission loss from a Pekeris waveguide using VGS parameters. Michelle Wang (Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84604, msw1998@byu.edu), Tracianne B. Neilsen, and Mark K. Transtrum (Phys. and Astronomy, Brigham Young Univ., Provo, UT)

In geoacoustic inversion, selecting an appropriate seabed parametrization, especially with an unknown number of sediment layers, is a challenge that is compounded by potential bias when establishing bounds in the parameter search space. One approach to addressing these issues is rooted in the techniques of Information Geometry. Information Geometry informs model selection and parameterization by quantifying which model parameters are informed by observational data. This paper provides an information geometric analysis of the Pekeris waveguide, where the acoustic properties of the lower half-space are derived from the viscous grain-shearing (VGS) model. Specifically, we consider single frequency transmission loss (TL) across a wide range of VGS parameters. By exploring the limits and boundaries of the geometric manifolds, particularly as parameters approach both low and high extremes, this approach provides indications of parameter hierarchies and correlations. Results will include slices of the model manifold and an evaluation of the boundary structure, providing insight into the relative impact of VGS parameters and the delineation of limiting regions. In doing so, this paper seeks to inform model selection and parameterization in geoacoustic inversion studies. [Work supported by the Office of Naval Research. Grant N00014-21-S-B001.]

4:00

2pUW7. Biogeoacoustic variability in muddy ocean bottom sediment. Kevin M. Lee (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758-4423, klee@arlut.utexas.edu), Megan Ballard (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Andrew R. McNeese (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Gabriel R. Venegas (Dept. Civil and Env. Eng. and Ctr. Acoust. Res. Ed., Univ. of New Hampshire, Durham, NH), Jason Chaytor (U.S. Geological Survey, Woods Hole, MA), Kelly M. Dorgan (Dauphin Island Sea Lab, Dauphin Island, AL), and Preston S. Wilson (Walker Dept. Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Both physical and biological benthic processes can influence seabed heterogeneity and contribute to spatiotemporal variability in geoacoustic properties. In particular, how biological processes affect both sediment acoustic properties and their variability is poorly understood. To address this deficiency, recent measurements investigated spatial variability in the upper few decimeters of sediment near the water-seabed interface within a fine-grained sediment deposit on the New England shelf. At each measurement location, acoustic probes were inserted into the sediment to collect direct *in situ* measurements of sediment sound speed and attenuation at near-ambient conditions, after which cores were collected from the inter-probe propagation paths for *ex situ* analysis of sediment physical, biological, and acoustic properties. Relationships among sediment properties, such as bulk density, porosity, grain size distribution, organic matter composition, infaunal community composition, and acoustic measurements spanning several frequency decades (10–1000 kHz) will be explored in this paper. Frequency dependence of sediment acoustic properties will also be discussed in the context of sediment acoustics models for mud based on the viscous grain shearing and extended Biot theories. [Sponsored by ONR.]

4:15

2pUW8. A T-matrix approach to wave propagation and scattering in layered environments with rough interfaces. Anatoliy N. Ivakin (Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, aniv@uw.edu)

Wave propagation and interactions at the interfaces of multi-layered media can be described in terms of transition matrix coefficients, or T-matrixes, taken from solutions for a plane wave transformation, reflection, scattering, and transmission, at an interface between two homogeneous half-spaces. These solutions can be found separately for each interface of the layered system and each combination of adjacent media. Then scattering amplitudes or the T-matrix of the whole multi-layered system can be obtained using an iterative procedure that starts from a simple case of two half-spaces at the basement of the system. A quite similar iterative procedure is frequently used for calculating the reflection coefficient of compressional plane waves for a multi-layered fluid system with flat interfaces using the reflection coefficients of each interface. In this paper, we show that a similar, but a more general T-matrix approach, can be developed to include interface roughness, different types of media and waves, for instance fluid, elastic or poroelastic layers, compressional and shear waves (vertically and horizontally polarized). As an example, scattering from a rough elastic layer is considered. An explicit first-order expression for the scattering strength is obtained and it's applications to remote sensing of sea ice layer are discussed. [Work supported by ONR.]

4:30

2pUW9. Geoacoustic inversion using 3D modal rays on the New England Shelf Break. Brendan J. DeCourcy (Woods Hole Oceanographic Inst., 86 Water St., Falmouth, MA 02543, bdecourcy@whoi.edu) and Julien Bonnel (Woods Hole Oceanographic Inst., Woods Hole, MA)

During the Seabed Characterization Experiment in 2021 (SBCEX21), hydrophones deployed on the seabed of the New England Shelf Break recorded acoustic signals from SUS charge explosions. Acoustic data from one of these TOSSIT hydrophones displays evident modal structures which cannot be fully captured with 2D propagation models. In this presentation, 3D acoustic modal ray tracing is shown to provide a stronger match for recorded modal travel times, and is used to invert for geoacoustic parameters of the seabed. Performance of 2D and 3D modal propagation models are presented, and limitations of the methods are shown in the context of large parameter space inversion efforts. [This research is supported by The Office of Naval Research.]

4:45

2pUW10. The effect of microfabric heterogeneity on shear wave properties in fine-grained sediments. Gabriel R. Venegas (Cntr. for Acoust. Res. and Edu. and Dept. of Civil and Environ. Eng., Univ. of New Hampshire, 33 Academic Way, Rm. W137, Durham, NH 03824, g.venegas@unh.edu), Yu-Hsuan Chao (Dept. of Bioeng., Univ. of Pittsburgh, Pittsburgh, PA), Jane McCue (Dept. of Civil and Environ. Eng., Univ. of New Hampshire, Durham, NH), Kang Kim (Dept. of Bioeng. and Dept. of Med., Univ. of Pittsburgh, Pittsburgh, PA), and John M. Cormack (Dept. of Medicine, Univ. of Pittsburgh, Pittsburgh, PA)

The water-sediment interface is highly susceptible to physical, biological, and chemical processes, which can create significant levels of heterogeneity in sediment biogeoacoustic properties at various spatial scales that affect transmission and scattering of elastic waves. Using a suite of microscopy, spectroscopy, and medical sonoelastography techniques, relationships between heterogeneity in the sediment microfabric and shear wave speed can be investigated at spatial scales ranging from centimeters to micrometers. Here, we study fine-grained sediments extracted from two unique intertidal mudflats in the Great Bay Estuary, New Hampshire, USA. Shear waves were excited via acoustic radiation force and external vibration in the 50–200 Hz frequency band, and particle velocity was sensed in the top 2 centimeters of sediment at approximately 0.5-micrometer spatial resolution using a research-grade medical imaging array. By directly imaging the propagating shear wave, spatial heterogeneity of shear wave speed was directly measured. Next, subsamples were extracted and prepared for microscopy, where volume fraction and orientation of organic, mineral, and aqueous components were studied at various spatial scales. A comparison of between shear wave speed and microfabric parameters will be presented. [Work sponsored by ONR.]

5:00

2pUW11. Seabed Analysis on the New England shelf break using ambient sound data and trans-dimensional geoacoustic inversion. Martin Siderius (Elec. and Comput. Eng., Portland State Univ., 1900 SW 4th Ave., Ste. 160, Portland, OR 97201, siderius@pdx.edu), Stan Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada), and Jorge Quijano (JASCO Appl. Sci., Victoria, BC, Canada)

The New England Shelf Break Acoustics (NESBA) Signals and Noise experiment was conducted in April-May 2021. Ambient sound data were collected over several days in the mud-patch area as well as in deeper water closer to the shelf break. A 16-hydrophone vertical array was used to measure the natural sound of breaking waves on the sea-surface. Using beamforming in the 500–700 Hz band, these data were used to obtain an estimate of the bottom reflection coefficient as well as the seabed layering. The reflection coefficient data were subsequently used with trans-dimensional inversion techniques to produce a geoacoustic model for the seabed (e.g., sound speed, density, and layering). Results show the ambient sound data can be used to produce well resolved geo-acoustic parameters, especially in the upper part of the seabed (e.g., <10 m). These results are compared between several locations on the New England Shelf Break area and are also compared with other published results using different estimation techniques. In addition, some of the issues related to the impacts of data errors and preferred measurement geometries will also be presented. [Work supported by the Office of Naval Research.]