Session 5aAA

Architectural Acoustics, Structural Acoustics, and Vibration and Noise: Absorptive and Diffusive Metasurfaces for Architectural Acoustic Application

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Chair's Introduction—8:00

Invited Papers

8:05

5aAA1. Optimal bi-state acoustic metamaterial for broadband sound absorption and diffusion: A real-estate dilemma. Eric Ballestero (Institut d'Acoustique - Graduate School, Laboratoire d'Acoustique de l'Université du Mans, UMR CNRS 6613, University of Le Mans, Le Mans 72085, France, eric.ballestero@univ-lemans.fr), Yang Meng (Institut d'Acoustique - Graduate School, Laboratoire d'Acoustique de l'Université du Mans, UMR CNRS 6613, Le Mans, France), Ping Sheng (Hong Kong Univ. of Sci. and Technol., Hong Kong, Hong Kong), Vincent Tournat (Institut d'Acoustique - Graduate School, Laboratoire d'Acoustique de l'Université du Mans, UMR CNRS 6613, Le Mans, France), Vicent Romero-Garcia (Instituto Universitario de Matemática Pura y Aplicada (IUMPA), Universitat Politècnica de València (UPV), València, Spain), and Jean-Philippe Groby (Institut d'Acoustique - Graduate School, Laboratoire d'Acoustique de l'Université du Mans, UMR CNRS 6613, Le Mans, France)

The advent of metamaterials has given a new breath to acoustic treatment design due to their ability to target much lower frequencies within deep-subwavelength dimensions. However, many practical applications may require a multi-functional structure instead of a static one with a single purpose and, thus, overturn geometry-specific performance. Such hybrid acoustic treatments have been an active field of research for the past decades, in which a trade-off between different acoustical mechanisms has to be struck, the major issue being the inversely decreasing returns for each acoustical mechanism, i.e., the more of one the less of the other. The aim of the present work is to build over previous metamaterial strategies and design a dynamic bi-state passive acoustic metamaterial that can change its acoustic properties within the same structural volume, going from highly absorbent ($\alpha \sim 0.85$) to diffusive ($\delta \sim 0.65$) over more than one octave for each acoustic phenomenon, i.e., 400–1100 and 1000–2500 Hz for absorption and diffusion, respectively, with an overall thickness L = 12 cm two times thinner than traditional acoustic treatments. Such design can help enhance the acoustics of rooms, but it can also be introduced to critical environments with limited space, such as aerospace.

8:25

5aAA2. Towards extraordinary sound absorption using coupled resonances. Yun Jing (Acoust., Penn State Univ., 201 Appl. Sci. Bldg., State College, PA 16802, jing.yun@psu.edu) and Jun Ji (Acoust., Penn State Univ., State College, PA)

In this paper, we provide an overview of recent advancements in novel sound absorption designs, predominantly based on optimized coupling between resonances. First, we present a sound absorption panel design capable of achieving high absorption in the low-frequency broadband range (50–63 Hz, one-third octave band). This panel demonstrates an average absorption coefficient of approximately 93%, all while maintaining a thickness of 15.4 cm (equivalent to 1/45 of the wavelength at 50 Hz). Transitioning from the one-port system, we delve into the realm of the two-port system, introducing an ultra-sparse structure for flow-free sound absorption. This structure exhibits near-perfect absorption (∼99%), and this absorption capability is achieved when the spatial period of monopole–dipole resonators is close to one working wavelength (95% wavelength). These developments signify significant strides in sound absorption technology, presenting innovative solutions with enhanced efficiency and performance across various frequency ranges and environments.

8:45

5aAA3. Acoustic performance of screens designed to act as metamaterials. Umberto Berardi (Toronto Metropolitan Univ., 350 Victoria St., Toronto, ON M5B 2K3, Canada, uberardi@ryerson.ca), Antonella Bevilacqua (Università della Campania Luigi Vanvitelli, Parma, Italy), Amelia Trematerra, and Gino Iannace (Università della Campania Luigi Vanvitelli, Aversa, Caserta, Italy)

Metamaterials are, nowadays, a mature field of research in acoustic and many applications with metamaterials have been developed in recent years. In metamaterials, the attenuation of sound is due to the interaction of the sound waves with their geometrically regular structures. The frequency at which the attenuation occurs is calculated with Braggs' law. This work presents sound attenuation measurements of structures made with metamaterials carried out in an anechoic chamber. The arrangements considered are both

two-dimensional and three-dimensional: the 2D arrangement was obtained with a regular distribution of linear elements made with bars with diameters of 9, 15, and 20 mm; the 3D arrangement was obtained with a regular distribution of a lattice of spheres with a diameter of 23 mm. The results of the acoustic measurements are reported in terms of insertion loss in the range from 1000 to 10 000 Hz.

Contributed Papers

9:05

5aAA4. Transparent sound absorption—More than 25 years of applications. Christian Nocke (Akustikbuero Oldenburg, Sophienstr. 7, Oldenburg, Nds. 26121, Germany, nocke@akustikbuero-oldenburg.de)

Optically transparent micro-perforated sound absorbers were introduced more than 25 years ago. One of the first projects carried out was the German Parliament (Bundestag). another project carried out recently is the Canadian Parliament. At first, in the German Parliament, acrylic glas boxes with a sub-millimeter micro-perforations have been developed an applied. Later, on other micro-perforated sheets, such as thin polycarbonate or PVC sheets, have been applied in various projects around the world such as in Ottawa. This contribution concentrates on optically transparent sound absorbers. Fully transparent absorbers can be installed in front of glass facades, under glas roofs and whenever no optical distrubances are wanted. From pandemic times, also transparent screens and other applications are well known. A short review of the applications of various different materials with transparent microperforated sound absorbers is presented. Finally, sound absorption data for different setups are presented.

9:20

5aAA5. Some considerations on the simulation of meta-materials acoustic behavior. Francesco Martellotta (Dept. Architecture, Construction and Design, Politecnico di Bari, Via Orabona 4, Bari 70125, Italy, francesco.martellotta@poliba.it), Ubaldo Ayr, Chiara Rubino, and Stefania Liuzzi (Dept. Architecture, Construction and Design, Politecnico di Bari, Bari, Italy)

One of the major advantages behind the success of metamaterials is related to the possibility to design them based on theoretical assumptions or when theory gets complicated, thanks to numerical methods. Many efforts have been made in the last years to better support researchers with benchmark tools that allow us to make their calculations more reliable, offering the opportunity to directly model the sound propagating inside the material or allow the determination of the parameters that can be used to feed phenomenological models. However, like any other simulation tool, a proper knowledge of the physics and the ways it is approximated are required to get the best results. The paper presents a selection of issues that may play a relevant role to improve the final results, including geometrical discretization, modelling of boundary conditions, and use of periodical structures. A discussion follows also pointing out the differences pertaining to transmission and absorption problems.

Break 9:35–9:50

9:50

5aAA6. Absorptive metasurface inaccuracies made of micropore and microslit panels in multiple structures. Ziqi Chen (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, chenz33@rpi.edu) and Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY)

Microperforated panels in forms of micropores/microslits (MPP/MSP) represent absorptive metasurfaces, which can achieve high absorption with single- or double-layer MSP/MPP absorbers. However, architectural acoustics practice usually requires broadband high absorption. Multiple metastructures in more than two layers become one of possible options. Multiple MPP/MSP metasurfaces arranged in layers inherently complicate the design process. This work applies a model-based Bayesian framework employing a potentially multilayered prediction model. The Bayesian framework involves two-levels of probabilistic inference to design a parsimonious number of layers as the higher level, quantitatively implementing Occam's razor. Once the number of multilayers is selected, the MPP/MSP parameters are readily available within the Bayesian framework, so that the overall metastructure satisfies the design scheme. This work experimentally validates the designed prediction in comparison with the design scheme. However, manufacture inaccuracies may lead to unacceptable deviations. This work applies causation analysis based on a causal model to reveal causal uncertainties/inaccuracies. Using the causal model for causal inference, the Bayesian multilayer design can be satisfactorily validated. This paper discusses comparative investigations of metasurfaces made of micropores and microslits.

10:05

5aAA7. Absorptive performance of layered metasurfaces using microslit panels. Phebe S. Cunningham (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., 110 Eighth St., Troy, NY 12180, cunnip@rpi.edu), Ziqi Chen, and Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY)

This work examines the effectiveness of layered metasurface arrangements, prioritizing space efficiency to create a broadband sound absorber. Innovative metasurfaces, including microslit panels with concentrated and coiled cavities, provide frequency-dependent absorptive properties. While these metasurfaces are independently known as efficient sound absorbers, single-layer microslit panel absorbers with a single cavity, in general, lack a wide effective bandwidth. In this work, theoretical models guide the effective creation of metastructures to achieve broadband absorption. These systems are then assembled and measured using an impedance tube to validate their acoustic performance. This paper discusses the formulation of the theoretical model, experimental validation of the model for layered absorbers, and the design specifications that can be met using various metasurface combinations.

10:20

5aAA8. Robotic fabrication of clay acoustic resonators. Brady Peters (John H. Daniels Faculty of Architecture, Landscape, and Design, Univ. of Toronto, 1 Spadina Crescent, Toronto, ON M5S 2J5, Canada, brady.peters@daniels.utoronto.ca), Nicholas Hoban, John Nguyen, and Nermine Hassanin (John H. Daniels Faculty of Architecture, Landscape, and Design, Univ. of Toronto, Toronto, ON, Canada)

Digital fabrication offers the potential for the creation of complex geometries and mass-customized products; however, most 3D printers do not scale sufficiently to create architectural scale components. Robotic fabrication methods may bridge the gap, offering the possibility of architectural-scale 3d-printing capabilities. It has been found that the combination of multiple Helmholtz resonators tuned to different frequencies can create broadband absorption. This research pairs CAD parametric design with robotic clay extrusion as a method of acoustic resonator mass customization. The history of architectural acoustics together with recent archeological discoveries unveils a long-established practice of using clay vases as acoustic devices. And while the efficacy of these vases in historical settings has been contested, the use of large arrays of carefully tuned acoustic vases remains largely unexplored in contemporary practice. This paper presents the acoustic vases' unique history, defines its geometry and performance, and projects the potentials of the acoustic vase in current practice through modelling, simulation, and fabrication. A full-scale prototype wall with 126 robotically 3d-printed clay resonator vases was designed and constructed. The 1:1 prototype was shown in the "Robotic Clay," which was exhibited at the Canadian Clay and Glass Gallery in Waterloo, Canada.

10:35

5aAA9. Multiphysical design for optimized ventilation and noise attenuation in ducts: An approach using metamaterials. Gioia Fusaro, Rachele Billi, Simone D'Auria (Univ. of Bologna, Bologna, Italy), and Dario D'Orazio (Univ. of Bologna, Viale Risorgimento, 2, Bologna 40126, Italy, dario.dorazio@unibo.it)

Noise generation and propagation through mechanical ventilation and air conditioning (MVAC) ducts are among the main issues related to controlled mechanical ventilation. Commercial solutions allow sensible noise attenuation; however, they also significantly reduce the duct section, forcing the MVAC power to rise to overcome the flow resistance from the commercial silencer, increasing the noise source. The aim of this work stems from such problems, focusing on soundproofing ventilation ducts through acoustic metamaterials without decreasing the inner duct section to limit the flow losses. Moreover, a crucial part of this study falls into the multiphysical interaction between acoustics and fluid dynamics, which is a phenomenon that may result in a challenging analytical model. In conditions of flowing motion, the wave vector has been modeled according to the Lighthill model: the component of the velocity rotor potential becomes significant, influencing the soundproofing performance due to the metamaterial placed on the inner surface of the duct. This work aims to provide a proper setup for wave-based simulation and present some preliminary results.

FRIDAY MORNING, 17 MAY 2024 — ROOM 210, 8:00 A.M. TO 9:45 A.M.

Session 5aBAa

Biomedical Acoustics: General Topics in Biomedical Acoustics: Methods

Keith A. Wear, Chair
Center for Devices and Radiological Health, Food and Drug Administration, Bldg. 62 Rm. 2114, 10903 New Hampshire Ave., Silver Spring, MD 20993

Contributed Papers

8:00

5aBAa1. A spatiotemporal deconvolution approach for measurements of therapeutic ultrasound pressures, intensities, and beamwidths. Keith A. Wear (Ctr. for Devices and Radiological Health, Food and Drug Administration, Bldg. 62 Rm. 2114, 10903 New Hampshire Ave., Silver Spring, MD 20993, keith.wear@fda.hhs.gov) and Sam Howard (Onda Corp., Sunnyvale, CA)

Complete characterization of high intensity focused ultrasound (HIFU) systems includes measurements of transmitted pressure at the focal plane with a hydrophone. However, highly focused, nonlinear HIFU beams can result in considerable spatial averaging artifacts due to finite area of the hydrophone active element. Spatiotemporal deconvolution (STD) can correct for hydrophone spatial averaging (Wear, *IEEE Trans. UFFC* **69**, 1243–1256 (2021)]. The objective of the present work is to extend validation of STD from diagnostic levels (6 MPa) to therapeutic levels (49 MPa). Three HIFU transducers (1.45 MHz F/1, 1.53 MHz F/1.5, and 3.91 MHz F/1) were driven in tone bursts (Agilent HP3314A function generator, ENI 240L amplifier) to generate HIFU fields in a degassed, deionized water tank with focal compressional pressures of 49, 16, and 29 MPa. Fields were measured in focal planes with two hydrophones (Onda HFO 100-μm fiber-optic; Onda HNA-0400 400-μm robust needle). STD reduced inter-hydrophone differences from $-18 \pm 11\%$ to $0 \pm 8\%$ (peak compressional pressure), $-8 \pm 4\%$ to $-5 \pm 5\%$ (peak rarefactional pressure), $-20\% \pm 7\%$ to $-3 \pm 8\%$ (pulse intensity integral), and $21 \pm 12\%$ to $-2 \pm 3\%$ (beam FWHM). STD reduces dependence of measurements on hydrophone active element size. STD reduces reliance on fiber optic hydrophones, which can be expensive, difficult to use, and prone to radiation-force-induced sensor misalignment.

8:15

5aBAa2. Absolute temperature estimation using thermal strain imaging. Omar Gachouch (LabTAU, INSERM, Ctr. Léon Bérard, F-69003, 151 Cr Albert Thomas, Lyon, Rhone Alpes 69003, France, omar.gachouch@inserm.fr), Bruno Giammarinaro, Teymour Kangot, Caterina Monini, and Rémi Souchon (LabTAU, INSERM, Ctr. Léon Bérard, F-69003 Lyon, France)

The objective of this study is to present a new thermometry method, based on thermal strain imaging, to estimate the absolute temperature using ultrasound. Unlike the conventional method presented in previous studies, this method does not require knowledge of the initial temperature or of speed of sound. The method was tested in simulations and experimentally. Simulations were performed at different temperatures (from 20 to 90 °C), using k-wave. Then, experimental measurements were performed in Zerdine phantom samples at different temperature (from 23 to 88 °C). A linear array transducer, with a 12-angle plane wave imaging sequence, was used to acquire the thermal strain images and to derive the temperature at the focus. To control the temperature during the measurements, a thermocouple was used. Both simulations and experiments showed a good estimation of the temperatures. The new ultrasound-based method is a promising technique to estimate and monitor temperature during thermal ablation.

8:30

5aBAa3. Using different methods to measure ultrasonic attenuation in cortical bone: A comparison. Brett A. McCandless (Mech. and Aerosp. Eng., North Carolina State Univ., 1840 Entrepreneur Dr., Raleigh, NC 27614, bamccand@ncsu.edu), Kay Raum (Ctr. for Biomedicine, Charité-Universitätsmedizin, Berlin, Germany), and Marie Muller (MAE, North Carolina State Univ., Raleigh, NC)

The assessment of bone mineral density (BMD) and bone microstructure is important in screening for various bone diseases. Dual x-ray absorptiometry is the most commonly used technique for evaluating BMD; however, alternatives are necessary due to the use of ionizing radiation and low resolution, the latter of which prevents obtaining information on bone microstructure. Quantitative ultrasound presents a potentially attractive alternative. The microstructure of porous media, such as cortical bone, influences the attenuation of ultrasound; as such, ultrasound attenuation can be used to obtain information about the microstructure of cortical bone. Different models and measurement methods may be used to estimate ultrasonic attenuation in a cortical bone. In this study, three different methods of measuring ultrasonic attenuation were used. Finite-difference time-domain simulations were conducted in maps mimicking cortical bone. Pore densities and pore size distributions in the bone matrix, as well as absorptive properties of the bone and porous matrices, were specified for all simulations. Attenuation was measured within the pulse bandwidth as a function of frequency using the independent scattering approximation (ISA), a cortical backscatter method (CortBS), and the reflections from the two surfaces of the cortical bone map. Excellent agreement was found between these three methods.

8:45

5aBAa4. Simulation of high frame rate spread-spectrum color Doppler imaging of pulsatile flow. Kian Esmailian (Biomedical Eng., Western Univ., 100 Perth Dr., London, ON N6A 5K8, Canada, kesmaili@uwo.ca) and James Lacefield (Biomedical Eng., Western Univ., London, ON, Canada)

Spread-spectrum Doppler, a method introduced by our lab, preserves the maximum unaliased velocity of ultrafast Doppler while retaining some of the image quality benefits of compounding plane waves transmitted at different angles. The technique employs a sequence of pulses transmitted at M angles that are repeated L times in different random orders. Shuffling the slow-time samples so the angles repeat in the ascending order concentrates echoes from stationary off-focus targets in M harmonic frequency bins while spreading the in-focus signal across all frequencies. Off-focus echoes are suppressed, without compounding, by applying a notching comb filter, while the portion of the in-focus signal spread to the other $(L-1)M$ bins is retained for velocity estimation. Field II simulations were used to assess the method's ability to track pulsatile velocity fluctuations in a straight vessel. The angle-corrected peak velocity was accurate to within $\pm 10\%$ of the true value when imaging at a Doppler frame rate of approximately 115 frames per cardiac cycle. Further improvement of the method will require a filter to attenuate off-focus echoes from non-stationary tissue.

9:00

5aBAa5. An MR-compatible fiber-optic probe for measuring focused ultrasound-induced temperature rises without viscous heating artifacts. Sara L. Johnson (Dept. of Radiology and Imaging Sci., Univ. of Utah, 804 E 300 S, Apt. 25, Salt Lake City, UT 84102, sarajay144@gmail.com), Henrik Odeen, Allison Payne (Dept. of Radiology and Imaging Sci., Univ. of Utah, Salt Lake City, UT), and Harry Vine (OSENSA Innovations, Caldwell, NJ)

Focused ultrasound (FUS) beam interactions with thermocouples and fiber optic (FO) probes cause a "viscous heating artifact" (VHA), which prevent accurate temperature measurements during acoustic sonication. This work demonstrates a novel fiber-optic temperature probe which is insensitive to VHA, validated in an MR-guided FUS treatment setting. The FO probe (OSENSA Innovations; PRB-140, 0.14 mm diameter) was inserted with an 18G catheter into a tissue-mimicking gelatin phantom. The FO probe tip was located with high-resolution MR images ($0.25 \times 0.25 \times 0.5\,\mathrm{mm}^3$) for targeting with FUS. Continuous-wave FUS sonications (50 W, 20 s) were delivered at a distance of $\sim$1.5 mm from the FO probe tip using a 256 phased-array transducer (Imasonic, France; 1 MHz, $2.1 \times 2.3 \times 9.8$ FWHM spot size) and FUS-induced heating was measured with 3D MR temperature imaging (MRTI; $0.5 \times 0.5 \times 1\,\mathrm{mm}^3$ resolution, 3.9 s acquisition). The VHA effect was not observed in the PRB-140 FO probe measurement data. The FO probe heating and cooling curves closely matched those measured by MRTI, with a root mean-squared error (RMSE) of 0.61 °C. In contrast, the RMSE of VHA-sensitive FO probe was 6.00 °C for a similar acoustic power output. This ultrasound artifact-immune FO temperature probe is highly advantageous for MR temperature sequence development and precise temperature monitoring in FUS treatment applications.

9:15

5aBAa6. A pipeline to generate large-scale, realistic cardiac ultrasound recordings including clinically relevant artefacts. Nitin Burman (Dept. of Cardiovascular Sci., Katholieke Universiteit Leuven, UZ Herestraat 49 - Box 7003, Leuven 3000, Belgium, nitin.burman@kuleuven.be), Sophie V. Heymans (Dept. of Cardiovascular Sci., Katholieke Universiteit Leuven, Kortrijk, Belgium), Claudia Manetti, Joost Lumens (Faculty of Health, Medicine and Life Sci., Maastricht Univ., Maastricht, Netherlands), and Jan D'hooge (Dept. of Cardiovascular Sci., Katholieke Universiteit Leuven, Leuven, Belgium)

Simulated ultrasound (US) data are widely used to develop and validate (machine learning-based) US data processing algorithms. In this regard, the quantity and quality of the simulated US data are crucial. Here, we have developed an US simulation pipeline to generate realistic cardiac US recordings on a large scale. In this pipeline, we used clinical cardiac US scans to sample the echogenicity of the US scattering sites. In parallel, a non-linear US simulator, k-wave, was employed to generate clinical artifacts due to the presence of ribs and lungs, including reverberation and shadowing. The position of the ventricle, the probe, and the simulated artifact data were then spatially registered in order to modify the originally sampled echogenicities. Motion of the myocardial scattering sites was kinematically governed by a stable mechanical heart model (CircAdapt). The resulting dynamic echogenicity map was fed into a fast convolution-based ultrasound simulator (COLE) to generate cardiac US recordings with clinical appearance including artefacts. The generated US data follow realistic speckle statistics and is a potential augmentation tool for machine learning based US data processing algorithms. [Work funded by European Union's Horizon 2020 research and innovation programme under the Marie Sklodowska-Curie grant agreement No. 860745.]

9:30

5aBAa7. Optimizing intracardiac flow assessment with high frequency ultrasound in a mouse model. Geraldi Wahyulaksana (Radiology, Weill Cornell Medicine, 416 E 55th St., New York, NY 10022, gew4002@med.cornell.edu), Colin K. L. Phoon (Div. of Pediatric Cardiology, Hassenfeld Children's Hospital at NYU Langone, New York, NY), Glenn I. Fishman (Leon H. Charney Div. of Cardiology, NYU Langone Health, New York, NY), and Jeffrey A. Ketterling (Radiology, Weill Cornell Medicine, New York, NY)

Cardiomyopathy, a disorder affecting the heart muscle, is associated with severe symptoms, such as heart failure, arrhythmia, or cardiac arrest. Echocardiography is the primary clinical tool for myocardial assessment, which provides diagnostic and prognostic insights. However, current clinical parameters, such as myocardial strain, often detect the condition only after symptoms become pronounced, showing the need for early-stage assessment. Based on preliminary studies, intracardiac flow patterns have the potential to detect abnormal myocardial changes earlier than the current clinical parameters. Mouse models are commonly imaged with ultrasound in the cardiovascular disease research but their use for flow pattern evaluation has been hindered by spatial and temporal resolution limitations. Here, we address this limitation by conducting data acquisition using a Verasonics scanner with a range of high-frequency probes (16 to 42 MHz), transmit cycles, and transmit angles. A singular value decomposition (SVD) filter and frequency de-aliasing were also investigated to optimize flow quantification. Our findings show that a higher transmission frequency, while providing superior spatial resolution, may not yield the best flow quantification due to increased Doppler aliasing. Furthermore, SVD filtering and Doppler de-aliasing should be used with caution because improper implementation can adversely impact the quantification results.

Session 5aBAb

Biomedical Acoustics: General Topics in Biomedical Acoustics: Drug Delivery and Therapy

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

Contributed Papers

8:00

5aBAb1. Ultrasound-triggered *in situ* hydrogel formation for spinal disc repair. Veerle A. Brans (Inst. of Biomedical Eng., Univ. of Oxford, Inst. of Biomedical Eng. (Botnar Res. Centre), Old Rd., Oxford OX3 7LD, United Kingdom, veerle.brans@eng.ox.ac.uk), Anna P. Constantinou (Dept. of Mater., Imperial College London, London, United Kingdom), Matthew J. Kibble, Nicolas Newell (Dept. of Bioengineering, Imperial College London, London, United Kingdom), Luca Bau (Inst. of Biomedical Eng., Univ. of Oxford, Oxford, United Kingdom), Molly M. Stevens (Dept. of Mater., Imperial College London, London, United Kingdom), Constantin Coussios (Inst. of Biomedical Eng., Univ. of Oxford, Headington, Oxford, Oxfordshire, United Kingdom), and Michael Gray (Inst. of Biomedical Eng., Univ. of Oxford, Oxford, United Kingdom)

Over 600 million people suffer from lower back pain attributable to spinal disc degeneration. Treatments range from conservative physiotherapy to costly, invasive options like spinal fusion surgery. Injectable engineered materials may allow minimally invasive disc repair, but self-curing lacks process control, and optical *in situ* curing is depth-limited. We, therefore, propose ultrasound-triggered implant formation, enabling spatiotemporal control at clinically relevant tissue depths. We developed an anionic polysaccharide-based hydrogel, seeded with calcium-loaded thermosensitive liposomes. Focused ultrasound was used to heat this injectable precursor material to just above the lipid phase-transition temperature of 41 °C, inducing calcium release and ionically crosslinked network formation. Controlled heating was achieved by elevating the acoustic attenuation of the precursor solution with purified glass microspheres. The heating and gelation processes were controlled in real-time using thermometry and acoustic cavitation emissions. We optimized the ultrasound frequency and pressure amplitude to provide controlled heating with minimal cavitation. After these parameters were employed for ultrasound-mediated gelation, the rheological properties of the resultant gels were compared to literature values for native spinal disc material. Finally, *in situ* gel formation was evaluated in *ex vivo* bovine tail discs, from injection to mechanical assessments, confirming the ability to remotely trigger injectable disc-mimicking materials.

8:15

5aBAb2. Precise, low-intensity triggered release in biological systems using ultrasound-responsive antibubbles. Athanasios Athanassiadis (Heidelberg Univ., Im Neuenheimer Feld 225, Heidelberg 69120, Germany, thanasi@uni-heidelberg.de), Nicolas Moreno-Gomez, Dimitris Missirlis, Yannik L. Trautnitz, and Peer Fischer (Heidelberg Univ., Heidelberg, Germany)

Antibubbles are an emerging carrier for low-intensity, ultrasound-triggered release that can carry large payloads in their liquid cores. As we have previously shown, antibubbles can release their payloads at adjustable pressures, ranging from as low as a few kPa (MI < 0.01) to above a few hundred kPa (MI > 0.2). The specific release threshold and behavior can be determined during fabrication, making it possible to realize either single-shot release or multi-stage dosing of a payload. These characteristics make antibubbles an attractive alternative to conventional microbubble delivery agents particularly in sensitive tissues where risks of ultrasound need to be kept to a minimum. In this talk, we discuss recent experiments in which we demonstrate triggered release from antibubbles in *in vitro* biological systems. We show that payload release can be controlled using patterned or focused ultrasound fields, and we characterize the distribution and cellular uptake of payloads inside cell-laden matrices. These results not only demonstrate the potential of antibubbles for therapeutic applications but also open the door to targeted delivery in complex tissue scaffolds for tissue engineering.

8:30

5aBAb3. Targeted delivery of miR-1 to the heart using clinical contrast ultrasound. Davindra Singh (Biology, Concordia Univ., 1455 de Maisonneuve Blvd., Montreal, QC H3G 1M8, Canada, davindra.singh@mail.concordia.ca), Stephanie He (Biology, Concordia Univ., Montreal, QC, Canada), and Brandon Helfield (Phys., Concordia Univ., Montreal, QC, Canada)

Pathological left-ventricle hypertrophy is a cardiovascular disorder resulting in the thickening of the ventricle wall due to abnormal cardiomyocyte growth. The downregulation of miR-1 in hypertrophic cardiomyocytes has been identified as an early disease marker, with the delivery of miR-1 as a promising treatment strategy. Ultrasound and microbubbles offer an exciting approach to image-guided and site-specific cardiac gene delivery. The objective of this study is to show the feasibility of viable ultrasound and microbubble-mediated delivery of miR-1 *in vivo*. Sprague–Dawley rats were injected via tail vein with a suspension of miR-1 mimic (0.6 mg/kg) and Definity. The rats were then treated with a high MI (1.34) flash sequence for 20 min using a C5-2 probe with a Phillips iU22. Delivery was confirmed on isolated heart tissue with RT-qPCR and Western blots for miR-1 and protein expression, respectively. miR-1 delivery in healthy male rats resulted in a 1.33-fold ($p = 0.05$) increase in miR-1 as compared to sham controls. This resulted in a decrease in hypertrophic protein expression (1.18-fold in TWF1, $p = 0.17$; 1.23-fold in MEF2A, $p = 0.07$; 1.25-fold in CX43, $p = 0.03$). Our data demonstrate the feasibility of using ultrasound and microbubbles as an image-guided delivery method for molecular therapeutics in cardiovascular disorders.

8:45

5aBAb4. Monodisperse microbubble-mediated drug delivery: Influence of microbubbles size on drug delivery outcome. Yuchen Wang (Erasmus MC, Dr. Molewaterplein 40, Rotterdam 3015 GD, Netherlands, y.wang@erasmusmc.nl), Hongchen Li (Biomedical Eng., Erasmus MC, Rotterdam, Netherlands), Bram Meijlink (Biomedical Eng., Erasmus MC, Utrecht, Utrecht, Netherlands), Jiali Luo, Robert Beurskens (Biomedical Eng., Erasmus MC, Rotterdam, Netherlands), Benjamin Johnson (Univ. of Leeds, Leeds, United Kingdom), and Klazina Kooiman (Erasmus MC, Rotterdam, Netherlands)

As the microbubble's resonance frequency is size-dependent, polydisperse microbubbles induce varying drug delivery outcomes at one ultrasound frequency. This study aimed to investigate drug delivery outcome of monodisperse MBs (mMBs) with radii ranging from 1.5–2.9 μm insonified at 2 MHz. Phospholipid-coated mMBs were generated using the Horizon microfluidic flow-focusing device. *In vitro* experiments were conducted on single microbubble-endothelial cells ($n = 68$) using confocal microscopy and 10 Mfps ultra-high-speed imaging. At 220 kPa PNP for 10 cycles, the 1.5 μm mMBs exhibited the highest PI uptake in 81.3% of cases, followed by 77.3% for 2.2 μm, 36.8% for 2.7 μm and 13.3% for 2.9 μm mMBs. Conversely, the 1.5 μm mMBs had the second lowest excursion amplitude ($R_{;max}$-$R_{;0}$) of 0.8 ± 0.3 μm as this was 1.2 ± 0.3 μm for 2.2 μm mMBs, 1.0 ± 0.3 μm for 2.7 μm mMBs, and 0.7 ± 0.2 μm for 2.9 μm mMBs. Additionally, for the 1.5, 2.2, and 2.7 μm mMBs, PI uptake and tunnel formation occurred more often (1.6, 1.8, and 1.3 times, respectively) than PI uptake and a resealing membrane pore. During insonification, mMB pinch-off occurred more frequently in tunnel formation (70.8%) than resealing pore formation (53.3%). This research revealed the impact of mMBs size on drug delivery outcome.

9:00

5aBAb5. Focused ultrasound-guided delivery of gene editing protein in human induced pluripotent stem cells. Kyle Hazel (Biology, Concordia Univ., 7141 Sherbrooke St W, Montreal, QC H4B 1R6, Canada, kyle.hazel@hotmail.com), Davindra Singh, Mathieu Husser, Elahe Memari, Stephanie He (Biology, Concordia Univ., Montreal, QC, Canada), and Brandon Helfield (Phys., Concordia Univ., Montreal, QC, Canada)

Focused ultrasound (FUS) in combination with microbubbles (MBs) can induce cavitation-mediated plasma membrane permeabilization in nearby cells, thus permitting entry to otherwise impermeable macromolecules. Cas9 is an endonuclease protein currently at the forefront of gene editing due to its efficiency, ease of use, and low cost. In complex with single guide RNA (sgRNA), Cas9 can target specific gene sequences to cause double-stranded breaks, interrupting gene function in the process. Cas9 is best delivered as a ribonucleoprotein (RNP) for the most effective results, however, suffers from inefficient delivery methods for *in-vivo* applications due to its large size (160 kDa). FUS and microbubbles can be an effective alternative to currently applied systems (e.g., adeno-associated vectors) to deliver Cas9:sgRNA RNPs for CRISPR-mediated knockout. Currently, we are exploring the use of FUS for Cas9-mediated knockout of EGFP in both EGFP-expressing human induced pluripotent stem cells (hiPSC) and human cardiomyocytes. Treatment of hiPSC under acoustic conditions (1 MHz, 1000 cycles, 5 ms intervals, >208 kPa) suitable for cavitation-mediated sonoporation led to EGFP knockout in hiPSCs. By modulating FUS parameters and setup, we can optimize delivery of Cas9 as an RNP for treatment of genetic diseases, such as hypertrophic cardiomyopathy.

9:15

5aBAb6. Using ultrasound-targeted microbubble cavitation to open the blood–brain barrier for drug delivery in Alzheimer's disease. Grace E. Conway (Univ. of Pittsburgh, 3550 Terrace St., Scaife Hall 969.3, Pittsburgh, PA 15213, gec36@pitt.edu), Xucai Chen, Stacey J. Rizzo, Afonso C. Silva (Univ. of Pittsburgh, Pittsburgh, PA), and Flordeliza S. Villanueva (Medicine/Cardiology, Univ. of Pittsburgh, Pittsburgh, PA)

Drug delivery for Alzheimer's disease (AD) is challenging due to restricted diffusion across the blood–brain barrier (BBB). Ultrasound-targeted microbubble cavitation (UTMC) can be used to transiently open the BBB. We hypothesized that opening the BBB with UTMC would increase the brain concentration of LY2886721 (LY), an orally administered drug developed for AD, in 5XFAD mice, a model of AD. We treated the right hemisphere using an RK-50 system (FUS Instruments Inc). Definity microbubbles were injected, and ultrasound (1.45 MHz, 2 Hz pulse repetition frequency, 10 ms pulse length, 0.6 MPa for 30 s) was applied to 11 locations across the right hemisphere. Thirty minutes after UTMC, mice were administered LY (30 mg/kg p.o.). In one cohort, 45 min after dosing with LY, there was an increase in the concentration of LY in the UTMC-treated hemisphere compared to the non-UTMC treated hemisphere ($p < 0.05$). In a second cohort, 2.5 h after dosing with LY, there was a decrease in soluble $A\beta_{;40}$ ($p < 0.05$), insoluble $A\beta_{;40}$ ($p < 0.05$), and insoluble $A\beta_{;42}$ ($p < 0.05$) in the UTMC-treated hemisphere compared to the non-UTMC treated hemisphere. UTMC can be used to open the BBB to increase the concentration of LY and decrease $A\beta$ in a preclinical model of AD.

9:30

5aBAb7. The activation of endothelial nitric oxide synthase, induced by calcium influx plays a crucial role in regulating endothelial hyperpermeability caused by ultrasound-targeted microbubble cavitation. Anurag N. Paranjape (Medicine/Cardiology, Univ. of Pittsburgh, 3550 Terrace St., Pittsburgh, PA 15217, anuragnparanjape@gmail.com), Xucai Chen, and Flordeliza S. Villanueva (Medicine/Cardiology, Univ. of Pittsburgh, Pittsburgh, PA)

Ultrasound-targeted microbubble cavitation (UTMC) plays a crucial role in improving drug delivery across the endothelial barrier. For successful application of UTMC in clinic, an understanding of molecular mechanisms involved is essential. Here, we hypothesized that Ca^{2+} and endothelial nitric oxide synthase (eNOS) pathways regulate UTMC-induced endothelial hyperpermeability. We used human coronary artery endothelial cells seeded on transwell inserts, exposed to microbubbles and ultrasound (frequency 1 MHz; PNP 250 kPa; pulse length 10 μs; pulse interval 10 ms; treatment duration 10 s). UTMC caused cellular influx of Ca^{2+}, which was inhibited by blocking mechanosensitive channels ($p < 0.0001$) using GsMTx4. Knockdown of Piezo1 using siRNA showed similar effects. UTMC-induced Ca^{2+} influx activated eNOS and enhanced nitric oxide production, which was necessary for UTMC-induced hyperpermeability. Our data suggest that UTMC induces a Ca^{2+} influx-dependent increase in S-nitrosylation: An increase in S-nitrosylation was observed on both β-catenin and VE-cadherin after UTMC, potentially contributing to the destabilization of adherens junctions that normally maintain barrier integrity. Our study explains the role of Ca^{2+} influx, eNOS activation, and enhanced S-nitrosylation of adherens junction proteins in the regulation of endothelial hyperpermeability. Further investigation of these pathways will aid in clinical translation and optimization of UTMC for delivering cell-impermeant drugs.

9:45–10:00 Break

10:00

5aBAb8. Towards real-time decompression sickness mitigation using wearable capacitive micromachined ultrasonic transducer arrays. Joshua B. Currens (Joint Dept. of Biomedical Eng., UNC - Chapel Hill, 10010 Mary Ellen Jones, Campus Box 7575, Chapel Hill, NC 27599, jcurrens@unc.edu), Muhammetgeldi Annayev, Remzi Erkan Kemal (Elec. and Comput. Eng., North Carolina State Univ., Raleigh, NC), Katherine M. Eltz, Arian Azarang (Joint Dept. of Biomedical Eng., UNC - Chapel Hill, Chapel Hill, NC), Michael Natoli, Rachel M. Lance, Richard E. Moon (Ctr. for Hyperbaric Medicine and Environ. Physiol., Duke Univ., Durham, NC), Paul A. Dayton (Joint Dept. of Biomedical Eng., UNC - Chapel Hill, Chapel Hill, NC), Feysel Yalcin Yamaner (Clearsens Inc., Durham, NC), Omer Oralkan (Elec. and Comput. Eng., North Carolina State Univ., Raleigh, NC), and Virginie Papadopoulou (Joint Dept. of Biomedical Eng., UNC - Chapel Hill, Chapel Hill, NC)

Decompression sickness (DCS) due to inert gas supersaturation remains one of the major risks for scuba divers and can occur despite adherence to prevention schedules for staged decompression. Post-dive echocardiography for venous gas emboli (VGE) detection has low sensitivity for DCS outcome and is unable to provide real-time physiological monitoring underwater. Alternatively, we present progress towards collecting ultrasound data while at pressure and an exploration into a quantitative assessment for

decompression stress. Ultrasound data of an imaging phantom were collected in a hyperbaric chamber up to 9 ATA using a Verasonics V1 system and custom capacitive micromachined ultrasonic transducer (CMUT). The DC voltage requirement for CMUT operation decreased as ambient pressure increased. Separately, a mouse model was used to simulate an extreme pressure profile and echocardiograms were collected every 20 min over 2-h post-decompression. Towards enhancement of DCS assessment, a quantitative analysis of the murine echocardiograms was implemented. Preliminary results show an increase in signal intensity within the venous blood from pre- to post-dive, indicating potential gas presence despite VGE absence. Our findings demonstrate the ability to obtain ultrasound data at pressure and a potential continuous assessment method, which may provide a practical direction for real-time decompression stress quantification.

10:15

5aBAb9. Designing a benign prostatic hyperplasia dual-mode cavitation cloud and boiling histotripsy therapy transducer. Yashwanth Nanda Kumar (Ctr. for Industrial and Medical Ultrasound, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, ynandaku@uw.edu), Kaizer Contreras, Yak-Nam Wang, Wayne Kreider, Stephanie Totten (Ctr. for Industrial and Medical Ultrasound, Univ. of Washington, Seattle, WA), George R. Schade (Dept. of Urology, Univ. of Washington, Seattle, WA), and Adam D. Maxwell (Ctr. for Industrial and Medical Ultrasound, Univ. of Washington, Seattle, WA)

Benign prostatic hyperplasia (BPH) is a condition that causes an enlarged prostate leading to lower urinary tract symptoms (LUTS), affecting quality of life. Histotripsy is a non-invasive focused ultrasound technique that mechanically disintegrates tissue wherein cavitation cloud (CH) and boiling histotripsy (BH) are the two commonly used therapy modes. Recent studies have shown successful ablation of BPH tissue *in vitro* by transducers that were designed for a transabdominal approach. However, this approach may not be successful in treating tissue *in vivo*, as evidenced in some of the clinical studies, due to challenges with the anatomical position of prostate within humans. Therefore, a need exists in treating it transrectally with a transducer that can perform both CH and BH for efficient ablation. Initial design studies were performed to determine the appropriate source parameters and materials needed for fabricating the combinational therapy transducer for *in vivo* use. A 3 MHz multi-element source with a focal length of 40 mm and gain of 64 was chosen. The equivalent aperture is 48.5 mm with a 10 mm central opening for image guidance. Non-linear simulations show sufficient peak positive and negative pressures can be achieved for performing both regimes. Further details on the fabrication and characterization of the transducer with benchtop results will be presented.

10:30

5aBAb10. Correlation of *Escherichia coli* inactivation with histotripsy bubble cloud size. Pratik A. Ambekar (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, pambek@uw.edu), Tatiana D. Khokhlova (Dept. of Medicine, Univ. of Washington, Seattle, WA), Yak-Nam Wang (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Pavel B. Rosnitskiy (Appl. Phys. Lab., Univ. of Washington, Moscow, Russian Federation), Daniel Leotta, Gilles P. Thomas, Stephanie Totten, Shelby Pierson, Matthew Bruce (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Adam D. Maxwell (Urology, Univ. of Washington, Seattle, WA), Keith Chan (Vantage Radiology and Diagnostic Services, Seattle, WA), W. Conrad Liles (Dept. of Medicine, Univ. of Washington, Seattle, WA), Evan P. Dellinger (Dept. of Surgery, Univ. of Washington, Seattle, WA), Adeyinka Adedipe (Dept. of Emergency Medicine, Univ. of Washington, Seattle, WA), Wayne Monsky (Dept. of Radiology, Univ. of Washington, Seattle, WA), and Thomas Matula (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Bacterial loads can be effectively reduced with cavitation-mediated focused ultrasound, or histotripsy. Our previous *in vitro* work with *Escherichia coli* (*E. coli*) established strong trends of bactericide with increasing peak negative pressure amplitude, pulse length, PRF, and treatment time. The current study correlates bactericide as a function of the histotripsy bubble cloud size produced for these treatment parameters at several frequencies. Histotripsy was applied to *E. coli* suspensions in 10-ml sample vials at 810 kHz, 1.2 MHz, or 3.25 MHz for 40 min. Separately, cavitation was recorded using a Photron Fastrax high-speed camera for acoustic parameters equivalent to those used in the *E. coli* studies. The images were used to quantify the maximum size of each bubble cloud, with the assumption that the cloud was axially symmetric around the propagation direction. A strong linear relationship exists between log kill versus cloud size ($R^2 = 0.96$). Remarkably, across all variables studied, the log-reduction in viable bacteria exhibited a direct proportional relationship with the dimensions of the bubble cloud. This strong correlation between bacterial reduction and cavitation bubble cloud size could have significance for clinical applications where bacteria cannot easily be sampled. [Work supported by NIH R01AR080120 and R01EB023910.]

10:45

5aBAb11. Ultrasound diagnosis and treatment of heterotopic ossification. Fea Morgan-Curtis, Lucas Ruge-Jones, Grace M. Wood (Graduate Program in Acoust., Penn State, University Park, PA), Lisa Berntsen (Dept. of Biomedical Eng., Penn State, University Park, PA), Jacob C. Elliott (Graduate Program in Acoust., Penn State, Res. West, State College, PA 16801, jce29@psu.edu), Daniel Hayes (Dept. of Biomedical Eng., Penn State, University Park, PA), and Julianna C. Simon (Graduate Program in Acoust., Penn State, University Park, PA)

Heterotopic ossification (HO), or the presence of bone in soft tissues, can occur after musculoskeletal trauma, causing pain and reduced mobility. However, even the most sensitive diagnostic modality requires 2–3 weeks after initiation to detect HO, and the only treatment is surgical resection after HO matures (>2 years). Here, we evaluate the color Doppler ultrasound twinkling artifact for early diagnosis of HO and focused ultrasound (fUS) for treatment of early HO. We began by evaluating twinkling and fUS parameters in ossified cell culture. We then evaluated imaging and treatment parameters in mice with HO. Results show that twinkling correlated well with the presence of ossified cells; although the exact size/number of ossified cells required for twinkling was unclear. In mice, twinkling was found to detect HO in some mice as early as 3 days-post-injection, which is earlier than other imaging modalities. For treatment, fUS at 1.07-MHz and 0.2% duty-cycle with $p+/p- = 17/7$ MPa was found to be most successful at disrupting mineralizations with little damage to surrounding cells. Mice treated with fUS were found to have less advanced HO compared to sham mice. These results highlight the promise of ultrasound for the diagnosis and treatment of early HO. [Work supported by CDMRP PR201164].

11:00

5aBAb12. The roles of pulse length and duty cycle in the fractionation of tendinopathic tendons. Grace M. Wood (Graduate Program in Acoust., Penn State, 1 Old Main, University Park, PA 16802, gmw5253@psu.edu), Jacob C. Elliott, and Julianna C. Simon (Graduate Program in Acoust., Penn State, University Park, PA)

Collagenous tissues, like tendon, are resistant to fractionation by focused ultrasound (fUS). Prior work has shown thermal pretreatments increase susceptibility to fractionation by fUS, suggesting the potential for successful fractionation by tuning boiling histotripsy parameters. Here, we investigate the roles of pulse length (PL) and duty cycle (DC) on fUS treatments of tendinopathic tendons. *Ex vivo* bovine tendons were injected with collagenase to induce tendinopathy. The following day, tendons were treated for 15–20 min with 1–3 MHz fUS ($p_{;+} \leq 127$ MPa/$p_{;-} \leq 35$ MPa). Three PLs were chosen below, slightly above, and well above the calculated time-to-boil. The pulse repetition frequency varied between 0.2–1 Hz, allowing for evaluation of DCs between 0.1% and 0.8%. Preliminary results show focal fiber separation at the PL slightly above the calculated time-to-boil; areas of tissue disruption are smaller and less frequent for the other PLs and accompanied by thermal damage at the highest PL. As frequency increases, the tissue disruption becomes smaller and infrequent for the lowest and highest PLs. When the PL was held constant, minimal change in tissue fractionation was found when changing DC. Thus, for the tested combinations, PL is more influential than DC for tuning the fractionation of tendinopathic tendons. [Work supported by NIH RO1EB032860.]

11:15

5aBAb13. Tendon as a model for testing the efficacy of histotripsy for chronic deep vein thrombosis. Kevin Zhao (Radiology, Univ. of Chicago, 5841 South Maryland Ave., Chicago, IL 60637, kzhao@som.geisinger.edu), Erik Saucedo, and Kenneth B. Bader (Radiology, Univ. of Chicago, Chicago, IL)

Histotripsy is a focused ultrasound therapy under development for multiple diseases, including deep vein thrombosis (DVT). Benchtop studies to gauge the efficacy of histotripsy for chronic DVT present a challenge due to pathologic changes in composition that are difficult to replicate *in vitro*. Tendon is a readily available tissue with an extensive extracellular matrix similar to chronic DVT. This study aims to assess histotripsy-induced changes to tendon at driving parameters effective for acute thrombus ablation. Porcine patellar tendons were bisected into control ($N = 19$) and treatment ($N = 21$) segments. Histotripsy pulses were applied for 1 to 20 min at rates of 250 or 500 Hz. The pulse duration (6.7 μs) and peak negative pressure (35 MPa) were previously shown to be effective for acute thrombus ablation. Treated specimens exhibited gross swelling in targeted regions. Manual segmentation of ultrasound images was used to estimate the area of tissue affected by histotripsy, which was found to expand by up to 30% over the exposure durations for both pulsing rates. Histological analysis revealed loss of linear fiber organization and liquefaction within treated regions. Overall, these data indicate histotripsy effectively disrupts tendon integrity, making it a potential tool for the management of chronic DVT.

11:30

5aBAb14. Active targeting of nanotherapeutics using power cavitation imaging with a linear array transducer. Kamso Onyemeh (Biomedical Imaging Program, Weill Cornell Graduate School of Medical Sci., 1300 York Ave., New York, NY 10065, kao4006@med.cornell.edu), Mark Burgess (Dept. of Medical Phys., Memorial Sloan Kettering Cancer Ctr., New York, NY), Raashed Raziuddin (Pharmacology Program, Weill Cornell Graduate School of Medical Sci., New York, NY), and Daniel Heller (Molecular Pharmacology Program, Memorial Sloan Kettering Cancer Ctr., New York, NY)

Power cavitation imaging (PCI) is an emerging strategy for monitoring blood–brain barrier (BBB) opening, enabling spatial discrimination of cavitation intensity through power Doppler-analogous processing of passive cavitation images. Single-element cavitation detectors, while beneficial in signal monitoring for cavitation controllers, provide limited spatial information, conferring the need for an image-guided approach to ensure accurate cavitation localization and regulation. This study aims to evaluate the capability of PCI to spatially correlate real-time acoustic cavitation emissions with mechanical bioeffects as a predictor of P-selectin-targeted nanocarrier delivery. Preliminary sonoporation experiments were performed *in vitro* with brain tissue derived mouse endothelial cells and 3 kDa tetramethylrhodamine (TMR)-dextran. A focused ultrasound transducer (0.5 MHz, 5000 cycles/pulse, 5 Hz PRF, 30 second treatment duration) was used to sonicate cells in small volume (25 μl) cell suspension. Homogenous distribution of TMR-dextran in the cytosol and nucleus was observed via fluorescence microscopy at lower pressures (7% $\pm$ 2% fluorescent cells), while higher pressures had no significant differences above control (3% $\pm$ 2% vs 1% nontreated) due to microbubble destruction. Future experiments will correlate sonoporation with P-selectin expression and validate a linear array PCI system against contrast-enhanced MRI as a guide for P-selectin-targeted nanocarrier delivery in mouse models of medulloblastoma.

Session 5aNS

Noise, Structural Acoustics, and Vibration and Engineering Acoustics: Pickleball Noise

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

Chair's Introduction—8:00

Contributed Papers

8:05

5aNS1. Is pickleball a good neighbor? Evelyn Way (Maxxon Corp., 920 Hamel Rd., Hamel, MN 55340, eway@maxxon.com)

Upscale multi-family residences need the latest amenities to attract tenants. A recent project featured a party room, exercise facility, and pickleball court on the top floor of a low-rise, adaptive reuse of an office building to residential renovation. Pickleball courts located directly above residential units allowed for measurements of sound transmission during pickleball play and comparison to traditional airborne and impact isolation metrics.

8:20

5aNS2. Preliminary analysis of more than 60 pickleball noise consultant reports. Charles E. Leahy (Law Office - Pro Bono Public Interest, 151 E Summit St., Harbor Springs, MI 49740, charles.leahy@sbcglobal.net)

Over 60 consulting reports authored by over 30 different US and Canadian consultants have been gathered from municipal agendas, court filings, and HOA members. The reports are analyzed with respect to tools, metrics, and criteria used to make findings and recommendations to city officials, HOA Boards, and lawsuit litigants. The reports range greatly in length and purpose, many with sound meter data, excerpts from municipal ordinances, and specifications for noise mitigation including setbacks, full enclosures, barriers, and equipment recommendations. Particular attention is given to the consultant's choice of sound meter metrics and the attention or lack of attention to the impulsive nature of pickleball noise and applicability of ANSI Standard 12.9 Part 4. When a local ordinance is referenced, we examine whether the consultant limited the report to provisions specifying a decibel limit and excluded comment on the general nuisance and plainly audible standards that are often present. Anecdotes are drawn from the data to identify best practices and worst practices of the consultants and their reports. This paper may be useful to consultants in planning their future pickleball studies; useful in deciding whether to refer these projects to a more experienced pickleball noise boutique firm; and/or useful to attorneys in recognizing the challenges of using the consultant report in the litigation setting.

8:35

5aNS3. Improving the persuasiveness of the noise consultant report—A critique and proposal. Charles E. Leahy (Law Office - Pro Bono Public Interest, 151 E Summit St., Harbor Springs, MI 49740, charles.leahy@sbcglobal.net)

The author, a retired mechanical engineer and patent attorney, sat on an HOA Board that converted four tennis courts to 13 pickleball courts without installing the noise absorbing barrier recommended in three reports by two acoustical consultants. A nuisance lawsuit seeks an injunction to close the courts and money damages. Consultants study the pickleball noise and recommend mitigation options to the deciders. Deciders are laymen, such as city officials and HOA boards, unfamiliar with noise science and under pressure to meet the insatiable need for more pickleball courts. When mitigation is not implemented, or not successful, litigation ensues. Judge and jury become deciders. This paper starts from the premise that the consultant makes technically competent recommendations and is motivated to fully educate the lay deciders. We explore the communication gap between consultants and deciders. Proposals include embedding wav. noise files in the written report, reference to ANSI standards for highly impulsive noises, deference to the lack of research on health issues of long-term exposure to persistent repetitive impulsive noise. We introduce the missing metrics of "pickle pops per hour" and the "predictable worst case noise impact" occurring when intensively used courts are introduced into residential neighborhoods.

8:50

5aNS4. Challanges in enacting pickleball noise regulations. Dana S. Hougland (Shen Milsom & Wilke, LLC, 1801 Wewatta, Fl. 11, Denver, CO 80202, dhougland@smwllc.com) and Tammy Maurer (City Council, City of Centennial, Centennial, CO)

Outdoor pickleball courts installations vary widely in environmental conditions, including terrain and construction. With the rapid rise in popularity of pickleball, new court installations and conversions of existing tennis courts have generated community pressure for local governments to enact noise restrictions. Local governmental agencies have been challenged to establish regulations, which must apply across a number of unique installation conditions and terrains. This presentation examines the challenges by a selection of local governmental agencies to enact and enforce regulations.

9:05

5aNS5. Categorization of pickleball paddles as a highly impulsive sound source. Lance Willis (Spendiarian & Willis, Tucson, AZ) and Charles E. Leahy (Law Office - Pro Bono - Public Service, 151 E Summit St., Harbor Springs, MI 49740, charles.leahy@sbcglobal.net)

As the sport of pickleball has grown in popularity and tennis courts close to homes are being converted to this use, noise regulation for pickleball courts has become a topic of discussion among city councils, planners, home owners associations, residents, and, increasingly, attorneys seeking injunctions against poorly sited or mitigated courts. Accurate noise impact assessment of the short duration impulsive sound produced by the impact of the ball on the paddles requires the use of the highly impulsive adjustment in ANSI S12.9 Part 4. Examination of the definitions of impulsive sound in this standard and in ISO 1996 Part 1 identifies characteristics of rapid onset, significant energy in the most sensitive part of the auditory spectrum, and duration as primary differentiators for categorization. The noise impact assessment methodology in ANSI S12.9 Part 4 for highly impulsive sound has proven an effective means of determining the amount of mitigation needed for pickleball facilities, being more repeatable and precise than other commonly used assessment methods while avoiding underestimation of annoyance.

9:20

5aNS6. Advancements in nanotechnology for acoustic management in pickleball. Eliot Arnold (4400 Shawnee Mission Parkway, Fairway, KS 66205, earnold@slncr.com)

This research investigates the use of advanced nano-fiber technology for sound and noise management in pickleball courts. The technology, known for its flexibility and adaptability, addresses the unique acoustic challenges of pickleball, a sport with a distinctive noise profile characterized by impulsive and unpredictable sounds. These nano-fibers are particularly effective in absorbing mid to high-frequency noises (800–5000 Hz) common in pickleball. Incorporating these nano-fibers into acoustic foams and textiles significantly enhances sound absorption, allowing for thinner materials while doubling performance compared to conventional materials. These fibers, about 1/500th the diameter of human hair, have a high surface area to volume ratio, aiding in sound scattering and increased friction with air molecules. This structure enables the efficient transformation of sound energy into heat, which is then effectively dissipated. Aligned with the Acoustical Society of America's standards, this abstract emphasizes a scientific breakthrough in sports acoustics, contributing to the reduction in urban noise pollution. The study underscores the impact of cutting-edge material technology in improving environmental acoustics and community well-being.

9:35

5aNS7. Understanding pickleball noise at the source: The vibroacoustics of the pickleball paddle and ball. Daniel A. Russell (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, dar119@psu.edu)

Pickleball continues to be America's fastest-growing sport, growing by roughly 150% every year with 48.3 million US adults (19% of the adult population) having played at least one game in 2023. This immense popularity and growth is accompanied by a similar increase in the number and voracity of complaints from the residential communities surrounding pickleball courts. This paper aims to explain the source of the pickleball noise—a short duration impulse with a strong tonal component near 1250 Hz—due to the impact of the pickleball ball and paddle, with a specific focus on the vibrational modes of the paddle that contribute most strongly to the tonal nature of the sound. Experimental modal analysis was performed on several paddles (wood, aluminum, composite, and carbon fiber) as well as the first paddle to meet the new USA Pickleball Quiet Category requirements. Spectral analysis of the impact identifies the offending tonal component near 1250 Hz to be a strongly radiating "membrane-type" mode of the paddle surface. Paddles for which this mode shape has a significantly higher or lower in frequency and amplitude result in a much less annoying impact sound. The role of vibrational modes in the ball will also be considered.

9:50–10:00 Panel Discussion

FRIDAY MORNING, 17 MAY 2024 — ROOM 206, 8:00 A.M. TO 10:55 A.M.

Session 5aPA

Physical Acoustics, Biomedical Acoustics, and Structural Acoustics and Vibration: Nonlinear Acoustics in Solids

John M. Cormack, Cochair
Div. of Cardiology, Dept. of Medicine, Univ. of Pittsburgh, Pittsburgh, PA 15261

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

Invited Papers

8:00

5aPA1. Direct, model-free acoustic evaluation of stresses and strains in loaded nonlinear soft solids. Michel Destrade (School of Math. Stat. Sci., Univ. of Galway, University Rd., Galway, Galway H91 TK33, Ireland, michel.destrade@nuigalway.ie), Guoyang Li (Dept. of Mech. and Eng. Sci., Peking Univ., Beijing, China), Artur L. Gower (Mech. Eng., Univ. of Sheffield, Sheffield, United Kingdom), Yanping Cao (Dept. of Eng. Mech., Tsinghua Univ., Beijing, China), Seok-Hyun Yun (Harvard Med. School and Wellman Ctr. for Photomedicine, Massachusetts General Hospital, Boston, MA), Zhaoyi Zhang, and Ziying Yin (Dept. of Eng. Mech., Tsinghua Univ., Beijing, China)

We show analytically and experimentally that the states of stress and strain existing in a mechanically loaded elastic material can be accessed directly from elastic wave speed measurements, without having to determine, or even know, its material linear and nonlinear elastic parameters. These techniques are expected to have important applications in the health monitoring of loaded structures. Examples include stressed hydrogels, muscles, and thin membranes, such as a stretched rubber sheet, a piece of cling film ($\sim$10 μm thick), and the

animal skin of a bodhrán, a traditional Irish drum. References: Z. Zhang, G.-Y. Li, Y. Jiang, Y. Zheng, A. L. Gower, M. Destrade, and Y. Cao, "Non-invasive measurement of local stress inside soft materials with programmed shear waves," Sci. Adv. 9, eadd4082 (2023); G.-Y. Li, A.L. Gower, M. Destrade, and S.-H. Yun, "Non-destructive mapping of stress and strain in soft thin films through sound waves," Commun. Phys. 5, 231 (2022).

8:20

5aPA2. Hysteretic friction between unfused surfaces as a potential source of acoustic nonlinearity and anelasticity in additively manufactured aluminum. Ward Johnson (National Inst. of Standards and Technol., 325 Broadway, MS 647, Boulder, CO 80305, wjohnson@boulder.nist.gov), Paul Heyliger (Dept. of Civil and Environ. Eng., Colorado State Univ., Fort Collins, CO), Jake Benzing, Orion L. Kafka, Newell H. Moser (National Inst. of Standards and Technol., Boulder, CO), Derek Harris, Jeremy Iten (Elementum 3D, Erie, CO), and Nik W. Hrabe (National Inst. of Standards and Technol., Boulder, CO)

Acoustic nonlinearity and loss determined by noncontacting nonlinear reverberation spectroscopy at resonant frequencies near 670 kHz are found to be anisotropic and correlated with porosity in commercially pure additively manufactured (AM) aluminum. These results, combined with results from Ritz vibrational modeling, x-ray computed tomography, and scanning electron microscopy, point towards a predominant role of acoustic shear displacement gradients across lack-of-fusion (LOF) defects. The nonlinearity and loss are also found to be dependent on accumulated duration of mode-specific acoustic excitation. Potential candidates for nonlinearity and anelasticity localized at LOF defects include dislocations and hysteretic friction/contact between unfused surfaces. Maps of geometrically necessary dislocation (GND) densities determined from electron backscatter diffraction provide no evidence of greater GND densities near LOF defects or significant dependence on porosity, and electron channeling contrast imaging similarly provides no evidence for greater dislocation densities within grains near LOF defects. In light of these results on dislocations, hysteretic friction of contacting surfaces within LOF defects is suggested as the most likely source of anisotropy and porosity dependence of the nonlinearity and loss. Under this hypothesis, the dependence on duration of acoustic excitation may be attributed to reconfiguration of surface asperities under dynamic stress.

8:40

5aPA3. Plasticity in nonlinear elastic metamaterials: Low-order dynamic modeling. Samuel P. Wallen (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, sam.wallen@utexas.edu), Michael R. Haberman (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Washington DeLima (Honeywell FM&T, KCNSC, Kansas City, MO)

Nonlinear elastic metamaterials have been shown to admit a variety of rich, dynamical features that can be leveraged to tailor the propagation of mechanical waves. Since these materials derive their properties from intricate, subwavelength geometries, direct numerical simulations are often prohibitively expensive at scales of interest. To overcome this limitation, reduced-order models, typically in the form of effective continua or discrete lattices that capture the essential features of the material at sufficiently long wavelengths, have been developed. While many prior studies have implemented these models successfully, the vast majority have considered only recoverable elastic deformations with linear damping and neglected history-dependent effects, such as plasticity and friction. In this presentation, we introduce an effective lattice modeling framework for nonlinear elastic metamaterials undergoing plastic deformation. Due to the history-dependent nature of plasticity, this framework generally yields a system of differential-algebraic equations whose computational cost is significantly greater than a purely elastic system of similar size. We apply the method to several examples of interest and explore means to obtain phenomenological elastic-plastic models for general material architectures.

9:00

5aPA4. Slow dynamics and the role of moisture. John Yoritomo (Acoust., Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375, john.yoritomo@nrl.navy.mil)

The elastic behavior of rocks and other composite materials cannot be entirely captured by the traditional theory of nonlinear elasticity, where the stress field is related to powers of the strain field. Rather, these materials display non-classical nonlinear elastic behavior, such as hysteresis, end-point memory, fast dynamics, and slow dynamics. Slow dynamics (SD) is characterized by a drop in material stiffness due to a minor mechanical conditioning, followed by a slow recovery back to the original macroscopic elastic state. SD has drawn particular attention because the recovery, often logarithmic in time, has been observed in a wide variety of materials, on lengths scales from the laboratory to the seismic, and on time scales from milliseconds to years. The universal character suggests a simple, fundamental mechanism for the SD recovery. This talk will present recent experiments that seek to test proposed SD mechanisms. In particular, the role of moisture will be investigated. The main experimental venue is the simplified structure introduced in previous work—a single bead confined between two slabs of a similar material. The main benefit of this structure over more commonly studied SD materials (sandstones and concrete) is the ability to control the environment around the contact points.

Contributed Papers

9:20

5aPA5. Wave transmission in 2D nonlinear granular-solid composite systems. Chongan Wang (Dept. of Mech. and Aerosp. Eng., Univ. of California San Diego, 3353 Lebon Dr., 101, San Diego, CA 92122, robertqhd1@gmail.com), Qifan Zhang (Wuhan Univ. of Technol., Wuhan, Hubei, China), Sameh Tawfick, and Alexander F. Vakakis (Univ. of Illinois at Urbana Champaign, Urbana, IL)

Wave propagation in granular media composed of contacting discrete elastic granules attracted considerable attention due to its highly tunable acoustic properties. While prior investigations focused on granular systems with natural boundaries (e.g., fixed or free), our study delves into two-dimensional nonlinear wave propagation within hybrid metamaterials composed of ordered granular media interacting with linearly elastic solids, employing discrete element modeling for the granular media and finite element analysis for the elastic solids. Challenges resulting from numerical instabilities arise from non-smooth contact nonlinearities and friction at granular and granular–solid interfaces. To tackle the numerical challenges, we developed an interrelated interpolation-iteration algorithm with a self-adaptive time scheme. Special consideration was given to the convergence

of contact forces at the granular–solid interface. Specifically, to ensure robust computational modeling without numerical instabilities, we monitored the eigenvalues of appropriately defined local maps governing the iterative computations of the contact forces. We demonstrated the capacity of these hybrid metamaterials for shock mitigation and non-reciprocal acoustics with properties that are passively tuned (self-adaptive with) energy. Such results and computational methods contribute to the predictive modelling and design of 2D granular media with flexible interfaces, with diverse applications, e.g., in shock protectors and acoustic diodes.

9:35–9:55 Break

9:55

5aPA6. Elastic bit and Berry phase: Investigating topological phenomena in a classical granular network. M. Arif Hasan (Mech. Eng., Wayne State Univ., 5050 Anthony Wayne Dr., Detroit, MI 48202, Hasan.Arif@Wayne.Edu) and Kazi Tahsin Mahmood (Mech. Eng., Wayne State Univ., Detroit, MI)

This study investigates the Berry phase, a key concept in classical and quantum physics, and its manifestation in a classical system. We achieve controlled accumulation of the Berry phase by manipulating the elastic bit (a classical analogue to a quantum bit) in an externally driven, homogeneous, spherical, nonlinear granular network. This is achieved through the classical counterpart of quantum coherent superposition of states. The elastic bit's state vectors are navigated on the Bloch sphere using external drivers' amplitude, phase, and frequency, yielding specific Berry phases. These phases distinguish between trivial and nontrivial topologies of the elastic bit, with the zero Berry phase indicating pure states of the linearized granular system and the nontrivial π phase representing equal superposed states. Other superposed states acquire different Berry phases. Crucially, these phases correlate with the structure's eigenmode vibrations: trivial phases align with distinct, in-phase, or out-of-phase eigenmodes, while nontrivial phases correspond to coupled vibrations where energy is shared among granules, alternating between oscillation and rest. Additionally, we explore Berry's phase generalizations for non-cyclic evolutions. This research paves the way for advanced quantum-inspired sensing and computation applications by utilizing and controlling the Berry phase.

10:10

5aPA7. Predicting second-harmonic generation in shear wave beams in tissue-mimicking phantoms. Philip G. Kaufinger (Appl. Res. Labs., University of Texas at Austin, Austin, TX 78758, pkaufinger@utexas.edu), John M. Cormack (Div. of Cardiology, Dept. of Medicine, Univ. of Pittsburgh, Pittsburgh, PA), Kyle S. Spratt, and Mark F. Hamilton (Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX)

Planar nonlinear shear waves in isotropic media are subject to only cubic nonlinear effects and generate only odd harmonics during propagation. However, wavefront curvature in shear wave beams breaks the symmetry and yields quadratic nonlinear effects, and therefore, a second harmonic may be present in shear wave beams depending on the polarization of the wave. Focused radially polarized shear wave beams have been generated in tissue-mimicking phantoms [Cormack *et al*., IEEE TBME (2024)], and it is postulated that the second harmonic may be used to estimate the quadratic nonlinearity in such a medium as an additional biomarker for diseased tissue. Here, an analytical solution obtained in the paraxial approximation for nonlinear propagation of a shear wave beam in an absorbing medium [Spratt *et al*., AIP Conf. Proc. **1685**, 080007 (2015)] is used to characterize the strength of second-harmonic generation. Feasibility of measuring the second harmonic experimentally in a weakly nonlinear, radially polarized focused shear wave beam propagating in a tissue-like medium is explored. Second-harmonic generation in focused shear wave beams with other polarizations is discussed. [P.G.K. was supported by the ARL:UT Chester M. McKinney Graduate Fellowship in Acoustics.]

10:25

5aPA8. Strong diffraction of nonlinear surface acoustic waves in crystals. Brittany A. McCollom (Appl. Res. Labs. and Walker Dept. of Mech. Eng., Univ. of Texas at Austin, Austin, TX 78713, bmccollom@utexas.edu), John M. Cormack (Div. of Cardiology, Dept. of Medicine, Univ. of Pittsburgh, Pittsburgh, PA), Edoardo Baldini (Dept. of Phys., Univ. of Texas at Austin, Austin, TX), and Mark F. Hamilton (Appl. Res. Labs. and Walker Dept. of Mech. Eng., Univ. of Texas at Austin, Austin, TX)

Nonlinear distortion and shock formation in planar surface acoustic waves in anisotropic crystals have been modeled without [Hamilton *et al*., JASA (1999)] and with [Cormack *et al*., JASA 2022] piezoelectricity. Weak diffraction has been included in the paraxial approximation for nonlinear surface wave beams in isotropic solids [Shull *et al*., JASA (1995)]. Here, a procedure for including strong diffraction, i.e., without the paraxial approximation, in the model for nonlinear surface waves in crystals is presented, which incorporates the angular spectrum approach described by Kharusi and Farnell (JASA 1970). Anisotropy is defined by expressing the phase speed of a plane wave, and therefore, the magnitude of the corresponding wavenumber, as a function of the direction of propagation. Next, an explicit expression relating the two wavenumber components in the planar surface along which the wave propagates must be obtained as a function of direction, requiring iterative solution of a transcendental relation. The beam is then propagated incrementally away from the source, advancing the angular spectrum in k-space and including nonlinear interactions in the spatial domain to characterize the combined effects of diffraction and nonlinearity. Preliminary results are presented for converging nonlinear surface waves in crystals. [Work supported by IR&D at ARL:UT.]

10:40

5aPA9. Studying the nonlinearity effects in ultrasound-assisted water purification and treatment systems. Pooja Dubey (George W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., 2 Rue Marconi, Georgia Tech-Europe, Metz 57070, France, pooja.dubey@gatech.edu) and Nico Declercq (George W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., Metz, France)

In light of industrial sectors' rapid and exponential growth, the deleterious specter of water pollution looms ominously over our environment. One of the significant challenges is obtaining a sustainable and energy-efficient water purification/treatment system. Since the 1990s, several research studies have been proposed to demonstrate the usefulness of ultrasound as a water purification tool. As a result, newly designed devices such as ultrasound-assisted electrochemical treatment and ultrasound-assisted heat exchanging devices are becoming more common. However, such devices' high voltage ultrasonic emission is a significant problem because the nonlinear acoustic effects are not well understood and are, therefore, not adequately integrated into the design of the devices. Furthermore, the presence of biofilms in these devices creates more complexity due to the interaction of high-amplitude ultrasonic waves with the biofilm network. To better overcome the inefficient functioning of such devices and adverse operational issues, the current study aims to investigate and explain the nonlinear ultrasonic effects in ultrasonic-assisted purification devices, with and without biofilm deposits. The obtained insight will help develop an effective design strategy for high efficiency.

Session 5aSA

Structural Acoustics and Vibration: General Topics in Structural Acoustics

Alyssa Bennett, Cochair
Naval Surface Warfare Center Carderock, 9500 MacArthur Blvd, Bethesda, MD 20817

Allison M. King, Cochair
Mechanical Engineering, University of Michigan, 2370 GG Brown, Ann Arbor, MI 48109

Contributed Papers

8:00

5aSA1. Acoustic source localization on finite structures using remote sensors. Allison M. King (Dept. of Mech. Eng., Univ. of Michigan, 1231 Beal Ave. Ann Arbor, MI 48109, kingalli@umich.edu) and David R. Dowling (Naval Architecture and Marine Eng., Univ. of Michigan, Ann Arbor, MI)

Acoustic waves are well-suited for remote sensing applications and structural health monitoring purposes because they convey information about their source and can be measured using non-contacting methods. Source localization is an important structural health monitoring task; however, traditional time-of-flight array signal processing techniques used to localize acoustic sources are ill-suited for many structural engineering applications due to the potential for complicated propagation paths, the dispersive propagation of acoustic waves in structures, and the coupling of the vibrating structure and the surrounding medium. Thus, source localization experiments were conducted using matched field processing (MFP) for a 0.9-m diameter round aluminum plate excited by the impact of a 1.3-cm stainless-steel ball bearing dropped from 7.6-cm. A 14-sensor linear remote acoustic array placed 8.9-cm above the plate measured the sound radiated by the 0.64-cm thick plate. MFP array signal processing localization techniques were used along with a physics-based finite element acoustic model to localize the excitation on the structure. Source localization results in both a quiet environment and environment with additive white Gaussian noise are discussed, and these results are compared to those from an acoustic model where plate edge reflections are neglected. [Work was sponsored by a SMART Scholarship and the NEEC.]

8:15

5aSA2. Passive structural health monitoring of a vibrating shell using data-based matched field processing. Sandrine T. Rakotonarivo (Marine Physical Lab., SCRIPPS Inst. Oceanogr., UCSD, LMA - UMR 7031 AMU - CNRS - Centrale Marseille, 4 impasse Nikola Tesla, Marseille 13453, France, sandrine.rakotonarivo@univ-amu.fr), Theo Langlet (CEA Cadarache, MISTRAL Lab., LMA, AMU, Marseille, France), Jit Sarkar (Marine Physical Lab., SCRIPPS Inst. Oceanogr., UCSD, San Diego, CA), and William Kuperman (Marine Physical Lab., SCRIPPS Inst. Oceanogr., UCSD, La Jolla, CA)

This study presents experimental results on passive structural health monitoring of a vibrating elastic structure for defect localization. The approach is based on matched field processing (MFP), which requires a model of the pristine structure. An MFP methodology for localizing a defect in a shell equipped with vibration sensors was developed and numerically demonstrated in [JASA EL **2**, 025601 (2022)]. This paper further extends this study to experimental implementation of this MFP methodology without requiring full knowledge of the structure parameters and the boundary conditions of the pristine structure. The method is tested on experimental data for localizing a defect on a rectangular steel plate with non-perfectly, fixed boundary conditions.

8:30

5aSA3. Sensing touch location on an elastic surface by monitoring structural vibrations. Benjamin R. Thompson (Elec. & Comput. Eng., Univ. of Rochester, 120 Trustee Rd., Rochester, NY 14620, bthomp23@ur.rochester.edu), Tre DiPassio, Jenna Rutowski, Mark Bocko, and Michael C. Heilemann (Elec. & Comput. Eng., Univ. of Rochester, Rochester, NY)

In recent years, touchscreens have become ubiquitous, and projected capacitance (p-cap) has arisen as the dominant touch-sensing technology. Though powerful, p-cap has some characteristics that make it less suited for certain applications. Active acoustic sensing (AAS) systems work by sensing the vibration response of an elastic surface to a system-generated excitation and associating specific response characteristics with touch locations on the surface. These systems have the potential to offer advantages over p-cap in terms of scalability, cost, and performance in harsh environments. AAS systems may also offer the ability to sense touch pressure as an additional input parameter. We developed an empirical model of surface vibrational response as a function of the location of an applied force and then used this model to inform the signal processing and machine learning approaches employed in a prototype AAS system. In this presentation, we provide results from our empirical model as well as details of the prototype system, including its construction and performance.

8:45

5aSA4. Direction of arrival estimation in reverberant environments using a single vibration sensor on an elastic panel. Jenna Rutowski (Elec. & Comput. Eng., Univ. of Rochester, 500 Joseph C. Wilson Blvd, Rochester, NY 14627, jrutowsk@ur.rochester.edu), Tre DiPassio, Benjamin R. Thompson, Mark Bocko, and Michael C. Heilemann (Elec. & Comput. Eng., Univ. of Rochester, Rochester, NY)

The vibrational response of an elastic panel to an incoming acoustic pressure wave is dependent on the coupling between the incident angle and the panel's bending modes. By examining the relative modal excitations recorded by a single structural vibration sensor affixed to the panel, the direction of arrival (DOA) of the incident wave may be inferred. In reverberant environments, determination of the DOA of a sound source is complicated by acoustic reflections. Panel microphones may be particularly susceptible to this effect due to their large surface areas and finite modal decay times. A study on the impact of reverberation on DOA estimation using panel microphones was conducted. The panel's response to wake word utterances was recorded in eight spaces, with reverberation times (RT60s) ranging from 0.27 to 3.00 s. These responses were used to train neural networks for DOA estimation. Results indicate an inverse relationship between RT60 and DOA estimation reliability. Within $\pm 5°$, DOA estimation reliability was measured at 95% in the least reverberant space and decreased to 78% in the most reverberant space. Results also suggest that DOA estimation using panel microphones can adapt to diverse acoustic environments by training the system with data from multiple spaces with different RT60s.

9:00

5aSA5. Experimental modal testing of growing thin ice. Carl R. Hart (U.S. Army Engineer Res. and Development Ctr., Cold Regions Res. and Eng. Lab., 72 Lyme Rd., Hanover, NH 03755, carl.r.hart@erdc.dren.mil), Cody M. Best, Emily Asenath-Smith, Kiera L. Thompson Towell (U.S. Army Engineer Res. and Development Ctr., Cold Regions Res. and Eng. Lab., Hanover, NH), Michelle E. Swearingen (Construction Eng. Res. Lab., US Army ERDC, Champaign, IL), and Michael B. Muhlestein (U.S. Army Engineer Res. and Development Ctr., Cold Regions Res. and Eng. Lab., Hanover, NH)

Current practices for assessing the thickness of growing ice include drilling and coring. This is inherently risky and motivates the use of a stand-off method. Laser Doppler vibrometry is a potential technique, however, the interpretation of vibrometer signals must be informed by a physical understanding of ice subjected to mechanical vibrations. The vibrational response of growing, thin ice is largely unknown. For thick ice, adequate predictions for the vibrational response assumes ice behaves as an elastic plate. It is hypothesized that thin ice responds in a fashion somewhere between an elastic membrane and an elastic plate. A modal test of thin ice growing within an insulated plastic basin was conducted over the course of two tests. The resulting dataset spans a range of ice thicknesses from 8 to 45 mm. Examination of Bode and Nyquist plots showed that the two lowest observable ice resonances, as measured at the center of the ice sheet, exhibit a dependence on ice thickness distinctly different from a fixed elastic plate loaded by a half-space of water. Preliminary finite-element analysis calculations indicates that the basin wall compliance is a significant factor in modifying the ice resonance versus thickness relationship.

9:15

5aSA6. Coupling the force analysis technique and full-field vibration measurements for the identification of a time-space-varying sound pressure loading on a membrane. Anaïs Mougey (Université de Sherbrooke, 2500 boulevard de l'université, Sherbrooke, QC J1K 2R1, Canada, anais.mougey@usherbrooke.ca), Manuel Melon, Félix Foucart (Le Mans Université, Le Mans, France), and Olivier Robin (Université de Sherbrooke, Sherbrooke, QC, Canada)

This work is grounded on the force analysis technique, an identification method that directly uses a structure's equation of motion to formulate an inverse problem, explicitly identifying the force causing the structure's motion. Prior research employing this technique has been predominantly conducted in the frequency domain and was limited to stationary, mechanical excitations using physical sensor arrays such as accelerometers. The objective of this research was mostly quantitative (amplitude, location), while the proposed approach is rather qualitative identification. Indeed and by combining the force analysis technique with full-field and non-contact vibration measurements conducted on a system, here a membrane, this communication describes a proof-of-concept for the identification of a time-space-varying sound pressure loading. A compact and tonal sound source is used to draw freehand shapes against the membrane surface, and the objective is to follow/reconstruct the trajectory followed by this source. Results are provided for different drawn shapes or letters, and the effect of mechanical or calculation parameters on the reconstructed information is studied. Finally, potential research directions are discussed and fed by preliminary measurements on a percussion instrument.

9:30

5aSA7. Forces on a singing wineglass rim. Megan F. Orzolek (Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, mfo4@psu.edu), Alexander M. Mertz, and Michael L. Jonson (Penn State, University Park, PA)

The characterization of friction is a multiscale problem involving a balance between asperity contacts and fluid film forces, most commonly modeled with a velocity-dependent coefficient of friction. Periodic stick–slip phenomena are often associated with the friction force that causes mechanical vibration. This type of vibration can cause noise to radiate via structural modes of a system. Notable examples include journal bearings, brake squeal, turkey calls, and wineglasses. Singing wineglasses have been used for centuries as musical instruments, so musical acousticians have studied the structural vibrations and acoustic output. However, the tri-axial force applied to the rim has not been measured. A test rig was designed to measure the vertical, tangential, and radial dynamic forces applied to the spinning glass rim and simultaneously capture the radiated noise. The sound radiated from the wineglass was observed to depend on the relative velocity at the rim and applied vertical load, so a stability region for a clear sound was determined. Additionally, the strongest radiating modes and harmonics in the pressure were found in the concurrently measured nonlinear force. This work may be extended to confirm the force measurement by existing friction models, a water medium, and more complicated systems.

9:45–10:00 Break

10:00

5aSA8. Abstract withdrawn.

10:15

5aSA9. Leak detection in an operational underground water distribution network using active acoustics. Pranav Agrawal (Civil and Environ. Eng., Univ. of California, Los Angeles, 580 Portola Plaza, 5731 Boelter Hall, Los Angeles, CA 90095-1593, pranav0505@g.ucla.edu), Stan Fong, Dirk Friesen (Digital Water Solutions, Waterloo, ON, Canada), and Sriram Narasimhan (Civil and Environ. Eng., Univ. of California, Los Angeles, Los Angeles, CA)

Leaks present in the water distribution networks (WDNs) lead to an enormous loss of a valuable resource, not only affecting the economic efficiency of water utilities but also posing significant potential safety hazards, and an increased burden on infrastructure maintenance. In the past, several studies have focused on passive acoustic-based leak detection by primarily sensing and analyzing the acoustic signals emitted from the leak source. In this study, we installed a low-frequency transducer in the water column of an operational WDN near Los Angeles, California, and excited the system using steady-state sinusoidal and sine burst signals. The acoustic pressure signals inside the pipe network were measured at multiple locations using state-of-the-art hydrophone-enabled devices retrofitted to fire hydrants. The experiments were conducted in the presence of a simulated leak in the network. The leak acts as an impedance discontinuity for the acoustic wave propagation, and therefore, the excited signals undergo partial reflection at the leak location. Based on signal processing techniques, we attempt to detect and localize the leaks in the WDN. The goal of using the active acoustics is to detect small leaks and increase the range of sensing.

10:30

5aSA10. Tri-axial ground-borne vibration measurements during rail pass-bys. Harry Ao Cai (HGC Eng., 2000 Argentia Rd., Ste. 203, Plaza 1, Mississauga, ON L5N 1P7, Benin, hcai@hgcengineering.com), Nathan Gara (HGC Eng., Mississauga, ON, Canada), and Brian Howe (HGC Eng., Mississauga, Cambodia)

Ground-borne vibrations from rail pass-bys, transmitted from train wheels rolling on the rails, have the potential to cause various adverse effects at nearby receptors, such as annoyance and re-radiated noise. To assess the impact of rail pass-bys, surface-level vibration can be characterized in three components: one vertical and two horizontal directions. Prior experience, best-practice guidelines from the Federal Transit Administration Transit Noise and Vibration Impact Assessment Manual, and theory of propagation of Rayleigh surface waves indicate that the vertical component dominates the horizontal components, such that vibrations from rail pass-bys can be adequately characterized by the vertical component only. This purpose of this study is to assess the sufficiency of characterizing the impact of ground-borne vibrations caused by rail pass-bys based solely in the vertical direction. Tri-axial vibration measurements were conducted for freight and passenger trains in Ontario, Canada, using a multi-channel signal analyzer for simultaneous measurement of multiple-axis vibration levels. The measured levels were then examined across the three components.

10:45

5aSA11. Analysis, design, and installation of vibration isolation for lightweight helipads. Alfredo Rodrigues (CDM Stravitec, 100 Sunrise Ave., 202, Toronto, ON M4A1B3, Canada, a.rodrigues@cdm-stravitec.com), Ashwin Dias (CDM Stravitec, Overijse, Belgium), and Freddy Saddik (CDM Stravitec, Dallas, TX)

This case study discusses the design and manufacturing steps for the installation of a vibration isolation solution for two lightweight helidecks installed on the rooftop of an existing medical center building. The design stage involved the analysis of the relevant information, involving several specialties, such as Acoustics (for sources of vibration and acoustic performance requirements), Heliport Design[AR1] (for the type of aircraft, MTOW, number of supports, static and dynamic loads), and Structural (for the types of connections to the existing building structure). A collaborative effort between CDM Stravitec and the helideck supplier led the design through a progressive and iterative process where a final design that responded to the strong acoustical requirements and the complex nature of a fully functional and integrated lightweight helideck. Along the design, compromises where made ensure that requirements are met for both acoustic and structural performance. The solution delivered was a prefabricated box, which could easily be installed on site, comprised the necessary components, such as springs and uplift restraints to deal with the challenging aspects of the isolation of lightweight helidecks. The paper also discusses the challenges and best practices encountered during the production and installation of the system.

11:00

5aSA12. Optimization of an aircraft fuselage assembly to minimize radiated sound power. Jordan Howes (Mech. and Mater. Eng., Queen's Univ., 130 Stuart St., Kingston, ON K7L2V9, Canada, jordan.howes@queensu.ca), Adam McKenzie, Wesley Dossett, Luke Crispo, and Il Yong Kim (Mech. and Mater. Eng., Queen's Univ., Kingston, ON, Canada)

During the structural design of an aircraft fuselage, the acoustic performance is generally not considered. However, vibrations transmitted through the fuselage generate noise leading to passenger discomfort. Currently, the main source of noise reduction comes from the addition of damping material to the fuselage. This work investigates the use of various design optimization techniques to reduce radiated sound power by changing the fuselage structure's geometry which consists of a skin panel, frames, and stringers. Design optimization tools, such as topology, size, and shape optimization were used to determine an optimal design for each design variable that minimized radiated sound power in the fuselage assembly. The design variables that were studied included skin panel thickness and design, frame and stringer spacing, and frame and stringer cross section. Equivalent radiated power was used as an objective function for numerical optimizations within Altair OptiStruct as an indirect method for minimizing radiated sound power numerically. To understand the source of the sound power improvements, a physics analysis was conducted for each design variable that compared sound power, equivalent radiated power, and radiation efficiency.

11:15

5aSA13. Comparison and analysis of onboard vibration and underwater noise radiated by ships of different classes. Kamal Kesour (Innovation Maritime, 53 Rue St-Germain Ouest, Rimouski, QC G5L 4B4, Canada, kkesour@imar.ca), Paul Camerin, Jean-Christophe G. Marquis (Innovation Maritime, Rimouski, QC, Canada), and Cédric Gervaise (SenseaFR, Grenoble, France)

The shipping industry plays a major role in the increase in anthropogenic noise in the marine environment. To understand and measure the underwater noise emitted by ships, the MARS (Marine Acoustic Research Station) project has deployed an acoustic station in the St. Lawrence River to measure ship signatures between 2021 and 2023. The acoustic station complies with the ANSI/ASA S12.64 2009 standard. In addition, the project team carried out simultaneous vibration diagnostics to identify sources of underwater noise on board several vessels. This paper presents the main results of the onboard vibration measurements, including a comparison of vibration levels measured on bulk carriers, tankers, and general cargo ships. The contribution of diesel generators, propulsion system, and propeller cavitation to the acoustic signature is also presented and analyzed for each vessel class. As expected, propeller cavitation was found to be the main source of shipborne noise of the investigated vessels, especially above 50 Hz. Finally, the acoustic signatures of ships equipped with electric and diesel propulsion systems are also compared, while identifying the frequencies related to each source.

11:30

5aSA14. Mode-matching analysis of acoustic scattering in bifurcated circular cylindrical waveguides with liner conditions and step-discontinuity. Qamar Abbas (Mathematics, Indus College Kahuta, Rawalpindi, Pakistan, Indus College Kahuta, Rawalpindi Rd., Kahuta, Punjab 47330, Pakistan, qamar299ICK@gmail.com) and Rab Nawaz (Mathematics, COMSATS Univ. Islamabad, Islamabad, Pakistan)

The study delves into the acoustic scattering from a bifurcated circular cylindrical waveguide, considering liner conditions and step-discontinuity. Specific impedance of acoustic liners is developed for analyzing scattering characteristics. Two problems, with and without liners, are formulated and solved using Mode-matching technique, based on eigenfunction expansions. The solution's accuracy, dependent on eigenfunction properties influenced by the medium and wall conditions, is validated through power conservation and matching conditions. Results reveal the considerable impact of reacting liners and step-discontinuity on scattering concerning frequency and duct dimensions. The study implicates to various fields where the control and understanding of acoustic scattering phenomena are crucial for optimizing the performance of devices and systems.

Session 5aSC

Speech Communication: Speech Production and Speech Tech Poster Session

Zhaoyan Zhang, Chair
UCLA School of Medicine, Los Angeles, CA 90095

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

Contributed Papers

5aSC1. Comparison of automated and manual measures of speaking rate in individuals with dysarthria. Lian J. Arzbecker (Communicative Disord. and Sci., Univ. at Buffalo, 103D Cary Hall, Buffalo, NY 14207, arzbecker.1@osu.edu) and Kris Tjaden (Communicative Disord. and Sci., Univ. at Buffalo, Buffalo, NY)

Speech patterns in speakers with dysarthria exhibit a broad range of prosodic and articulatory variations, potentially posing a significant challenge for automated speech recognition systems. This study seeks to address a gap in the literature by exploring a fully automated approach to determine speaking rates in English-speaking individuals with dysarthria. This research aims to elucidate whether a syllable-based automated method yields results akin to manual processes when calculating speaking rates across diverse groups [i.e., individuals with dysarthria secondary to multiple sclerosis (MS) or Parkinson's disease (PD), and healthy controls]. Moreover, this study examines the performance of an automated method across three distinct speech tasks: isolated sentences, paragraph reading, and spontaneous speech. The methodology involves adjusting parameters within a version of a published script specifically designed for calculating speaking rate through syllable detection (Praat Script Syllable Nuclei; de Jong *et al.*, 2021). Using a sample of 60 speakers (20 MS, 20 PD, and 20 healthy controls), this study aims to evaluate the accuracy and validity of an automated script in calculating speaking rate. Ultimately, these findings are intended to guide speaking rate measurement of dysarthria in research and clinical practice.

8:00

5aSC2. Subglottal resonances in healthy older adults. Steven M. Lulich (Speech Lang. & Hearing Sci., Indiana Univ., 200 South Jordan Ave., Bloomington, IN 47405, slulich@indiana.edu)

Resonances of the subglottal airways impact the vocal tract transfer function and voice production during speech through linear and non-linear acoustic coupling. It has been posited that subglottal resonances form acoustic boundaries between contrasting sets of vowels and consonants based on their coupling effects on formant frequencies. Thus far, studies of subglottal resonances have reported data almost exclusively from children and young adults; data from older adults are rare. This study investigates subglottal resonances in eight healthy older adults between 50 and 80 years of age (5 males, 3 females). Initial analyses indicate that the first and especially the second subglottal resonances are at lower frequencies than expected when compared against height-matched young adults but that the subglottal resonance frequencies are not dependent on posture, vowel acoustics, or standard measures of pulmonary function.

5aSC3. Modeling syllable rhythms for interactive alignment studies. Brandon Copping (Commun. Sci. and Disord., Univ. of Memphis, 4055 North Park Loop, Memphis, TN 38122, bcopping@memphis.edu) and Eugene Buder (Commun. Sci. and Disord., Univ. of Memphis, Memphis, TN)

Many acoustic cues have been proposed as indicators and coordinators of upcoming turn-exchanges in conversational interactions, including syllable rhythm. Rhythmic coordination can help to explain close coordination of speech during these exchanges with intervals on the order of 250 ms or less, or with fluent and non-disruptive overlaps. While general questions surrounding speech rhythm remain unanswered, investigations of syllable alignment across partners in conversation can bypass such questions by assessing directly for patterns that align across vocal exchanges. We develop models of syllable rhythm for this purpose using hand-coded syllable durations from spontaneous conversations and from turn-taking responses elicited by recorded stimuli. Our approach builds on evidence that turn-takers begin planning their utterance approximately 0.5 s prior to turn endings. Syllable duration strings from appropriately identified frames are replaced by Gaussian curves to create oscillatory waveforms, and cross-correlation functions are applied to identify rhythmic continuities across exchanges. Patterns identified using this technique are explanatory of close coordination in the spontaneous exchanges, and supra-syllabic units such as the foot and phrase are identified as well. This syllable framework can be developed further by incorporating additional acoustic measures such as f_{o} and amplitude to build a multidimensional construct for interactional rhythm alignment.

5aSC4. Mapping acoustic characteristics of emotional prosody in Mandarin disyllabic words: A machine-learning approach. Xuyi Wang (Speech-Language-Hearing Ctr., School of Foreign Lang., Shanghai Jiao Tong Univ., Dongchuan Rd. 800, Shanghai 200240, China, wxy_evie@sjtu.edu.cn), Hongwei Ding (Speech-Language-Hearing Ctr., School of Foreign Lang., Shanghai Jiao Tong Univ., Shanghai, China), and Yang Zhang (Speech-Language-Hearing Sci. and Masonic Inst. for the Developing Brain, Univ. of Minnesota, Minneapolis, MN)

This study conducted an acoustic-prosodic mapping analysis of emotional prosody in Mandarin Chinese. It utilized a validated audiometry corpus with 450 disyllabic words. The spoken words covered five basic emotions produced by a female speaker: angry, sad, happy, fearful, and neutral. A machine-learning approach was adopted to map key acoustic-prosodic features for Mandarin emotional vocalization. The results revealed distinctive acoustic profiles for each emotion, highlighting variations in fundamental frequency, intensity, speaking rate, and voice quality. Emotional utterances consistently exhibited higher mean F0 values than neutral expressions. Fear displayed the highest crest in F0. Angry and happy utterances showed greater vocal intensity and a faster speaking rate compared to fearful and sad expressions. While anger was associated with a creaky voice quality, sadness corresponded with a breathier voice quality. The current findings are limited with the use of the single-speaker corpus. Ongoing efforts aim to expand the corpus with more speakers to test the

generalizability and scalability of the analysis approach for subsequent investigations.

5aSC5. How nasal airflow can affect Nasalance magnitude. Liran Oren (Otolaryngol., Univ. of Cincinnati, University of Cincinnati, PO Box 670528, Cincinnati, OH 45267, orenl@ucmail.uc.edu)

Nasometry is a method for evaluating the function of the velopharyngeal valve. This technique provides a measure of Nasalance, calculated from acoustic energy captured with a device (nasometer) that can separate the oral and nasal acoustic signals using a sound separation plate. We previously observed that the presence of nasal emission can dramatically increase nasalance magnitude. The unintended elevation of nasalance occurs because airflow from the nares impinges on the nasometer's nasal microphone. We aim to quantify this effect using a customized nasometer in patients diagnosed with nasal emission. The customized nasometer has six microphones (three pairs) placed in a radial configuration on the top and bottom of the separation plate. Ideally, all microphones on the same side (i.e., top or bottom) should measure the same Nasalance value. A difference in the measurement of the nasal microphones would be attributed to flow interference. Preliminary results show that the airflow can artificially elevate the Nasalance magnitude by as much as 15 points depending on the produced syllable. These findings can help explain the common discrepancy observed in speech clinics between the Nasalance magnitude and the perceived severity of nasal emission.

5aSC6. Optimization of classifying accurate and misarticulated speech sounds for use in a gamified real-time ultrasound biofeedback system. Sarah C. Biehl (Biomedical Eng., Univ. of Cincinnati, 3159 Eden Ave., Cincinnati, OH 45219, biehlsc@mail.uc.edu), Sarah Dugan (Rehabilitation, Exercise, and Nutrition Sci., Univ. of Cincinnati, Dayton, OH), Sarah R. Li, Alex Knapp (Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH), Renee Seward (School of Design, Univ. of Cincinnati, Cincinnati, OH), Michael A. Riley (Rehabilitation, Exercise, and Nutrition Sci., Univ. of Cincinnati, Cincinnati, OH), Suzanne Boyce (Commun. Sci. and Disord., Univ. of Cincinnati, Cincinnati, OH), and T. Douglas Mast (Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH)

Integrating ultrasound biofeedback therapy (UBT) into a real-time, gamified interface to provide articulatory feedback for speech remediation promotes an external focus of attention, thereby reducing the complex cognitive demands required for standard UBT. Previous studies have shown that accuracy of American English rhotic /r/ can be predicted by a single parameter, the difference between tongue dorsum and blade displacements measured by ultrasound imaging during speech production. This parameter has classified speech productions of rhotic syllables as correct versus misarticulated with a classification accuracy up to 85%. However, implementation of this classification approach into real-time gamified UBT, including both measurement timing and establishment of difficulty levels for progressive therapy, would benefit from optimization using a larger dataset. 2,450 productions of 10 distinct rhotic syllables (including prevocalic and postvocalic contexts) from 50 children, with and without misarticulations, were analyzed. For each production, ultrasound image sequences were processed by TonguePART software to acquire tongue displacement trajectories, and accuracy was judged by trained listeners using a visual analog scale. Analyses were conducted to optimize selection of the image frame for classification, determine parameter thresholds appropriate for real-time prediction of /r/ production accuracy, and integrate these thresholds into a difficulty level design for gamified UBT.

5aSC7. Control parameters for coordinative structures in speech production. Matthew Masapollo (McGill Univ., 630 William St., Montreal, QC H3C 4C9, Canada, matthew.masapollo@mcgill.ca) and Susan Nittrouer (Speech, Lang., and Hearing Sci., Univ. of Florida, Gainesville, FL)

In skilled speech production, sets of articulators work cooperatively to achieve task-specific movement goals, despite rampant contextual variation. Efforts to understand these functional units, termed coordinative structures, have focused on identifying the essential control parameters responsible for allowing articulators to achieve these goals, with some research focusing on temporal parameters (relative timing of movements) and other research focusing on spatiotemporal parameters (phase angle of movement onset for one articulator, relative to another). Here, we compared findings across three recent studies where both types of control parameters were investigated, using electromagnetic articulography recordings. In each study, talkers produced VCV utterances, with alternative V (/ɑ/−/ɛ/) and C (/t/−/d/ or /p/−/b/), across variation in rate (fast–slow) and stress (first syllable stressed–unstressed). Two measures were obtained: (i) the timing of tongue-tip or lower-lip raising onset for intervocalic C, relative to jaw opening–closing cycles, and (ii) the angle of tongue-tip or lower-lip raising onset, relative to the jaw phase plane. All three studies showed that the correlations of tongue-tip/lower-lip movement onset latencies and jaw opening-closing cycle durations were stronger and more reliable than the correlations of tongue-tip/lower-lip phase angles and jaw opening-closing cycle durations, demonstrating that timing is the critical control parameter.

5aSC8. Generalization of inter-articulator timing control: Evidence from tongue-jaw and lip-jaw kinematics using electromagnetic articulography. Matthew Masapollo (McGill Univ., 630 William St., Montreal, QC H3C 4C9, Canada, matthew.masapollo@mcgill.ca), Ana Rodriguez, Kara Kent (Speech, Lang., and Hearing Sci., Univ. of Florida, Gainesville, FL), Rosalie Gendron (McGill Univ., Montreal, QC, Canada), Hannah Thomas (Speech, Lang., and Hearing Sci., Univ. of Florida, Gainesville, FL), Nathan Maxfield (Univ. of South Florida, Tampa, FL), and Susan Nittrouer (Speech, Lang., and Hearing Sci., Univ. of Florida, Gainesville, FL)

Recent research in speech production indicates that talkers reliably control the relative timing of articulator movement onsets across variation in production rate, syllable stress, and segmental makeup and that this precision of inter-articulator timing control instantiates phonetic structure. To date, these timing relations have been highly reliable for tongue-jaw kinematics. In the present study, we address the generality of these timing relations to lip-jaw kinematics. Eleven talkers recorded 240 /tV#Cat/ and 240 /bV#Cab/ utterances using electromagnetic articulography, with alternative V (/ɑ/-/ɛ/) and C (/t/−/d/ or /p/−/b/), across changes in production rate (fast-normal) and stress (first syllable stressed-unstressed). To quantify inter-articulator temporal coordination, the timing of either tongue-tip or lower-lip raising onset for the intervocalic C, relative to the jaw opening-closing cycle for V, was obtained. Results indicate that the same kinematic pattern occurred among both sets of articulators: any manipulation that shortened the jaw opening-closing cycle reduced the latency of either tongue-tip or lower-lip movement onset, relative to the onset of jaw opening. Furthermore, the movement onset latencies of the tongue-tip and lower-lip were both highly differentiated by utterance type, bolstering the view that inter-articulator timing relations instantiate phonetic structure in the resulting acoustic signal.

5aSC9. Real-time speech adaptations in conversations between human interlocutor and AI confederate. Fenqi Wang (Dept. of Linguist, Simon Fraser Univ., 8888 University Dr., Burnaby, BC V5A 1S6, Canada, fenqiw@sfu.ca), Jetic Gu (School of Computing Sci., Simon Fraser Univ., Burnaby, BC, Canada), Meagan Durana, Chihiro Mabohang (Dept. of Linguist, Simon Fraser Univ., Burnaby, BC, Canada), Dawn Behne (Dept. of Psych., Norwegian Univ. of Sci. and Technol., Trondheim, Norway), Allard Jongman (Dept. of Linguist, The Univ. of Kansas, Lawrence, KS), Joan Sereno (Dept. of Linguist, The Univ. of Kansas, Kansas City, KS), and Yue Wang (Dept. of Linguist, Simon Fraser Univ., Burnaby, BC, Canada)

Compared to human-directed adaptations, less is known about how humans adjust their speech for intelligibility benefits while interacting with an AI-powered voice interface. In this study, we investigate human speech adaptations in human-to-human versus human-to-AI unscripted conversations. Specifically, we examine the production of words containing intervocalic /t-d/ in a conversation between a speaker who distinguishes these two stops (e.g., metal–medal) and a speaker ("flapper") who merges the two stops into a flap /ɾ/. We predict that misperceptions of intervocalic /t-d/ may cause confusions, thus motivating adaptations. We record native Canadian-English speakers (flappers) while playing a video game on Zoom in two conversation settings: with (1) a human non-flapper and (2) an AI non-

flapper (computer-generated speech). Acoustic analyses of the productions by human flapper speakers include features specific to stop-flap distinctions as well as global features (e.g., overall duration). In both human- and AI-directed speech, we expect human interlocutors to change flapped productions to stops to enhance intelligibility, particularly late in the conversation. Moreover, we expect differences between human- and AI-directed adaptations, with the former dominantly employing sound-specific features and the latter relying more on global hyperarticulation. Understanding these interlocutor-oriented adaptations may inform the technology behind human-computer interfaces.

5aSC10. A new experimental design to study speech adaptations in spontaneous human-computer conversations. Jetic Gu (School of Computing Sci., Simon Fraser Univ., 8888 University Dr., Burnaby, BC V5A1S6, Canada, jeticg@sfu.ca), Fenqi Wang, Ivan Fong, Samuel To (Dept. of Linguist, Simon Fraser Univ., Burnaby, BC, Canada), Dawn Behne (Dept. of Psych., Norwegian Univ. of Sci. and Technol., Trondheim, Norway), Allard Jongman (Dept. of Linguist., The Univ. of Kensas, Lawrence, KS), Joan Sereno (Dept. of Linguist., The Univ. of Kensas, Kansas City, KS), and Yue Wang (Dept. of Linguist., Simon Fraser Univ., Burnaby, BC, Canada)

Interest is growing for how human interlocutors make phonetic adaptations during spontaneous conversations. Given the increasing popularity of AI chatbots, research also needs to account for adaptations in human–computer interactions, an area under-investigated presumably due to methodological challenges in generating controlled conversational responses. Most studies involve scripted computer output, which may obstruct the dynamicity and the oral–aural medium of a natural conversation. To circumvent these constraints, we present a new experimental design that generates unscripted audio computer responses in human–computer conversations during a collaborative game played on Zoom. This design is unique in several aspects. First, the game (Escape Room) requires discussions on placing pictures (depicting target words/sounds) in specific locations, where misperceptions of target words between interlocutors may cause confusions, thus motivating natural adaptations. Second, to enable real-time computer responses, we adopt the wizard-of-oz paradigm typically used in the field of human–computer interaction, where a human confederate inputs text responses behind-the-scenes. Third, a programmable text-to-speech synthesizer converts the text input to audio output. The design demonstrated in this presentation opens the door to new analyses, tracking the dynamicity of speech adjustments over time. Moreover, it is generalizable to studying speech adaptations across interlocutor backgrounds.

5aSC11. Abstract withdrawn.

5aSC12. PrEgg: A free and open source Praat script for electroglottography measurements. May Pik Yu Chan (Dept. of Linguist, Univ. of Pennsylvania, 3401-C Walnut St., Ste. 300, C Wing University of Pennsylvania, Philadelphia, PA 19104-6228, pikyu@sas.upenn.edu) and Jianjing Kuang (Linguist, Univ. of Pennsylvania, Philadelphia, PA)

Electroglottography is a popular method of phonation analysis due to its non-invasive nature. In order to ease the extraction of systematic EGG measures, we wrote PrEgg, an open source Praat script designed to extract Contact Quotient (CQ), Skew Quotient (SQ) and Peak Increase in Contact (PIC) values. The script offers two measures for CQ and SQ, calculated using the EGG threshold method developed by Rothenberg (1988) and the DEGG method from Henrich *et al*. (2004), and PIC is further calculated from the DEGG signal. To test the validity of the script, we ran the Praat script on EGG data on Yi provided in the "Production and Perception of Linguistic Voice Quality" project at UCLA. We compared our results on CQ measurements using both the threshold method and the DEGG method with the provided CQ and CQ_{PM} measurements extracted from EggWorks, a UCLA software. Overall results are comparable between PrEgg and EggWorks. PrEgg is accompanied by a manual and available at https://github.com/maypychan/praat-egg.

5aSC13. Personality perception in synthetic versus natural speech: The effects of voice quality and prosody. Minjeong Kim (Graduate School of Culture Technol., Korea Adv. Inst. of Sci. and Technol., 291, Daehak-ro, Yuseong-gu, Daejeon 34141, Republic of Korea, minjeong.kim@kaist.ac.kr), Jaehan Park (KT Corp., Seoul, Republic of Korea), Minhong Jeong (Graduate School of Culture Technol., Korea Adv. Inst. of Sci. and Technol., Daejeon, Republic of Korea), and Jieun Song (School of Digital Humanities and Computational Social Sci., Korea Adv. Inst. of Sci. and Technol., Dajeon, Republic of Korea)

Synthetic speech technology, now approaching human-like naturalness due to advancements in deep learning, has turned to focusing on personality or persona design as a common practice in the industry. The present study aimed to identify speech characteristics affecting personality impressions in synthetic and natural speech. Thirty native Korean speakers participated in a personality rating experiment in which they evaluated natural Korean sentences and their synthetic counterparts in terms of the Big-Five personality model. Acoustic analyses were performed to examine voice quality and prosody, including Intonational Phrase (IP) boundary tones. The results revealed that scores of agreeableness, conscientiousness, and emotional stability increased overall when the voices contained greater aperiodicity in the harmonics (i.e., were likely breathier) and were weaker in energy. The results also demonstrated that different prosodic features affected personality perception in synthetic and natural speech; synthetic speech with a wider F0 range received higher scores on extroversion, openness, and emotional stability. In contrast, the effect of IP boundary tones was most frequently found for female natural speech, which contained a wider range of tones, including multitonals (e.g., LHL%). Our findings suggest that intonation is one of the key factors which can be adjusted to generate synthetic speech with various personalities.

5aSC14. Modeling the neural encoding of vowel formants in the midbrain. Daniel D. Pyskaty (Depat. Linguist, Neurosci., Univ. of Rochester, Rochester, NY 14627, dpyskaty@u.rochester.edu), Joyce M. McDonough (Linguist, Univ. of Rochester, Rochester, NY), and Laurel H. Carney (Biomedical Eng., Univ. of Rochester, Rochester, NY)

The first (F1) and second (F2) formants of a vowel determine its identity, making it essential to understand how these formant regions are encoded in the brain. We investigated the neural encoding of vowel formants using computational model responses of the auditory nerve and midbrain to natural vowel tokens [Hillenbrand et al., JASA 97, 3099 (1995)]. The first formant peak can be difficult to identify based on model population responses. Responses in the F1 region are more affected by the surrounding harmonic structure, whereas F2 often has a more straightforward representation in the model response. This result is surprising given the sensitivity of listeners for F1 discrimination compared to F2, reflecting Vowel Dispersion Theory [Liljencrants & Lindblom, Language 48, 839 (1972)], which favors the F1 dimension over F2. While peaks are difficult to identify, complex patterns of neural discharge rates observed across low-frequency midbrain responses may work to encode F1. These results suggest revisiting assumptions of the neural representations of formants and highlight the need for a system that evaluates overall response trends to formant regions instead of searching for formant peaks. Neural models can play an important role in understanding the structure of vowel systems. [Work supported by NIDCD-R01-010813.]

5aSC15. Collecting non-native English speech through a web-and mobile-based application. Bozena Kostek (Audio Acoust. Lab., Gdansk Univ. of Technol., Narutowicza 11/12, Gdansk 80-233, Poland, bokostek@audioakustyka.org)

The motivation behind this study is to collect non-native English speech recordings seamlessly. Even though numerous corpora containing non-native English speakers exist, there are not so many resources of Polish speakers uttering sentences in English. That is why a web-based application is built to record L2 English speakers. The same application is also developed in the mobile form as it provides easy access to the software created and, at the same time, built-in microphones ensure sufficient quality of recordings. A set of sentences is prepared, including four main types, i.e., declarative, imperative, interrogative, and exclamatory. Recordings are saved in WAV audio format. Local data are first saved in the memory of the device on which the application is installed, and then, the user may listen to the recordings and then send them to the server using FTP (File Transfer Protocol). The user interface allows for English language level selection, by choosing one of the four possible options, i.e., basic, intermediate, advanced, and fluent. A group of volunteers provided speech samples, resulting in the collection of utterances from more than 600 speakers, available in the created corpus. Future development envisions adding additional modules that will allow for speech signal analyses.

5aSC16. Automatic detection of nasal closure and nasal release landmark acoustic cues. Janette Park (RLE, MIT, 50 Vassar St., Cambridge, MA 02139, janp@mit.edu), Jeung-Yoon Choi (MIT, Cambridge, MA), and Stefanie Shattuck-Hufnagel (RLE, MIT, Cambridge, MA)

This study describes the detection of nasal closure and nasal release landmarks, as part of a larger system for speech recognition based on acoustic cues. Landmarks are produced as a result of closures and releases in the oral region and are indicated by abrupt changes in the speech signal. Nasal closure and release landmarks have proven particularly challenging to detect and are the focus of this report. The process for implementing the nasal detection module includes extracting and processing a set of speech-related measurements, such as formant frequencies, spectral band energies, and their derivatives, from a large database of labeled speech files, and determining which of these measurements are potentially effective, using ANOVA analysis. Next, Gaussian mixture models are trained and tested on these measurements to classify nasal closures, nasal releases, and all other landmark cues. The resulting nasal closure and release landmark detection module will be used with other landmark modules for vowels, glides, fricative closures/releases, and stop closures/releases, as well as other acoustic cues to place and voicing, in the overall speech recognition system. The current performance of the module will be assessed and discussed.

5aSC17. Investigation of machine-learning-based stimuli for the remediation of children's speech errors. Kathryn Cabbage (Speech and Hearing Sci., Washington State Univ., 412 E Spokane Falls Blvd., Rm. 125P, Spokane, WA 99202, klcabbage@wsu.edu), Elaine R. Hitchcock (Commun. Sci. & Disord., Montclair State Univ., Bloomfield, NJ), Michelle T. Swartz (Speech Lang. Pathol., Thomas Jefferson Univ., Philadelphia, PA), and Thomas Carrell (Lincoln, NE)

Children with speech sound disorders (SSDs) demonstrate difficulty producing phonemes correctly and may exhibit poor speech perception compared to their age-matched peers; however, group differences in speech perception skills remain largely unexplained. Developmental models of speech production posit that children's ability to discriminate correct and incorrect productions in their own speech may be critical for developing accurate speech production. Historically, when children regularly mispronounce a phoneme, it has been essentially impossible to assess whether they perceive correct versus errors productions in their own speech, thus creating a clinical conundrum. How can we assess a child's ability to perceive the accuracy of their own phoneme production when they cannot produce a correct production? Recent technological developments allow for acoustic alteration of children's speech that digitally corrects speech sound errors while preserving natural characteristics of the child's voice. This machine-learning-based stimuli may then be used as training and feedback tokens for remediation when treating children with speech sound disorders. Such acoustic alteration is possible within an accessible, user-friendly environment that is clinically feasible for speech-language pathologists with little acoustic training. Thus, the purpose of this study is to evaluate the acoustic and perceptual accuracy of acoustically-altered child-speech compared to natural speech tokens.

5aSC18. Simulation of sentence-level speech with an acoustically driven model of speech production. Brad H. Story (Speech, Lang., and Hearing Sci., Univ. of Arizona, 1131 E. 2nd St., P.O. Box 210071, Tucson, AZ 85721, bstory@email.arizona.edu) and Kate Bunton (Speech, Lang., and Hearing Sci., Univ. of Arizona, Tucson, AZ)

In a model of speech production developed by Story and Bunton [JASA, 146(4), 2522–2528 (2019)] speech segments are encoded by specifying relative acoustic events along a time axis that consist of directional changes of the vocal tract resonance frequencies called resonance deflection patterns (RDPs). These events are transformed via acoustic sensitivity functions, into time-varying modulations of the vocal tract shape. For example, RDPs specifying /b, p/, /d, t/, and /g, k/ would typically be coded as $[-1\ -1\ -1]$, $[-1\ 1\ 1]$, and $[-1\ 1\ -1]$, respectively. The RDP changes, indicate, from left to right, the targeted directional shift of the first, second, and third resonances of the vocal tract. In addition, events that produce nasalization, voiced versus voiceless sounds, and changes in fundamental frequency are also specified the time axis. The purpose of this study was to demonstrate the use of this model to generate word- and sentence-level speech for eventual speech intelligibility studies. The process for simulating several words and sentences will be demonstrated.

5aSC19. Acoustic analysis as phonetic basis for phonological paraphasia in persons with aphasia: A preliminary study. J. Niranjana (Speech Lang. Sci., All India Inst. of Speech and Hearing, Kapila Ladies Hostel, AIISH, Mysore, Karnataka 570006, India, niranjanajkammath@gmail.com) and N. Hema (Speech Lang. Sci., All India Inst. of Speech and Hearing, Mysore, Karnataka, India)

Phonological planning deficit is considered to be the cause of phonemic paraphasia in persons with Aphasia. The errors are considered to be because of substitution of incorrect phoneme due to deficit in phoneme selection. However, the cascaded activation model suggests that phonemic paraphasia can result in the partial activation of the target and the competing phoneme that transcends to the lower articulatory level as a phoneme that reflects the acoustic properties of both sounds. Hence, this study aims to understand the existence of a phonetic basis for phonemic paraphasia in adults with Broca's Aphasia in the Tamil language. The study recruited five persons diagnosed with Broca's Aphasia in the age range of 50–60 years and five age and gender-matched controls. Confrontation naming task from WAB-R Tamil was used as the stimuli to elicit speech samples in affected individuals. All the recordings were done using PRAAT software. The burst duration and VOT of /p/ and /b/ sounds were compared between the experimental and control groups. Results suggested the presence of an acoustic trail of target sound in the substituted phoneme, challenging the existing literature on phoneme selection deficit at the planning level as the reason for the occurrence of phonemic paraphasias.

5aSC20. A module for automatic analysis of burst spectra for consonant place detection. Ella Tubbs (Res. Lab. of Electronics, Massachusetts Inst. of Technol., 77 Massachusetts Ave. Cambridge, MA 02139-4306, tubbs@mit.edu), Jeung-Yoon Choi, and Stefanie Shattuck-Hufnagel (Res. Lab. of Electronics, Massachusetts Inst. of Technol., Cambridge, MA)

Widespread methods for automatic speech processing have become increasingly powerful, but they do not aim to model aspects of human speech processing. Previous research has shown that individual acoustic cues are important in human speech perception. Developing a model that is capable of identifying individual acoustic cues in speech enhances our ability to extract meaningful information from a speech signal, irrespective of speaker variations or phonemic differences, in a way that provides a transparent and testable model of human speech processing. This research investigates a module for automatic analysis of the spectral burst cue in fricative and plosive speech sounds. The method determines the place of articulation of the spectral burst by utilizing spectral moment measurements near locations where a spectral burst is likely to occur as features in a Gaussian mixture model. This research lays the groundwork for a dynamic model that is developed to be consistent with the Bayesian belief updating framework, in alignment with prior work in human speech perception.

5aSC21. Effects of motor practice on the temporal coordination of articulatory movements for non-native onset clusters: Kinematic and acoustic evidence using electromagnetic articulography. Allen Shamsi (Linguist, Univ. of Florida, 289 Corry Village, Apt 10, Gainesville, FL 32607, allenshamsiev@ufl.edu), Matthew Masapollo (Psych., McGill Univ., Montreal, QC, Canada), Rachel Meyer, and Ratree Wayland (Linguist, Univ. of Florida, Gainesville, FL)

Research on cross-language speech production has shown that part of the challenge of producing non-native clusters arises from difficulties with temporally coordinating the successive consonantal gestures within a cluster. However, it remains unclear whether the application of practice-based motor learning paradigms can improve or stabilize this aspect of non-native cluster production. This study uses electromagnetic articulography (EMA) to measure the effects of motor practice on the temporal coordination of articulatory movements for non-native onset clusters. Monolingual speakers of American English intensively practiced producing monosyllabic sequences containing non-native onset clusters (e.g., MGAT) over two consecutive days. EMA was used to capture lingual, labial, and jaw motion during successive repetitions. For properly-sequenced cluster repetitions, analysis of the EMA sensor trajectories showed that, over the course of practice, there was a general reduction in inter-gestural timing variability and increased overlap between adjacent consonantal gestures. Furthermore, some of these improvements were maintained to the second day, indicating that subjects started forming durable performance gains. Acoustic analyses also showed a reduction in the duration of epenthetic vowel errors (/mgæt/ → /məgæt/) produced throughout practice. Collectively, the findings suggest that motor practice can improve the temporal coordination of articulatory gestures affiliated with non-native clusters.

5aSC22. Developing an algorithm to characterize context-based speech patterns as cue production profiles. Sofie C. Chung (Massachusetts Inst. of Technol., 50 Vassar St., Cambridge, MA 02139, scchung@mit.edu), Jeung-Yoon Choi, and Stefanie Shattuck-Hufnagel (MIT, Cambridge, MA)

The surface articulation of an underlying phoneme commonly fluctuates depending on the context. For instance, the associated acoustic cues for a /t/ phoneme produced at the beginning of an utterance may be distinct from the acoustic cues of a /t/ phoneme that is preceded and followed by a vowel. Thus, for an individual speaker or speaker group, we can investigate what specific phonemic contexts result in the associated acoustic cues to be produced. An algorithm has been developed which matches a target phoneme to its corresponding acoustic cues. For a given database, we can recover all of the contexts in which a phoneme was produced and tabulate the various acoustic cue production patterns that arose. Finally, we label these patterns by their speech production type (e.g., standard production, flapping, etc.) and produce an output in the form of a cue production profile. As a result, this profile would allow for the association of a set of acoustic cues in specific phonemic contexts for an individual speaker or speaker group, such as speakers from a specific dialect region. The algorithm can also be expanded to include prosodic cues to account for how stress and intonation can affect phoneme production.

5aSC23. Automatic detection and labeling of glides for the English and Spanish language. Edgar Morfin (MIT, 50 Vassar St., Cambridge, MA 02139, emorfin@mit.edu), Jeung-Yoon Choi, and Stefanie Shattuck-Hufnagel (MIT, Cambridge, MA)

This study outlines the development of a module for automatic detection of glide landmarks. We define glide landmarks as acoustic events that are observed during a narrowing in the oral cavity that does not result in cessation of airflow or in the conditions for turbulence noise. Glide landmarks are commonly associated with standard productions of underlying glides, a set that includes semivowels, liquids, and sounds produced by narrowing at the glottis, such as glottal stops, or the aspirant /h/. We lay out a framework that can be used to determine the acoustic measurements that are useful for detecting glide landmarks, and a Gausian Mixture Model is trained and tested, for automatic detection. A first investigation is carried out for /r/ production in English and Spanish. A closer look reveals that acoustic cues that describe abrupt onsets and offsets is useful for describing the trilled /r/ in Spanish, resulting in the need for an additional set of glide landmark cues for glide closure and glide release.

5aSC24. Analysis of physiological measures around conversational state changes. Benjamin Masters (Systems Design Eng., Univ. of Waterloo, 200 University Ave W, Waterloo, ON N2L 3G1, Canada, bpmasters@uwaterloo.ca), Susan Aliakbary Hosseinabadi, Dorothea Wendt (Eriksholm Res. Ctr., Snekkersten, Denmark), and Ewen MacDonald (Syst. Des. Eng., Univ. of Waterloo, Waterloo, ON, Canada)

The goal of this work is to extend the use of physiological measures of listening effort to interactive conversation. The initial work here investigates variations in head movement, eye gaze, and pupil dilation around conversational state changes during task-based conversations. Here, conversational state changes are defined as the points in time at which speakers start and stop talking. Windows around each of these types of state changes are analyzed for systematic differences of these parameters, which could be indicative of changes in attention and/or differences in speech production versus perception. Additionally, we calculate state change response functions, derived from a multivariate regression that maps from the state changes to the measured parameters and extracted features. The predictive power of these functions is explored, alongside comparisons of various considerations in their derivations. Our findings, based on data collected from 12 sets of interactive conversations taking place in varying levels of noise and simulated hearing loss, offer insight into how physiological responses during complex interactions can be measured and interpreted to infer when and where effort is directed throughout conversation.

5aSC25. Comparison of spectrograms and continuous wavelet transforms for multi-class classification of vocal pathologies by using convolutional neural networks. Bhawna Rathi (Music Technol., IUPUI, 535 W. Michigan St., IT371, Indianapolis, IN 46202, brathi@iu.edu) and Timothy Hsu (Music and Arts Technol., Indiana Univ. - Purdue Univ., Indianapolis, Indianapolis, IN)

The emergence of artificial intelligence has encouraged researchers to explore non-invasive methods for classifying vocal disorders of different pathologies. This paper introduces a framework designed to classify multiple distinct vocal pathologies, building upon the groundwork laid by the previous two-class classification research. This study uses datasets obtained noninvasively from the Indiana University (IU) Health Voice Center, implementing a Convolutional Neural Network (CNN) algorithm to differentiate vocal pathologies. In this approach, continuous wavelet transforms (CWT) and spectrogram images are derived from audio files of pathological voices and serve as inputs for the CNN classifier. Additionally, the research investigates the effects of data augmentation and the integration of a dropout layer, exploring how these variations affect the results. The findings reveal differing accuracy levels associated with parameter adjustments, such as learning rates and the different number of filter layers. Notably, CWT is found to yield higher accuracy compared to spectrograms. Furthermore, the study has achieved a high accuracy of 96% in multiclass classifications of diverse vocal pathologies within the training dataset, which underscores previous research advancements, demonstrating the feasibility of multiclass classifications.

5aSC26. Predicting vocal tract shape information from tongue contours and audio using neural networks. Sarah R. Li (Biomedical Eng., Univ. of Cincinnati, University of Cincinnati, 231 Albert Sabin Way, CVC, 3960, Cincinnati, OH 45267, lisr@mail.uc.edu), Alex Knapp, Jing Tang (Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH), Suzanne Boyce (Commun. Sci. and Disord., Univ. of Cincinnati, Cincinnati, OH), and T. Douglas Mast (Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH)

Midsagittal ultrasound imaging of the tongue is a portable and inexpensive way to provide articulatory information. However, although ultrasound images show a portion of the tongue surface, other vocal tract structures (e.g., palate) are not typically visible. This missing information may be useful for speech therapy and other applications, e.g., by characterizing vocal tract constrictions and informing how morphological variations affect speech patterns. Prediction of the vocal tract shape from information

available during ultrasound imaging (e.g., tongue contours and audio recordings) is, thus, potentially valuable. Recent advancements in articulatory prediction from audio recordings (i.e., acoustic inversion) and speech recognition using combined articulatory and acoustic data have used neural network models. Inspired by these models, this study investigates how well fusion of articulatory and acoustic features in speaker-independent models can predict expanded articulatory information. Specifically, recurrent neural network models will be trained to predict the vocal tract shape based on partial tongue contours and acoustic features, during production of vowels and central approximants. Features will be extracted from simultaneously recorded audio and 2D MRI (USC 75-Speaker Database). Different acoustic features and network architectures will be compared, with the goal of refining future models to predict vocal tract shapes during ultrasound imaging.

5aSC27. The effects of three-way consonant distinction in Northern Saami. Emily Posson (Linguist, Univ. of Minnesota - Twin Cities, 1000 SE 8th St., Apt. 10, Minneapolis, MN 55414, posso011@umn.edu) and Christopher Geissler (Linguist, Carleton College, Northfield, MN)

Of the world's languages, very few have been attested to possess a three-way consonant length distinction, among them Northern Saami (Uralic). However, the effects of this rare contrast on adjacent segments remain unexplored. This study reports on dialectal variation within Northern Saami and sheds light on the relationship between consonant length and the duration of preceding vowels. Three speakers were recorded: one Western Guovdageaidnu speaker and two Eastern Kárášjohka speakers. Target words were bisyllabic of the form $CV_V(C)$, with a medial short, long, or extra-long consonant, and were elicited in frame sentences. Results show dialectal differences in the number of consonant length distinctions, as well as the effects on preconsonantal vowels. The Western dialect exhibits three surface consonant lengths, while the Eastern dialect only has the longest and shortest. Interestingly, the Eastern speakers' extra-long C are systematically longer than their Western counterparts. Preconsonantal vowels vary with consonant length in the Western dialect, while Eastern preconsonantal vowels present a broader range of contrasts. These findings suggest that the Western dialect resembles other Saami languages with three consonant lengths (e.g., McRobbie-Utasi 2007). In comparison, the Eastern dialect may instantiate some of this contrast on the preceding vowel instead.

5aSC28. Clinical applications for automatic detection of creaky voice. Sarah R. Bellavance (Dept. of Commun. Sci. and Disord., New York Univ., 665 Broadway, 6th Fl., New York, NY 10012, srb664@nyu.edu) and Aaron Johnson (Dept. of Otolaryngol.-Head and Neck Surg., New York Univ. Grossman School of Medicine, New York, NY)

Creaky voice is a voice quality in which a low amount of subglottal air pressure, a condensed vocal fold structure, and a high closed quotient of vibration combine to create the auditory percept of a series of pulses at a low pitch. While this voice quality is often nonpathological, it can also co-occur with vocal pathologies. Identification of creak in the speech signal is most often done manually. Automatic creak detection algorithms have been created to streamline and produce replicable workflows. These algorithms have steadily increased in reliability, with COVAREP (Degottex *et al.*, 2014) as the newest state-of-the-art. While preliminary studies have demonstrated promising findings using artificial neural networks with clinical data, artificial neural networks typically improve with diverse data testing. The current study implements COVAREP on a novel dataset, both in terms of speakers and speech types. Deidentified patient diagnoses were matched to audio recordings collected from January 2021 through September 2023. Relevant portions of audio recordings were extracted using a Praat script, and COVAREP was implemented on the extracted audio files in MATLAB. Ongoing analyses correlating percentage of creak detected and vocal pathology diagnoses will be discussed. Finally, the results will be compared to those of previous work.

5aSC29. Comparing face-tracking action units with electromyography during speech. Hastiossadat Nozadi (Linguist, Univ. of BC, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, hnozad01@student.ubc.ca), Emma Irwin, Jamie Cheung, Yadong Liu, and Bryan Gick (Linguist, Univ. of BC, Vancouver, BC, Canada)

Video face-tracking software, such as OpenFace 2.0, can be used to make inferences about facial muscle activation [Baltrušaitis *et al*. IEEE 13, 59–66 (2018)]. However, the accuracy of these inferences based on the facial action units (FAUs) calculated by OpenFace 2.0 compared to corresponding muscle activity during speech is unclear. A previous study investigated muscle activation when both smile and speech occur simultaneously, focusing on the zygomaticus major (ZM) muscle [Liu *et al*. ISSP, 130–133 (2021)], but only presented data for a single speaker and did not compare FAU and EMG results. The present study compares OpenFace 2.0 action units with surface electromyography (EMG) data during speech in order to assess the validity of these inferences about facial muscle activation. We compare ZM activity and the lip corner puller FAU intensity results from a dataset collected for the previously mentioned [Liu *et al*. 2021] study. Data include four speakers producing read speech in smile conditions; 2 s will be extracted from EMG and FAU data before and after each utterance. Results will be reported on the relative accuracy of the FAU and EMG data. Implications will be discussed for speech communication research. [Work supported by NSERC.]

5aSC30. Feature generalizability for speaker-dependent detection of alcohol intoxication. Xinglei Liu (Tenvos Res. Labs, Sacramento, CA), Arian Shamei (Linguist, UBC, Vancouver, BC, Canada), and Rima Seiilova (Tenvos Res. Labs, Sacramento, CA, rima@tenvos.com)

Impairments to speech motor control from alcohol intoxication are variable across individuals, making speaker-dependent approaches ideal for speech-based intoxication detection [Schiel *et al*., 2010. Proc. INTERSPEECH 2010]. Here, we evaluated whether individual acoustic features have high generalizability across speaker-dependent models. We selected 97 speakers (54 male, 43 female) from the Alcohol Language Corpus [Schiel *et al*., 2012. LRE. 46, 503-521] who had sufficient sober and intoxicated (>0.08% blood-alcohol concentration) recordings for speaker-dependent modeling. For each speaker, we extracted 9 features from vowels (F0–F3, jitter, shimmer, harmonics-to-noise ratio, duration, and duration variability) and 7 from consonants (spectral skewness and kurtosis, center of gravity, duration and duration variability, harmonics-to-noise ratio), and fitted these to speaker-dependent random forest models with 5-fold cross-validation to evaluate feature importance from the associated mean decrease in Gini impurity (GI). Across all speakers, consonant-based features tended to have stronger generalizability than vowel-based features, with spectral skewness and kurtosis being the most generalizable (GI: 0.11 and 0.09), and vowel duration and F2 being the least generalizable (GI: 0.04 and 0.03). Further experiments to explore additional features and evaluate sex-specific generalizability are ongoing. [Research funded by Tenvos Incorporated for the development of commercial speaker state-detection algorithms.]

5aSC31. Real-time magnetic resonance imaging of velopharyngeal port posture during long pauses in naturalistic speech. Joshua O. Diebel (Linguist, Univ. of BC, 2511 East 2nd Ave., Vancouver, BC V5M1C7, Canada, joshdiablo135@gmail.com), Bryan Gick, Jahurul Islam (Linguist, Univ. of BC, Vancouver, BC, Canada), and Jade Weinstein (Speech Sci., Univ. of BC, Vancouver, BC, Canada)

Velum's behavior as a postural substrate in naturalistic speech is not well understood and is largely based on studies of inter-utterance velopharyngeal port posture (VPP). While existing literature [e.g., Gick *et al*., 2004 Phonetica, 61(4); Ramanarayanan, 2013 JASA, 134(1)] describes posture variations based on rest positions, ready positions, and inter-speech pauses, a less explored aspect is the behavior of the velum during extended pauses in naturalistic speech. Addressing this gap, the present study analyzes velum posture during naturalistic speech to characterize VPP during prolonged pauses. This study draws from a corpus of real-time magnetic resonance imaging (rtMRI) videos of L1 English speakers including those engaging in unstructured speech tasks. Results based on sequences corresponding to long pauses in naturalistic speech will be reported, outlining VPP

characteristics, and quantifying findings in terms of time spent in and shifts between velum positions. Implications for motor control differences in naturalistic and elicited speech data will be discussed.

5aSC32. Exploring facial gestures and visual speech articulation cues during the production of Canadian English voiced stops. Theresa Rabideau (Linguist, Univ. of Ottawa, 25 Elterwater Ave. Apt #14, Ottawa, ON K2H5J1, Canada, trabi037@uottawa.ca) and Suzy Ahn (Dept. of Linguist, Univ. of Ottawa, Ottawa, ON, Canada)

Decades of research on visual information for speech has demonstrated the informativeness of visual cues to enhance (Cho *et al*., 2020; Kawase *et al*., 2014) and influence (MacDonald and McGurk, 1978) speech perception. However, it is unclear which specific visual cues are spatially and temporally correlated to certain features of speech segments. This study explored visual cues of voicing during the production of Canadian English stops using facial recognition technology (Baltrušaitis *et al*., 2018) and manual coding (using ELAN 2022). We recorded the audio along with two videos, capturing both front and side views simultaneously, from six native Canadian English speakers. We paid special attention to the throat (larynx), chin, and neck areas which have been understudied by the previous literature. Preliminary data shows expanding movement in the submental triangle and throat during the production of voiced stops compared to voiceless stops. This finding supports tongue body lowering and larynx lowering found in the production of English voiced stops (Westbury, 1983). The comparison between utterance initial stops with post-vocalic stops shows that certain visual cues may be related to phonological voicing categorization irrespective of actual voicing during closure, while others reflect phonetic voicing reality.

5aSC33. Performance analysis of a dilated attention fast GAN for speech enhancement. Vahid Ashkani (Western Univ., 14-534 Platt's Ln., London, ON N6G 3A8, Canada, vashkani@uwo.ca) and Vijay Parsa (Western Univ., London, ON, Canada)

Recent advancements in speech enhancement have witnessed the emergence of generator-based methodologies. However, several of these approaches exhibit complexity in handling input variations, either excelling at low signal-to-noise ratios (SNRs) by utilizing intricate representations of noisy and clean speech or demonstrating superior performance only at higher SNRs. In this work, we investigated speech enhancement using a Dilated Attention Fast Generative Adversarial Network (DAF-GAN). The proposed DAF-GAN framework achieves stability in performance across different SNR conditions by efficiently processing large-scale signal lengths. The DFS-GAN features a dilated discriminator model operating via patches. The generator architecture incorporates multi-decoding and attention gates facilitated through skip-connections, strategically integrated within the Fast-U-Net model to optimize processing speed. An ideal ratio mask was used in the test phase to further refine the enhanced signal by emphasizing target speech while suppressing residual noise or artifacts. The DAF-GAN performance was assessed using objective metrics such as PESQ on a number of noisy speech databases. Results revealed that the DAF-GAN performed modestly in comparison with the state-of-the-art models. For example, analyses of the VoiceBank-DEMAND dataset yielded a PESQ score of 2.50 for the DAF-GAN.

5aSC34. Influence of airflow and ligament tension on the acoustics of a biomimetic larynx model. Bogac Tur (Phoniatrics, Univ. Hospital Erlangen, Waldstrasse 1, Erlangen 91054, Germany, bogac.tur@uk-erlangen.de), Lucia Gühring, Olaf Wendler, and Stefan Kniesburges (Phoniatrics, Univ. Hospital Erlangen, Erlangen, Bavaria, Germany)

The phonation process is a complex interaction involving the airflow from the lungs, the oscillation of the vocal folds' tissue, and the resultant acoustics. To understand the underlying physical mechanisms, an experimental model has been designed that allow the control of flowrate and longitudinal tension of the vocal folds. A synthetic biomimetic larynx model was applied that features airflow-driven vocal folds' oscillations. The longitudinal stiffness of the vocal folds is controlled using embedded ligament fibers. The model enables to measure aerodynamic and acoustic signals as function of airflow rate and the fiber tension. Based on the measured signals, the influence of airflow and fiber tension was statistically analyzed using the parameters F0, Psub, CPP, HNR, Shimmer, and Jitter. The statistical analysis revealed that both flowrate and fiber tension significantly influence acoustic and aerodynamic parameters. In general, the parameters showed different trends for increasing flowrate and fiber tension. Increase in fiber tension produced increasing parameters up to a maximum tension level followed by a saturation of the parameters. In contrast, the flowrate showed varying trends depending on the respective parameter. The results clearly show how flowrate and longitudinal tension control the phonation process and the resulting acoustic quality.

5aSC35. Analysis of potential influencing factors for acoustic quality in patients with ectodermal dysplasia. Marion Semmler (Div. of Phoniatrics and Pediatric Audiol., Univ. Hospital Erlangen, Waldstrasse 1, Erlangen 91054, Germany, marion.semmler@uk-erlangen.de), Bogac Tur, Ludger Schlautmann, Sophie Wolfsteiner, Laura Ziller, Ann-Katrin Hellmann, Maximilian Eckhardt, Olaf Wendler, and Anne Schützenberger (Div. of Phoniatrics and Pediatric Audiol., Univ. Hospital Erlangen, Erlangen, Bavaria, Germany)

Subjects with ectodermal dysplasia (ED) suffer from an inherited disorder in the development of ectodermal structures. They show a significantly reduced formation of teeth, hair, and a reduced number and activity of sweat and salivary glands. Recently, the voice of ED subjects has come into focus. It is assumed that the generally reduced glandular function is responsible for the altered vocal sound, although no specific cause has been identified. Previous findings included significant changes in the acoustic quality, the parameters derived from high-speed videoendoscopy, the laboratory analysis and rheological analysis of saliva samples as a substitute for laryngeal mucus. Based on these findings, we performed an extended statistical analysis directed towards the correlations between the different affected domains in order to reveal the contributing factors for the resulting acoustic voice quality. Although there were distinct statistical differences between ED males and male controls in the individual domains, only few statistical correlations were found between the subdomains i.e., acoustics, vocal fold vibrations and saliva composition/consistency. These results do not yet allow any definitive conclusions on the influencing factors of voice quality. A larger group of test subjects and analyses of the laryngeal mucus are required.

5aSC36. Comparing speech to fine and gross motor skills in Parkinson's patients. Brian Diep (Linguist, Univ. of BC, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, brdiep@mail.ubc.ca), Sylvia Cho (Linguist, Simon Fraser Univ., Burnaby, BC, Canada), Arian Shamei, and Bryan Gick (Linguist, Univ. of BC, Vancouver, BC, Canada)

Parkinson's Disease (PD) is a neurodegenerative motor disorder resulting from damage to dopaminergic neurons. The goal of this study is to evaluate the correspondence between speech and non-speech motor impairments. To explore this, we extract features from the mPower dataset [B. M. Bot *et al*., Sci Data 3, 160011 (2016)] containing mobile data from PD patients and healthy controls along with their performance on a vowel phonation, finger tapping, and walking task. We hypothesize that there is a shared motor system underlying each of these modalities and that disease progression will manifest in impairments to both speech and non-speech systems that rely on motor control. For acoustic features, we measure temporal consistency via F0-independent features (shimmer, jitter, and harmonics-to-noise ratio). For non-acoustic tasks, we adapt this set to measure spatial consistency and accuracy in finger tapping or walking. We perform clustering and multidimensional scaling (MDS) on our features to understand their correspondence across the modalities. Results will be reported with relevance to the relationship between PD and its effects on articulatory and general motor processes.

5aSC37. Using prosody to produce trust and doubt. Abbey L. Thomas (Brain and Behavioral Sci., The Univ. of Texas at Dallas, 800 W Campbell Rd., Richardson, TX 75080, abbey.thomas@utdallas.edu)

Work presented at a 2022 ASA meeting [Thomas, 2022] demonstrated that American English talkers use acoustic prosodic variables to distinguish attitudes of incredulity and trust from a "neutral tone of voice" when asking

WH-questions. The present study largely replicates these findings in new data. These data comprise both WH-questions and Yes/No questions, read by a new set of talkers in six imagined scenarios. In the previous study, "doubt" was likely confounded with "authority" due to the imagined scenario used to elicit the stimuli. In the present study, the three attitudes (doubt, neutral, trust) were crossed with dominance (participant was +/− dominant relative to their imagined conversational partner). This study examines additional acoustic variables, including the variability of word duration within utterance and the F0 slope of the longest word in each utterance. Averaging across the +/− dominance conditions and the utterance type (yes/no or WH question), duration-based variables reliably distinguished doubting from neutral utterances (with doubting utterances showing greater duration and greater variability in word duration). In contrast, F0-based variables differentiated trusting from neutral utterances, with trusting utterances showing a higher F0 over the course of the entire utterance.

5aSC38. Broad and fine acoustic categories in *bod*, *bond*, *bald*, and *bard*: A step toward acoustic phonology. Matthew C. Kelley (English, George Mason Univ., 4400 University Dr., Fairfax, VA 22030, mkelle21@gmu.edu)

Acoustics is central to the study of speech communication, but it is conspicuously under-represented in abstract representations of speech. Many flavors of phonological analysis tend toward articulatory descriptions, and transcriptions focus on strings of articulatory actions. All this is despite acoustics being easier to measure than articulation with current technology. The present study explores basic concepts for an acoustic phonology, with two types of postulated categories: broad and fine. Resonant, turbulent, transient, and occludent types of sounds comprise the broad categories, as general methods of filtering the speech source. Fine categories are conceptualized as specific types of acoustic actions within a broad category. These acoustic actions are goal-oriented, as for achieving a particular acoustic effect like the presence of antiformants or a lowered F2 or F3. However, these actions are not explicitly restricted to manipulating traditional phonetic features like formants. By default, fine categories are assumed to be produced in parallel when possible, yielding overlap effects like anticipatory nasalization, lateralization, and rhotacization. These concepts are explored in a microanalysis of *bod*, *bond*, *bald*, and *bard* from the speaker in the Massive Auditory Lexical Decision data set, with an eye to seeding the ground for a future acoustic phonology.

5aSC39. Respiratory laryngeal coordination during vowel-plosive-vowel transition in shouted speech. Zhaoyan Zhang (UCLA School of Med., 1000 Veteran Ave., 31-24 Rehab. Ctr., Los Angeles, CA 90095, zyzhang@ucla.edu)

This study investigates differences in respiratory-laryngeal coordination between normal and shouted speech in the production of a vowel-plosive-vowel utterance, with the goal toward synthesis of such utterances in loud speech. The results showed that compared to normal condition, shouted speech produced higher intraoral pressure during vocal tract closure, higher airflow at the release of vocal tract closure, increased fundamental frequency and increased duration of glottal closure during the following vowel, and increased respiratory activities. For unvoiced plosives, shouted speech also produced increased voice onset time of the following vowel, but its importance to the perception of shouted speech was small. Computational simulations further showed that such increased voice onset time is likely due to increased maximum vocal fold abduction and/or delayed initiation of vocal fold adduction, which are necessary to avoid voice onset immediately after the release of vocal tract closure in conditions of high subglottal pressure.

5aSC40. Principal dimensions of laryngeal vocal control in a computational model of voice production. Zhaoyan Zhang (UCLA School of Medicine, 1000 Veteran Ave. 31-24 Rehab Ctr., Los Angeles, CA 90095, zyzhang@ucla.edu)

Voice production is constrained by laryngeal physiology and physics. Such constraints may present themselves as principal dimensions that are shared among speakers in how they produce and perceive voice. In this study, we attempt to identify such principal dimensions in voice outcome measures and the underlying laryngeal control mechanisms in a three-dimensional computational model of voice production. A large-scale voice simulation was performed with parametric variations in vocal fold geometry, stiffness, glottal gap, and subglottal pressure. Principal component analysis was applied to data combining both the laryngeal control parameters and acoustic and aerodynamic voice outcome measures. The results showed two dominant dimensions of vocal control. The first dimension describes interaction between respiration and vocal fold adduction in a way that increases glottal flow amplitude and vocal intensity at the cost of decreasing high-frequency harmonic production. The second dimension mainly describes control of medial surface thickness and glottal closure, which allows simultaneous increase in vocal intensity and high-frequency harmonic production but also increases risk of vocal fold injury. A third dimension with a reduced weight describes control of the fundamental frequency. The importance of these principal control dimensions to vocal expression of emotion and vocal health is discussed.

5aSC41. Spectral energy properties of falsetto voice. Yuan Chai (Linguist, Univ. of Washington, Guggenheim Hall 4th Fl. Box 352425, Seattle, WA 98105, yuanchai@uw.edu) and Patricia A. Keating (Linguist, UCLA, Los Angeles, CA)

Breathy voice and falsetto voice have distinct articulatory mechanisms and auditory percepts. They clearly differ in their noise spectrum and typical f0. However, previous literature has described the harmonic energy profile of each in similar terms, namely high energy in H1 and low energy in higher-frequency harmonics. Are the harmonic spectra of these two voice types in fact the same? We will provide a quantitative comparison of the harmonic energy in falsetto versus breathy (and also modal and creaky) voice using a corpus of readings by Taiwan Mandarin speakers of a "Little Red Riding Hood" story where they enacted character voices. We annotated each vowel in this corpus for its perceived phonation type. In raw values, $H1^*-H2^*$ is the same in falsetto and breathy voice, but when f0 is controlled statistically, $H1^*-H2^*$ is lower in falsetto than in breathy voice (falsetto has less falloff from H1 to H2). $H1^*-H5k^*$, a measure of overall spectral tilt, is the same in falsetto and breathy voice when f0 is controlled, though falsetto voice has overall higher energy than breathy. Other harmonic differences will be presented in detail, along with comparisons on other voice measures.

5aSC42. Duration imitation is not mediated by phonological contrast: Evidence from a checked-unchecked tonal contrast in Taiwanese Southern Min. Wei Zhang (McGill, 3620 Rue Lorne Crescent, 318, Montreal, QC H2X2B1, Canada, weizhang201707@gmail.com), Meghan Clayards (McGill, Montreal, QC, Canada), and Yu-An Lu (National Yang Ming Chiao Tung Univ., Taipei, Taiwan)

Phonetic imitation is mediated by phonological contrast, as evident in features such as formant, VOT and F0. However, a recent study observed that duration imitation was not mediated by phonological contrast. In contrast to other studies, duration served as a non-primary cue to the phonological contrast in this recent study. This current study further investigates duration imitation in a case where duration serves as the primary cue. We utilized the tonal contrast of T3 versus T33 in Taiwanese Southern Min (TSM), to which duration was identified as the primary cue. We created a seven-step tonal continuum between T3 and T33 by manipulating the tone durations, and recruited seventeen native TSM speakers to imitate each step as closely as they could. The bi- or uni-modality of the distribution of the imitated durations for all seven steps was analyzed using Bayesian regression models. Results showed that the imitated durations were more consistent with an unimodal distribution, suggesting that, unlike other features, the imitation of duration is not mediated by tonal contrast, whether it acts as a primary cue or not. Thus, features exhibit different resistance to phonological mediation in phonetic imitation.

5aSC43. You good?: Examining the role of intonation and eyebrow movements in sentence type distinction. Kendall Lowe (Linguist., Univ. of Michigan, 611 Tappan St., Ann Arbor, MI 48109, loweke@umich.edu), Yoonjeong Lee, Jelena Krivokapić, and Natasha Abner (Linguist., Univ. of Michigan, Ann Arbor, MI)

By analyzing the pitch accents, edge tones, and eyebrow movements of African American English (AAE) speakers, this study examines the coordination of prosody and co-speech gestures. Twelve self-identified AAE speakers engaged in scripted dialogues with an AAE-speaking confederate via Zoom. Half of the dialogues contained the polysemous AAE idiomatic phrase "you good" which has declarative and interrogative forms hypothesized to be distinguished by various eyebrow and intonation patterns. The other dialogues included non-idiomatic phrases semantically related to their "you good" equivalents. These dialogues were designed to explore whether the coordination patterns for the idioms occur with non-idiomatic utterances of the same sentence type. Acoustic data is prosodically annotated using the MAE-ToBI transcription system and visual eyebrow data is tracked using OpenFace 2.0, a facial landmark detection toolkit. The derived eyebrow movement trajectories are time-aligned with the pitch trajectories for the semi-automatic labeling of multimodal signals. The temporal relationship between eyebrow movement landmarks, pitch accents, and edge tones is examined to investigate how different meanings and sentence types are created through their coordination. Preliminary examination indicates intonationally similar contours for the idiomatic and non-idiomatic interrogative phrases, but shared and consistent intonational contours are not evident in the declarative data.

5aSC44. The impact of remote microphones and facial masks on speech production and conversational behaviors in hearing-impaired individuals. Menatalla K. Ellag (Syst. Des. Eng., Univ. of Waterloo, 200 University Ave W, Waterloo, ON N2L3G1, Canada, mkellag@uwaterloo.ca), Jinyu Qian, Ieda Ishida (Sonova Canada, Mississauga, ON, Canada), and Ewen MacDonald (Syst. Des. Eng., Univ. of Waterloo, Waterloo, ON, Canada)

This study investigates the acoustics and conversation behavior of the speech produced during conversations among groups of hearing-impaired individuals. Four groups of four hearing-impaired individuals, all using hearing aids, engaged in discussions on provided topics in the presence of background noise. Conversations were held in four conditions based on two factors (using versus not using a remote microphone; wearing versus not wearing a face mask). Analysis of recorded conversations focused on speech production measures (e.g., fundamental frequency, articulation rate, formant frequencies, etc.) and conversational behaviors (e.g., inter-pausal unit length, floor-transfer offsets, turn duration, etc.). Although both influence the potential difficulty of holding a conversation, distinct effects of mask, remote microphone, and their interaction were observed for measures of speech production and conversational behaviors.

5aSC45. The effect of task on speech production and conversation behavior during conversation. Menatalla K. Ellag (Syst. Des. Eng., Univ. of Waterloo, 200 University Ave. W, Waterloo, ON N2L3G1, Canada, mkellag@uwaterloo.ca), Kate Avison, and Ewen MacDonald (Syst. Des. Eng., Univ. of Waterloo, Waterloo, ON, Canada)

The purpose of this study was to investigate how native-English, healthy-hearing individuals adapt their speech production and conversation behavior in the presence of noise and how this can vary based on conversational goal. Pairs of participants engaged in both free-form conversations as well as conversations based on solving a task (a "spot the difference" task using the Diapix UK pictures). Although seated in separate rooms, talkers could communicate via headset microphones and headphones with gains set to simulate levels that would be present if they were seated in the same room. The effects of task and noise on measures of speech production (e.g., articulation rate, speech level, etc.) and conversational behaviors (e.g. floor transfer offsets, turn length, etc.) are investigated. These results provide insights into how to infer listening effort via acoustical measures of communication in a broader range of settings.

5aSC46. Voice analysis for intoxication detection in laboratory versus law enforcement contexts. 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)

There is substantial interest in deploying voice biomarkers for the detection of alcohol intoxication, yet it remains unknown how other mental and physical states (e.g. emotion, stress) influence voice biomarkers in intoxicated speech. We compared measurements of voice quality (jitter, shimmer, noise-harmonics ratio) across two datasets of alcohol-intoxicated speech: (1) The alcohol language corpus, which contains laboratory elicited speech from 167 individuals in both sober and intoxicated conditions and (2) a custom dataset of police (control, $n = 14$) and suspect (intoxicated, $n = 32$) interactions during traffic stops where intoxication was verified via breath analysis. Measurements were extracted from all stressed vowel tokens and compared across conditions using two-sample t-tests within sex-specific groupings of each dataset. For both males and females, jitter was significantly lower during intoxication as measured from laboratory-elicited speech, but significantly higher for intoxicated individuals when measured from police–suspect interactions. These results suggest that voice biomarkers for alcohol-intoxication are easily confounded by other emotional and physical states (e.g., stress during police interaction), and thus, present a particular challenge for speaker-independent detection systems where baseline voice quality measurements across different emotional states are unknown. [Research funded by Tenvos Incorporated for the development of commercial speaker state-detection algorithms.]

Session 5aUW

Underwater Acoustics: Underwater Sonar and Communication

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Contributed Papers

8:00

5aUW1. Sonar image generation using time-domain scattering echo simulated by Kirchhoff approximation extensions. Yan Wu (Shanghai Jiao Tong Univ., No. 800, Dongchuan Rd., Minhang District, Shanghai 200000, China, wuyan01@sjtu.edu.cn) and Jun Fan (Shanghai Jiao Tong Univ., Shanghai, China)

Sonar images are a topic of great interest in many underwater applications, such as detecting and classifying proud targets. Given the high cost of experimental measurement and the lack of measurement data, especially the difficulty in obtaining non cooperative target image data, a sonar image generation method of target and seabed joint modeling based on echo simulation is proposed in this work. The single and multiple scattering contributions arising from the target, the seabed, and their interactions have been derived under the Kirchhoff approximation extensions solution. And the occlusion effect is considered in the calculation, which is reflected in the sonar image as the target shadow. This study demonstrates that images generated based on echo can provide more complete target feature information and can be used as an alternative method to obtain sonar image quickly and accurately.

8:15

5aUW2. Evaluating underwater communication: A atatistical analysis of shallow water acoustic channels. Hammad Hussain (Electron. System, Norwegian Univ. of Sci. and Technol., Trondheim 7491, Norway, hammad.hussain@ntnu.no) and Syed Sajjad H. Zaidi (PNEC, National Univ. of Sci. and Technol., Islamabad, Pakistan)

Precise channel information is crucial for effective communication systems, particularly in underwater environments. This paper focuses on simulating acoustic channels in underwater communication, using an existing statistical method instead of introducing a new one. The traditional bellhop method cannot account for channel variations, whereas the statistical-based channel simulator we utilize can handle both small-scale and large-scale fading phenomena. Large-scale fading includes uncertainties such as location, time-varying environmental statistics, and variations in received power. Small-scale fading has Doppler shifts due to motion (e.g., vehicular or wave motion) and scattering, rapidly altering channel statistics. To assess the effectiveness of the statistical-based simulator, we used the five-element underwater acoustic dataset collected from an experiment on November 6, 2009. Our findings indicate that this simulator, which leverages the existing statistical method, outperforms the bellhop method in accurately modelling the acoustic channel. This approach shows potential for improving the design and deployment of underwater communication systems.

8:30

5aUW3. Extraction of comprehensive feature for active target detection using two hand-crafted acoustic features. YoungSang Hwang (Dept. of Ocean Syst. Eng., Sejong Univ., 209, Neungdong-ro, Gwangjin-gu, Seoul 05006, Republic of Korea, ldzezl@naver.com), Geunhwan Kim, Wooyoung Hong, and Youngmin Choo (Dept. of Ocean Systems Eng., Sejong Univ., Seoul, Republic of Korea)

We propose a model for creating a new feature suitable for classifying targets and non-targets using a small active sonar data. The proposed model extracts a comprehensive feature from two different hand-crafted features using deep-learning (DL). To extract comprehensive feature, we contrived a complementary learning module (CLM), which consists of two stages. In the first stage, a transformer encoder was adopted to reinforce the input features and then complementarily learned through attention mechanism in the following stage. The output features from the CLM are passed through a shallow CNN to classify targets and non-targets. In addition, an uncertainty quantification method is proposed using two CLMs to quickly add new samples acquired in the real ocean dataset. Using the decisions from two CLMs, we calculate the variance and determine which samples from the entire dataset are worthy of prioritized addition. Two real-ocean datasets were used to examine the model generalization and compare the classification results with conventional DL models. The generalization performance of the proposed model was comparable or superior to other DL models and uncertainty quantificaion method improved the generalization performance. [Work supported by Korea Research Institute for defense Technology planning and advancement(KRIT)—Grant funded by the Korea government (DAPA—Defense Acquisition Program Administration) (No. KRIT-CT-23-026, Integrated Underwater Surveillance Research Center for Adapting Future Technologies, 2024)]

8:45

5aUW4. Acoustic channel estimation via adaptive regularized NLMF algorithm for underwater communication. Hammad Hussain (Electron. Syst., Norwegian Univ. of Sci. and Technol., Trondheim 7491, Norway, hammad.hussain@ntnu.no) and Syed Sajjad H. Zaidi (PNEC, National Univ. of Sci. and Technol., Islamabad, Pakistan)

Estimating shallow underwater acoustic channels is challenging due to the time-varying and sparse nature of the multi-path profile. This paper proposes a novel training-based channel estimation technique that leverages the efficient time-varying regularized normalized least mean fourth (R-NLMF) algorithm. We demonstrate that this algorithm effectively caters to the sparse structure of the channel and significantly improves the performance of the channel estimator. The proposed technique is validated using experimental data from shallow underwater acoustic experiments conducted on Nov 6, 2009, named the "Five-Element Underwater Acoustic dataset." Our results show that the adaptive learning rate and regularized NLMF algorithm provide accurate channel estimation and effectively mitigate the effects of large-scale and small-scale fading.

9:00

5aUW5. Waveguide invariant navigation of an AUV with a towed line array. Junsu Jang (Scripps Inst. of Oceanogr., UC San Diego, 9679 Caminito Del Feliz, San Diego, CA 92121, jujang@ucsd.edu) and FLORIAN MEYER (Univ. of California San Diego, La Jolla, CA)

Passive acoustics navigation of an autonomous underwater vehicle (AUV) can reduce localization errors while being relatively low-cost. A promising approach for navigation in shallow water is to exploit the waveguide invariant, which makes it possible to estimate the range between an acoustic source with a known position and an acoustic receiver on the AUV. This range estimate can then be fused in a sequential Bayes framework to improve the navigation information of the AUV. While previously proposed methods consider a range-independent environment with a single acoustic source and a single receiver, we investigate a more realistic scenario with multiple sources in a potentially range-dependent environment. In particular, a shallow water environment with a constant slope and sound speed is considered. To resolve multiple sources, it is assumed that the AUV tows an acoustic line array. Simulated acoustic data provided by a normal mode program are used to study the feasibility of the AUV navigation in shallow water by exploiting the waveguide invariant in this setting.

9:15

5aUW6. Performance analysis of passive acoustic ranging using a single hydrophone in shallow waters. Jeong Bin Jang (Korea Univ. of Sci. and Technol., 32, Yuseong-daero 1312 beon-gil, Yuseong-gu, Daejeon 34103, Republic of Korea, jangjb@kriso.re.kr) and Sung-Hoon Byun (Korea Res. Inst. of Ships and Ocean Eng., Daejeon, Republic of Korea)

Passive source range estimation is one of the important research fields on the sonar system. Recent research has carried out range estimation using single hydrophone. This study presents the result of analyzing the impact of environmental conditions on range estimation performance when estimating the range of a moving source with a single hydrophone. Range estimation of a moving source using a single hydrophone estimates source velocity from the cross-correlated fields of the signal measured at time intervals. This method of estimating the source range from estimated source velocity is used. Therefore, it is important to accurately estimate the source velocity for ranging, and the accuracy of estimating the source velocity is affected by the propagating mode characteristics in waveguide. This study investigates the changes in velocity estimation performance of a moving source according to various underwater environment and mode formation characteristics through simulation analysis and discusses ways to reduce velocity estimation error. [Work supported by KRIT (Contract No. 22-305-B00-001).]

9:30–9:45 Break

9:45

5aUW7. Generalization performance analysis of anomaly detection-based active sonar classifier using anomaly score landscape. Geunhwan Kim (Dept. of Ocean Syst. Eng., Sejong Univ. 209, Neungdong-ro, Gwangjin-gu, Seoul 05006, Republic of Korea, kimgw200@sejong.ac.kr), YoungSang Hwang, Keunhwa Lee, and Youngmin Choo (Dept. of Defense Syst. Eng., Sejong University, Seoul, Republic of Korea)

This study explores generalization performance of anomaly detection-based active sonar classifier using the deep neural network (NN). Despite its superior performance, NN remains challenging to interpret its decision due to black-box nature, which, consequently, makes it hard to trust the results. To address this, we employ the loss landscape method used to analyze the generalization performance of supervised learning-based NN. Owing to the inapplicability of the conventional loss landscapes to anomaly detection-based classifier, we propose a novel anomaly score landscape for the anomaly detection-based classifier. Experimental results using active sonar datasets reveal that anomaly detection-based classifier maintains consistent landscape structures between train and test datasets, compared to supervised learning counterparts. This study provides insights into the interpretability and generalization capabilities of supervised- and anomaly detection-based active sonar classifiers.

10:00

5aUW8. Estimation of sound source direction by holographic method in the presence of shallow water internal waves. Sergey A. Pereselkov (Math. Phys. and Information Technol., Voronezh State Univ., Russia, Voronezh, Universitetskay pl, 1, Voronezh 394018, Russian Federation, pereselkov@yandex.ru), Venedikt Kuz'kin (Sci. Ctr. for Wave Res., General Phys. Inst., Moscow, Russian Federation), Nikolai Ladykin, and Alexey Pereselkov (Math. Phys. and Information Technol., Voronezh State Univ., Voronezh, Russian Federation)

A holographic signal processing method for estimation of sound source direction in the presence of intensive internal waves (IIWs) is considered in the paper. It is assumed that IIW propagate along acoustic track between source and receiver. The direction estimation is based on holographic processing [Pereselkov S. and Kuz'kin V., JASA 151(2), 666–676] of vector receiver (VR) channels (x-th and y-th). The two source interferograms (x-th and y-th) is generated in frequency-time domain. The results of the two-dimensional Fourier transformation (2D-FT) of interferograms are source holograms (x-th and y-th). The direction estimation is calculated by ratio of the absolute values of the holograms angle distributions. Numerical experiment results of source direction estimation in the presence of IIW for low-frequency band (150-250 Hz) are considered. It is shown that holograms (x-th and y-th) consist of two disjoint regions corresponding to nonperturbed field and field perturbation by IIW. This structure of the holograms allows to separate the unperturbed field component in absence of IIW (x-th and y-th). So, IIW influence on direction estimation can be reduced significantly. The error in direction estimation by holographic method caused by IIW is analyzed in the paper. [Work supported by grant from the Russian Science Foundation 23-61-10024.]

10:15

5aUW9. Holographic signal processing for sound source depth estimation by single receiver in shallow water. Sergey A. Pereselkov (Math. Phys. and Information Technol., Voronezh State Univ., Russia, Voronezh, Universitetskay pl, 1, Voronezh 394018, Russian Federation, pereselkov@yandex.ru), Venedikt Kuz'kin (Sci. Ctr. for Wave Res., General Phys. Inst., Moscow, Russian Federation), Pavel Rybyanets, and Alexey Pereselkov (Math. Phys. and Information Technol., Voronezh State Univ., Voronezh, Russian Federation)

The holographic method for estimating of the low-frequency broadband source depth by single receiver in a shallow water is developed in the paper. By using broadband signals from receiver, the source interferogram (sound intensity distributions) is generated in frequency-time domain. Within framework of holographic signal processing (Pereselkov S. and Kuz'kin V., JASA 151(2), 666–676) is analyzed by two-dimensional Fourier transform (2D-FT). The result of the 2D-FT is called the Fourier hologram (source hologram). The structure of source hologram allows to separate noise and spectral spots relating to modes interference. By using inverse 2D-FT, the cleared source interferogram with information of the mode amplitudes is reconstructed. The ratio of neighboring modes amplitudes allows to estimate the source depth. Results of a numerical experiment of source depth estimation by holographic method in the low-frequency band (100–300 Hz) are analyzed. The stability of the holographic method to errors in measuring of mode amplitudes and variations of the waveguide parameters are considered. It is shown that error estimation of the source depth tends to the established value with increasing noise level. The qualitative and quantitative explanations of the simulation results are presented in the paper. [Work supported by grant from the Russian Science Foundation 23-61-10024.]

10:30

5aUW10. Passive acoustic localization using dictionary learning. Ivan I. Rodriguez-Pinto (Littoral Acoust. & Target Phys., Naval Surface Warfare Ctr. Panama City Div., 110 Vernon Ave. 2C09B, Panama City, FL 32407, ivan.i.rodriguez-pinto.civ@us.navy.mil) and Raymond Lim (Littoral Acoust. & Target Phys., Naval Surface Warfare Ctr. Panama City Div., Panama City, FL)

Ray-trace simulations are used to demonstrate the use of Dictionary Learning to improve passive localization of a source monitored with a vertical linear acoustic array. The learned dictionary, generated using historical sound velocity profile (SVP) data from a region of interest, is an over complete, sparse representation of the SVP training set. By minimizing an objective function that measures the differences between multipath reception intervals detected at a receiver for propagation through candidate and baseline SVP profiles, we show that an optimal SVP match to the "unknown" baseline can be reconstructed by an efficient dictionary search. Ray traces back-propagated through the reconstructed SVP according to beamformed receive angles computed with the baseline SVP are then found to well estimate the source position as the centroid of a cluster of multipath signal intersections. The accuracy for small unit-sparsity dictionaries of size up to 50 was evaluated on a randomly sampled, 30 SVP testing set obtained from an area close to that of the training set, demonstrating mean location errors less than 5% in both distance and depth. Five representative profiles spanning the range of observed sound speed variations were used to assess localization performance at source depths from 50 to 450 m and at source ranges from 2000 to 4500 m. Compared to using the average sound speed profile to estimate source position, using the learned dictionary produced a general performance increase in position accuracy.

10:45

5aUW11. Acoustic diversity, bearing estimation, and localization of sound sources in shallow water and deep ocean off the U.S. Northeast coast using a coherent hydrophone array. Sai Geetha Seri (Elec. Eng., Northeastern Univ., 360 Huntington Ave. Boston, MA 02115, seri.s@northeastern.edu), Max Radermacher (Northeastern Univ., Boston, MA), Hamed Mohebbi-Kalkhoran (Mech. Eng., Massachusetts Inst. of Technol., Cambridge, MA), and Purnima R. Makris (Elec. and Comput. Eng., Northeastern Univ., Boston, MA)

An in-house developed 160-element large-aperture hydrophone array from Northeastern University was utilized for ocean monitoring in both the shallow waters of the Great South Channel (GSC) and the deep-water area off the continental slope south of Rhode Island in September 2021. Utilizing passive ocean acoustic waveguide remote sensing (POAWRS) technology, the system enabled real-time, broad-area surveillance of the oceanic environment. A novel multistage clustering was introduced for annotating the vast volume of underwater acoustic data, drawing inspiration from the human cognitive process of categorizing sounds. Initially, sounds are separated into broad groups, akin to a first human listening pass. Subsequent stages of clustering refine these categories, utilizing different sets of features to achieve a granularity of annotation that closely mirrors expert human annotation processes. It was found that the acoustic landscape of the GSC was primarily composed of diverse marine mammal sounds, such as fin whale 20 Hz pulses, humpback whale songs, minky whale buzz sequences, sperm whale and dolphin echolocation clicks, and unidentified whale calls. Conversely, the deep-water area off the continental slope featured predominantly fish sounds, chorus-like sounds, and dolphin vocalizations. Detailed analyses of the most prevalent vocalizations are presented, including their bearing-time trajectories and localizations.

11:00

5aUW12. Evaluating the directivity of compact underwater acoustic recording devices. Robert T. Taylor (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, rtaylor119@utexas.edu), Megan Ballard, Colby W. Cushing (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Preston S. Wilson (Walker Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Kevin M. Lee, Andrew R. McNeese, Luis Acuna (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Julien Bonnel (Woods Hole Oceanographic Inst., Woods Hole, MA)

Commercially available underwater acoustic recorders have become commonplace tools in ocean-acoustics research, due to their ease of deployment, compact size, and relatively low cost. The size of these systems typically results in a configuration with the hydrophone in close proximity to the electronics housing, flotation devices, and other equipment that can degrade the generally assumed omnidirectional response of the hydrophone. The mid-frequency acoustic regime (0.5–10 kHz) is particularly affected due to the similarity between the length-scales of these structures and the corresponding acoustic wavelengths. Calibration measurements made at an open-water test facility with a calibrated source/receiver pair characterized the frequency-dependent receiver directivity for three underwater recording devices with different housings and hydrophones: the PVC air-filled Loggerhead Snap, PVC oil-filled SoundTrap ST300, and titanium air-filled SoundTrap ST600. Furthermore, directivity measurements of a TOSSIT mooring [Zitterbart *et al.*, HardwareX (2022)] were taken with the SoundTrap ST300 and SoundTrap ST600. Results suggest the frequency-dependent acoustic directivity of the recorders should not be neglected. In particular, the Loggerhead Snap had variations in receive level of over 20 dB as a function of receiver orientation angle and frequency, introducing a bias that obscures spectral levels of the *in situ* environment.

11:15

5aUW13. Optimizing parametric acoustic arrays for high-fidelity low-frequency signals. Ernst Uzhansky (Mech. Eng., Technion — Israel Inst. of Technol., Izhaq Greenboim str, Apt. 12 (Koren Family), Haifa 3498793, Israel, ernstuzhansky@gmail.com), Maya Friedlender, and Izhak Bucher (Mech. Eng., Technion — Israel Inst. of Technol., Haifa, Israel)

Parametric acoustic array (PAA) can leverage the nonlinear behavior of sound travelling through the nonlinear medium (e.g., air, water, human tissues) to generate low-frequency broadband high-directivity sound due to the self-demodulation of finite-amplitude modulated ultrasonic waves. However, while propagating, self-demodulated signals (hereon message) interact one with another, leading to message deformation and increase of the total harmonic distortion (THD). This makes it problematic to use powerful PAA to control complex low-frequency signals. This work explores a method for reducing THD by optimizing the input signal using various digital signal processing algorithms. The sustainability of the required spectral components with distance and the spatial boundaries of the operation of the proposed optimization algorithm are also investigated experimentally with a 520-channel hexagonal PAA and numerically. The simulations were done via k-wave acoustic toolbox considering nonlinearities and power law absorption.

Session 5pPA

Physical Acoustics: General Topics in Physical Acoustics

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Jacob C. Elliott, Cochair
The Penn. State Univer., Research West, State College, PA 16801

Contributed Papers

1:00

5pPA1. Dynamics of cylindrical crevice microbubbles. Eric Rokni (Phys., Rollins College, 511 W. Kenilworth Cir., Mequon, WI 53092, eric.rokni@gmail.com) and Julianna C. Simon (Graduate Program in Acoust., Penn State Univ., University Park, PA)

Microbubbles are theorized to form in crevices on mineralizations causing twinkling, a rapid color shift observed when imaging hard mineralizations with Doppler ultrasound. While the classic Rayleigh–Plesset equation and subsequent modifications describe bubble dynamics in a free field, they do not account for crevice boundaries. In this study, cylindrical boundary conditions were incorporated into the derivation of the Rayleigh–Plesset equation and validated experimentally. Cylindrical crevices with diameters ranging from 0.08–1.2 μm and depths of 1 μm were etched into a silicon wafer. Bubbles that formed in these crevices were photographed at high-speed through an inverted microscope while being driven with ultrasound at 0.75, 2.5, and 5 MHz. Experimentally, for all tested crevice sizes, the bubbles did not visibly grow. In contrast, computational results using standard water surface tension (72.5 mN/m) predicted the bubbles should grow $\sim$6 mm. However, when applying a higher surface tension that was measured on the silicon-water interface ($\sim$3000 mN/m), the bubbles were only predicted to grow $\sim$1 nm, supporting the experimental findings. These results offer insight into the mechanism causing twinkling and provides avenues for future investigation into the effect of crevice size and shape on twinkling. [Work supported by NSF CAREER 1943937 and GRFP DGE1255832.]

1:15

5pPA2. Cryoprotectant agent characterization via acoustical and optical analyses. Alicia Casacchia (Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, 204 E. Dean Keeton St., Austin, TX 78712-1591, acasacchia@utexas.edu), Matthew J. Uden (Dept. of Psychol., The Univ. of Texas at Austin, Austin, TX), Tanya Hutter, Preston S. Wilson, and Mark F. Hamilton (Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX)

In the field of cryopreservation, there exists a class of substances known as cryoprotectant agents (CPAs), which display an ability to vitrify. These CPAs are used to prevent cellular damage during the preservation of human tissues. In a vitrified state, cells exist in a glass-like state, meaning there is no formation of mechanically damaging ice crystals during cryogenic freezing and thawing. Although a CPA's ability to vitrify increases with concentration, its toxicity limits viable levels of use. This work seeks to find a correlation between acoustic properties of various CPAs and their concentrations via analyses of cavitation noise and cavitation jets, to accompany future studies on CPA toxicity. Four common CPAs, namely, dimethyl sulfoxide, ethylene glycol, propylene glycol, and formamide were investigated at various concentrations in aqueous mixtures and compared to pure water. The acoustic spectra of each CPA show an observable dependence on concentration levels and provide a potential way to probe the hydrogen bonding dynamics within the mixtures. [This is a SAWIAGOS project.]

1:30

5pPA3. Ultrasonic measurement of milk heat coagulation time (HCT). Agustin Harte (Dept. of Eng. Sci., Penn State Univ., 615 Meeks Ln., Port Matilda, PA 16870, afh5830@psu.edu), Lauren Katch, Federico Harte (The Penn State Univ., Port Matilda, PA), and Andrea P. Arguelles (Eng. Sci. and Mech., Penn State Univ., State College, PA)

The dairy industry relies on heat coagulation tests (HCTs) to determine high-quality milk heat stability for typical thermal processes. For the HCT, an oil bath heats 1–2 ml of milk to 140 °C. The test lasts 20–30 min and finishes with visual confirmation of milk coagulation. This time for coagulation is recorded as the HCT time. While standard in the industry, this test is time-consuming and prone to operator-dependent bias. Thus, an automated alternative for measuring HCT time is desirable. This work proposed the use of ultrasonic monitoring and employed 10 MHz contact and immersion ultrasonic transducers in combination with conventional pH and rheological sensors. Room-temperature skim milk was coagulated in standard containers with a Glucono-Delta-Lactone acid concentration of 3% while simultaneously being monitored by the ultrasound, rheological, and pH sensors, at testing rates of 1 s, ca. 6 s, and 2 min, respectively. The resulting ultrasonic wave speed measurements revealed an increasing trend throughout the coagulation process. Additionally, an inflection point within the increasing wave speed corresponded to rheological and pH parameters indicating coagulation. This work demonstrates the potential for ultrasonics to be used as sensors for coagulation within the dairy industry.

1:45

5pPA4. Measurements of nonlinear electric–acoustic interactions in liquids. Robert Lirette (Commun. Technol. Lab., National Inst. of Standards and Technol., 325 Broadway, MS67201, Boulder, CO 80305, robert.lirette@nist.gov), Tomasz Karpisz (Commun. Technol. Lab., National Inst. of Standards and Technol., Boulder, CO), Małgorzata Musiał, Jason A. Widegren (Mater. Measurement Lab., National Inst. of Standards and Technol., Boulder, CO), Aaron Hagerstrom, Nathan Orloff, and Angela C. Stelson (Commun. Technol. Lab., Natl. Inst. of Standards and Technol., Boulder, CO)

Electric and acoustic waves can interact nonlinearly inside a fluid medium. These interactions can create a heterodyned signal consisting of both sum and difference frequency products of the electric and acoustic frequencies. Here, we present direct measurements of nonlinear mixing products produced by interacting electric and acoustic waves inside of various fluids. The experiment consisted of a fluid channel that was in direct contact with an electrically conductive co-planar wave guide (CPW). An acoustic signal was applied to the channel via a transducer mounted above it, and simultaneously an electric signal was applied along the CPW from a vector network analyzer (VNA). The resulting signal was then measured with VNA set to frequency offset mode to capture the sum electric-acoustic mixing product. Results are presented for various organic solvents and salt solutions, each showing a different characteristic nonlinear response. This metrology allows for direct probing of non-linear effects in electrici–acoustic systems and

could lead to the development of new spectroscopic methods for characterizing chemicals and solutions.

2:00

5pPA5. Custom lens design for ultrasonic inspection of immersed anisotropic parts. Lauren Katch (Eng. Sci. and Mech., Penn State Univ., 212 Earth and Eng. Sci. Bldg., State College, PA 16802, Luk50@psu.edu) and Andrea P. Arguelles (Eng. Sci. and Mech., Penn State Univ., State College, PA)

Inspection of anisotropic parts using immersion ultrasonic waves poses resolution challenges caused by directionally dependent wave properties. Currently, focused lenses with spherical or cylindrical geometries are the standard method for creating convergent, small cross-section, and high amplitude beams within solids. These lenses form a conical beam that uniformly refracts at the fluid-solid interface of the immersed part when the solid is isotropic, producing the desired focusing behavior. However, for anisotropic solids, refraction becomes directionally dependent, which distorts the beam and causes non-uniform focal profiles and, in some cases, multiple foci. In this work, the forward wave propagation for an immersed anisotropic solid is modeled and exploited to develop custom lens geometries. These custom lenses utilize inverted slowness profiles to produce conical beams within the anisotropic solid. By incorporating these lenses, the focused beam behavior is improved and becomes more circular, higher in amplitude, and smaller in the cross-sectional area. These lenses have the potential to improve inspection resolution in anisotropic solids, an effort beneficial to the additive, aerospace, and electronics industries.

2:15

5pPA6. Loss factors of viscoelastic solid materials and the Poisson's ratio. Tamás Pritz (Budapest Univ. of Technol. and Economics, Műegyetem rkp. 3-9, Budapest 1111, Hungary, tampri@eik.bme.hu)

The dynamic properties of viscoelastic materials can effectively be characterized in the frequency domain by the complex moduli of elasticity and the complex Poisson's ratio. The ratio of imaginary to real part in these complex frequency functions is referred to as loss factor. In this paper, the relations between the loss factors of various complex moduli (shear, bulk, Young's, longitudinal) including the complex Poisson's ratio are investigated for homogeneous, isotropic, linear viscoelastic solids possessing positive Poisson's ratio. The novelty of the work is that the loss factor relations are expressed and investigated as a function of Poisson's ratio, ν, the magnitude of which is known for real solids, namely: $0 \leq \nu < 0.5$. It is shown that if $\nu = 0$ (e.g., cork), then all modulus loss factors are identical with the shear one, while the Poisson's loss factor is zero. In the general case when $0 < \nu < 0.5$ (e.g., hard plastics), the shear loss factor is the largest, and the others are smaller in the following order: Young's, longitudinal, bulk, and Poisson's loss factor. Finally, if $\nu \rightarrow 0.5$ (e.g., rubber), then the Young's loss factor approaches the shear one, while the others are usually much smaller than that.

2:30–2:50 Break

2:50

5pPA7. Acoustic signatures of helium abundance in hydrogen for planetary science. Rishabh Chaudhary (Mech. Eng., Tufts Univ., 200 College Ave., Medford, MA 02155, rishabh.chaudhary@tufts.edu), Robert D. White, Zarina Kosherbayeva (Mech. Eng., Tufts Univ., Medford, MA), Don Banfield, Anthony Colaprete (NASA Ames Res. Ctr., Mountain View, CA), Ian Neeson (VN Instruments, Elizabethtown, ON, Canada), Andrew Powell, and Andi Petculescu (Dept. of Phys., Univ. of Louisiana at Lafayette, Lafayette, LA)

The aim of this work is to model and measure the acoustic signatures of varying mixtures of helium, ortho-hydrogen, and para-hydrogen. In particular, we hope to demonstrate a sensor for helium abundance and hydrogen ortho:para ratio for planetary science missions to Saturn or Uranus. Gas composition as a function of depth is important for understanding atmospheric dynamics and planetary formation of the gas giants but is not well known. In the presence of molecular relaxation, sound speed and absorption in polyatomic gas mixtures become dispersive due to frequency-dependent heat capacities. This may potentially allow the identification of gas composition through relaxational acoustic spectroscopy. We compare theoretical predictions to measurements done in a pressure vessel up to a maximum pressure of 20 bar of relevance to gas giant atmospheres. Experiments for different gas mixtures at various pressures and temperatures are conducted using wide-bandwidth ultrasound between 20 kHz and 10 MHz. A technique is proposed to determine the ratio of hydrogen to helium using speed of sound and attenuation spectra. This approach may be further utilized to determine the ortho:para ratio. Challenges include providing sufficient bandwidth and adjusting for temperature, diffraction, and reflections. [Work supported from NASA Center Innovation Fund.]

3:05

5pPA8. Multi-sensor acoustic intensity analysis of the NG-19 Antares 230 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 Astron., Brigham Young Univ., Provo, UT)

In August 2023, the Antares 230 successfully launched for the NG-19 resupply mission to the International Space Station. Acoustical data were collected from various locations surrounding the launch pad from 60-200 m away from the vehicle. Each measurement location was configured such that acoustic intensity could be determined for a wide range of frequencies. Intensity calculations are made using the phase and amplitude gradient estimate (PAGE) method for information past the spatial Nyquist frequency of the station configuration. For the analyses presented here, the PAGE method utilized the generic MATLAB phase unwrapping function. This paper will show intensity vectors as a function of frequency and time to estimate the approximate location of the sound source. A least-squares optimization routine is used for the intersection point of each stations' intensity vector.

3:20

5pPA9. Nonlinear waveform steepening in time reversal focusing of airborne, one-dimensional sound waves, and the absence of Mach stems. Michael M. Hogg (Phys. & Astron., Brigham Young Univ., N283 ESC, Provo, UT 84602, mikerhogg@gmail.com), Brian D. Patchett (Phys., Utah Valley Univ., Orem, UT), and Brian E. Anderson (Phys. & Astron., Brigham Young Univ., Provo, UT)

Time reversal (TR) is a process that can be used to generate high amplitude focusing of sound. Previous research has shown that TR in reverberant environments can generate peak levels that exceed 200 dB with airborne, audible sound [Patchett and Anderson, J. Acoust. Soc. Am. 151(6), (2022)], and 134 dB with airborne ultrasound [Wallace and Anderson, J. Acoust. Soc. Am. 150(2), (2021)]. Particularly, in the focusing of audible sound, these high amplitude focused waveforms exhibit multiple nonlinear properties including waveform steepening and a nonlinear increase in compressions due to the generation of free space Mach stems. This study attempts to remove the Mach stems by generating the focus in a 1D system, since Mach stems require higher dimensional spaces to form. The system of pipes used ensures a planar 1D reverberant environment. Results show that waveform steepening remains as expected but that the nonlinear increase in compression amplitudes disappears without the ability for Mach stems to form. This provides further confirmation that Mach stems are the cause of the nonlinear increase in compression amplitudes observed with high amplitude TR.

3:35

5pPA10. Characterization and impact of dissolved gases in a high-frequency flow reactor. Yihao Huang (Eng. Sci., Univ. of Oxford, Dept. of Eng. Sci., University of Oxford, Parks Rd., Oxford OX13PJ, United Kingdom, yihao.huang@exeter.ox.ac.uk), Pankaj Sinhmar (Eng. Sci., Univ. of Oxford, Oxford, United Kingdom), Tristan Nerson (LAUM, Le Mans Université, Le Mans, France), Cherie Wong, Lillian N. Usadi (Eng. Sci., Univ. of Oxford, Oxford, United Kingdom), Jason L. Raymond (Fralin Biomedical Res. Inst. at VTC, Oxford, United Kingdom), and James Kwan (Eng. Sci., Univ. of Oxford, Oxford, United Kingdom)

Addressing the growing demands for sustainable green technology in chemical processes, we aim to develop a novel high-frequency ultrasonic flow reactor that utilizes converting ultrasound waves, resulting in intense

localized acoustic pressure fields. Our hypothesis that our design of converging waves will establish a uniform spatial distribution of cavitation activity with high energy density, thereby augmenting the rate of radical formation and significantly enhancing overall reactor performance. To validate the hypothesis, KI and terephthalic acid dosimetry were performed. A comprehensive calorimetric study was employed for acoustic power, yielding valuable insight into the energy dynamics of the system. Furthermore, our research systematically assesses the impact of dissolved gases and varying flow rates on reactor performance. Notably, we observe that dissolved gases in the reaction medium exert a substantial influence on reactor efficiency. Additionally, an increased flow rate beyond the optimum decelerates hydroxyl radical generation due to a lower residence time. The detailed characterization, facilitated by dosimetry techniques, unveils the potential of the sonochemical reactor for industrial applications.

3:50

5pPA11. Sono-catalytic syngas production by low-temperature biomass gasification. Cherie Wong (Eng. Sci., Univ. of Oxford, Begbroke Directorate, Begbroke Hill, Woodstock Rd., Oxford OX5 1PF, United Kingdom, cherie.wong@eng.ox.ac.uk) and James Kwan (Eng. Sci., Univ. of Oxford, Oxford, United Kingdom)

Syngas holds a crucial role in diverse industrial applications, including chemical synthesis and power generation. It has been traditionally generated through high-temperature processes, such as steam reforming or biomass gasification, demanding temperatures of up to 1200 °C, which results in substantial energy consumption. Consequently, there is a growing interest in exploring alternative, low-temperature biomass gasification processes. This paper introduces an innovative method for syngas production through sono-catalytic biomass conversion at significantly lower temperatures. The approach employs a sonochemical reactor that concentrates acoustic waves to activate inertial cavitation. Furfural, a biomass-derived platform chemical, was chosen as the primary chemical for the study, showcasing the potential of sonochemical syngas production. The reaction converted furfural into various gas products, particularly a mixture of hydrogen and carbon monoxide, with the chemical reaction hypothesized to be triggered by pyrolysis during bubble collapse hot spots. The incorporation of heterogeneous metal foams into the system further enhanced furfural conversion rates. This improvement was attributed to the preferential nucleation of cavitation bubbles on the porous metal foam surface, bringing the hot spots into proximity with furfural molecules. This application of sonochemistry in furfural pyrolysis at low temperatures holds promise for a more economical approach to biomass valorization.

4:05

5pPA12. Torsional pendulum driven by sound-matter orbital angular momentum transfer involving acoustic vortices. Elena Annenkova (Laboratoire Ondes et Matière d'Aquitaine, Univ. of Bordeaux, 351 Cours de la Libération, Talence 33405, France, a-a-annenkova@yandex.ru), Benjamin Sanchez-Padilla, and Etienne Brasselet (Laboratoire Ondes et Matière d'Aquitaine, Univ. of Bordeaux, Talence, France)

The transfer of the orbital angular momentum from sound to matter using acoustic vortex beams has been experimentally revealed only recently. The physics of such a process may involve two distinct mechanisms that can add one to another: a dissipative one driven by sound absorbing targets and a nondissipative one driven by angular momentum conversion as a result of a scattering process. In both cases, the mechanical consequence is the appearance of an acoustic radiation torque exerted on the irradiated body. Here, following recent developments of torsional pendulum driven by sound-matter orbital angular momentum transfer, we report on our recent advances, which are two-fold. First, we report on drastic improvement by more than two orders of magnitude of the quality factor of the mechanical resonance, reaching values >1000. Second, we extend the detection and measurement of acoustic radiation torque from a scalar framework to situations where the vectorial nature of the angular momentum matters. These results contribute to the development of sensitive wave-matter devices for rotational metrology applications encompassing material and wave aspects. [Work received funding from the European Union's Horizon 2020 research and innovation programme under the Marie Sklodowska-Curie grant agreement No. 101027737.]

4:20

5pPA13. Effects of increasing orbital number on the field transformation in focused vortex beams. Chirag A. Gokani (Appl. Res. Labs. and Walker Dept. of Mech. Eng., Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, cgokani@arlut.utexas.edu), Michael R. Haberman, and Mark F. Hamilton (Appl. Res. Labs. and Walker Dept. of Mech. Eng., Univ. of Texas at Austin, Austin, TX)

Acoustic vortex beams possess helical wavefronts characterized by an orbital number n. In the presence of focusing, the toroidal vortex ring for $n = 1$ moves out of the focal plane $z = d$ and toward the source with increasing n, accompanied by a departure from its toroidal shape. In this talk, ray theory is developed to explain this field transformation. An expression is derived for the radius and cross-sectional area of the annular channel formed by the family of rays emanating from a thin circular ring centered at the origin in the source plane $z = 0$. This expression leads to explicit results for the field amplitude and caustic coordinates due to a focused vortex source with an arbitrary axisymmetric amplitude distribution. Comparison with field calculations in which diffraction is included demonstrate that the caustics describe the redistribution of the global maximum and its shift toward the source plane with increasing n. For moderate focusing gains (of order 10 or 20), a toroidal vortex ring for $n = 1$ transforms with increasing n into a spheroidal surface in the prefocal region $0 < z < d$ having volume $n\lambda d^2/6$, where λ is the wavelength, inside of which exists a shadow zone. [C.A.G. was supported by the ARL:UT McKinney Fellowship in Acoustics.]

Session 5pUW

Underwater Acoustics: Underwater Sources and Receivers

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Contributed Papers

1:00

5pUW1. Structure of interferogram and hologram of non-moving source in presence of shallow water internal waves. Mohsen Badiey (Elec. and Comput. Eng., Univ. of Delaware, 139 The Green, Rm. 140, Evans Hall Rm. 140, Newark, DE 19716, badiey@udel.edu), Sergey A. Pereselkov (Math. Phys. and Inform. Technol., Voronezh State Univ., Voronezh, Russian Federation), and Venedikt Kuz'kin (Sci. Ctr. for Wave Res., General Phys. Inst., Moscow, Russian Federation)

The results of holographic signal processing of the oceanographic experiment SWARM-95 on the coast of New Jersey are presented in the paper. The used acoustic signals were obtained on stationary acoustic track (non-moving airgun source) in the presence of shallow water intensive internal waves (IIW). The IIW led to significant 3D acoustic effects [Badiey et al., JASA 117(2), 613–625]. Within framework of holographic signal processing [Pereselkov and Kuz'kin, JASA 151(2), 666–676] the source interferogram (sound intensity distributions in frequency-time coordinates) is analyzed by two-dimensional Fourier transform (2D-FT). The result of the 2D-FT is called the Fourier hologram (source hologram). In result of holographic signal processing, the two separated sets of spectral spots are obtained in hologram domain. The first set of spectral spots corresponds to sound field in nonperturbed waveguide (without of IIW). The second set of spectral spots in hologram corresponds to hydrodynamic field perturbation by IIW. This structure of the hologram allows the reconstruction of interferogram of nonperturbed field for waveguide in the absence of IIW. The error in the reconstruction of the nonperturbed interferogram is estimated. [Work supported by grant from the Russian Science Foundation 23-61-10024.]

1:15

5pUW2. Quiet uncrewed surface vessels assess fish avoidance to motorized survey ships with varying noise levels in the Great Lakes. Thomas M. Evans (DNRE, Cornell, Ithaca, NY), Lars G. Rudstam (Natural Resources and the Environment, Cornell Univ., Bridgeport, NY), Suresh A. Sethi (Earth and Environ. Sci., Brooklyn College, Brooklyn, NY), Daniel L. Yule (USGS, Ashland, WI), David M. Warner (USGS, Ann Arbor, MI), S. D. Hanson (USFWS, New Franken, WI), Benjamin Turschak (Michigan Dept. of Natural Resources, Charlevoix, MI), Mark R. Dufour, Steven A. Farha (USGS, Ann Arbor, MI), Andrew Barnard (Acoust., Appl. Sci. Bldg., Penn State, 201C, University Park, PA 16802, barnard@psu.edu), Steven Senczyszyn (Mech. Engineering-Eng. Mech., Michigan Technol. Univ., Houghton, MI), Susan E. Wells (USFWS, Ithaca, NY), Scott R. Koproski (USFWS, Alpena, MI), Timothy P. O'Brien (USGS, Ann Arbor, MI), Kevin N. McDonnell (USFWS, Alpena, MI), Hannah Blair (DNRE, Cornell, Southampton, NY), James M. Watkins (DNRE, Cornell, Ithaca, NY), Patricia M. Dieter (USGS, Ann Arbor, MI), James J. Roberts (USGS, Sandusky, OH), and Peter Esselman (USGS, Ann Arbor, MI)

Acoustic surveys are a foundational component of many fisheries monitoring programs because they allow assessment of spatially extensive stocks. They are widely used to evaluate prey fish throughout the Great Lakes by numerous coordinating vessels. Traditionally, these surveys have been conducted by crewed and motorized vessels, but fish avoidance of motorized platforms has been reported in multiple studies and may bias survey estimates. Quiet uncrewed platforms are becoming increasingly available and offer the opportunity to explore bias in traditional surveys. Several identical quiet uncrewed surface vehicles (USVs) operated by Saildrone were equipped with 120 kHz Simrad EK80 transducers and deployed in Lakes Erie, Huron, Michigan, and Superior in the summers of 2021, 2022, and 2023. The USVs were overtaken by numerous motorized vessels over 2 km transects using transducers with the same frequency. Fishes showed a limited response to approaching vessels, and acoustic measures were similar between the USVs and motorized vessels. Therefore, acoustic surveys in the Great Lakes appear unbiased and are widely comparable. Findings from this work will inform interpretation of acoustic data in the Great Lakes and provide the largest scale testing of fish avoidance during acoustic surveys to date.

1:30

5pUW3. Characterization, understanding, and mitigation of underwater noise radiated by ships in the St. Lawrence Estuary. Pierre Cauchy (Institut des Sci. de la mer (ISMER), Université du Québec à Rimouski (UQAR), 310 allée des Ursulines, Rimouski, QC G5L 3A1, Canada, pierre_cauchy@uqar.ca), Pierre Mercure-Boissonnault, Cécile Perrier de la Bathie, Faniry Rabetoandro (Institut des Sci. de la mer (ISMER), Université du Québec à Rimouski (UQAR), Rimouski, QC, Canada), Cédric Gervaise (Chorus, Grenoble, France), Sylvain Lafrance (Innovation Maritime, Rimouski, QC, Canada), and Guillaume St-Onge (Institut des Sci. de la mer (ISMER), Université du Québec à Rimouski (UQAR), Rimouski, QC, Canada)

Underwater noise generated by commercial traffic is the main source of anthropogenic noise pollution at low frequencies, increasingly present at a global scale and of critical interest in the St. Lawrence Estuary (eastern Canada), where a rich biodiversity meets the shipping corridor linking the Great Lakes to the Atlantic Ocean. The Marine Acoustic Research Station (MARS, www.projet-mars.ca/en) is an applied research project dedicated to characterizing, understanding, and mitigating underwater traffic noise, contributing to the global effort of improving cohabitation between human activities and marine life. A cutting edge marine acoustic observatory has been specifically designed to collect high-quality measurements of the underwater noise radiated by ships. It is deployed yearly since 2021, operating during the ice-free season. A database of over 2000 measurements representative of the underwater noise radiated by the fleet operating in the St Lawrence Estuary has been collected, with a demonstrated repeatability of less than 1.5 dB which confirms the ability of the observatory to effectively assess the efficiency of noise reduction measures. The MARS database supports the development of a noise prediction model and provides feedback to shipowners and relevant information regarding the St. Lawrence fleet to the government for future underwater vessel noise reduction targets.

1:45

5pUW4. Dynamic scheduling for the underwater acoustic localization of multiple moving targets. Youngchol Choi (KRISO, 32 1312 beon-gil, Yuseong-daero, Yuseong-gu, Daejeon 34103, Republic of Korea, ycchoi@kriso.re.kr) and Sea-Moon Kim (Ocean System Eng. Res. Div., Korea Res. Inst. of Ships and Ocean Eng., Daejeon, Republic of Korea)

We propose a novel method for the underwater acoustic localization of multiple moving transponders with a single ping. We make the transceiver receive the response signals from transponders without packet collisions by scheduling the packet transmission time of each transponder. To this end, the transceiver computes a doubly conservative round trip time (RTT) for each transponder based on the latest RTT, packet durations, maximum relative speed between the transceiver and the transponder, and the transceiver assigns a waiting time to each transponder such that the earliest packet arrival time of the next transponder should be greater than the latest packet reception completion time of the previous transponder. This dynamic scheduling method improves the update rate of the underwater acoustic localization system for multiple moving targets by ensuring the reception of packets of transponders in a packet train manner. In addition, even in scenarios where positioning and communication occur simultaneously, the proposed method guarantees robust and stable update rate as response signals from multiple moving transponders never collide at the transceiver, regardless of packet length variations due to changes in communication payload. [Work supported by KIMST funded by the Agency of Korea Coast Guard (KIMST-20210547).]

2:00

5pUW5. The underwater wind-generated noise characteristics and wind speed prediction of typhoon "Ma-on" by single vector hydrophone. Xiaoming Cui (Southern Marine Sci. and Eng. Guangdong Lab. (Guangzhou), 1119 Haibin Rd., Nansha District, Guangzhou 511458, China, nwpucxm@163.com), Huayong Yang (Southern Marine Sci. and Eng. Guangdong Lab. (Guangzhou), Guangzhou, China), Qing Hu (School of Ocean Eng. and Technol., Sun Yat-sen Univ., Zhuhai, China), Chao Li, and Siyuan Cang (Southern Marine Sci. and Eng. Guangdong Lab. (Guangzhou), Guangzhou, Guangdong, China)

Underwater ambient noise is greatly affected by sea surface wind speed, especially during typhoons when the wind speed over the sea surface rapidly increases within a short period of time. The relationship between noise and sea surface wind speed can be used to predict typhoon intensity. Based on the normal mode theory as well as the wind-generated sea noise propagation model and the wind-generated noise source level model, this paper uses the underwater noise vector sound field data measured by a single vector hydrophone during the passage of typhoon "Ma-on" to examine the relationship between underwater ambient noise intensity and the changes in wind speed and frequency during typhoon conditions. It also calculates the sea surface wind speed during a typhoon through inversion. The inversion result is essentially consistent with the forecasted typhoon wind speed provided by the meteorological observatory.

2:15

5pUW6. Derivations of transfer functions for estimating ship underwater radiated noise from onboard vibrations. Esen Cintosun (DRDC Atlantic, 9 Grove St., Dartmouth, NS B3A 3C5, Canada, Esen.Cintosun@ecn.forces.gc.ca)

The adverse impacts of underwater radiated noise (URN) from marine vessels on marine life are increasingly recognized. URN estimation and subsequent monitoring could be used to track URN and take measures to reduce it in sensitive environments (e.g., a marine life protected area). Two common analytical radiated noise power approximations and two empirical methods are assessed for URN estimation. The assessments were carried out by testing the methods on data from acoustic range measurements of an ORCA-class training vessel of the Royal Canadian Navy, named Patrol Craft Training (PCT) Moose and a DAMEN Combi Freighter 3850, named Heinz G. The analytical approximations are called equivalent radiated power and power from volume velocity; both are used to estimate URN directly from the onboard ship vibration measurements. The empirical methods are based on the correlations of onboard vibrations with measured URN, from which the derived transfer functions. The analytical and empirical transfer functions are compared. A statistical energy analysis model of PCT Moose is also used to estimate onboard vibrations from the ship's engine, generator, and propeller specifications for URN estimation at the design stage.

2:30

5pUW7. Bistatic spatial coherence for micronavigation of a volumetric synthetic aperture sonar. Kyle S. Dalton (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, ksd5377@psu.edu), Thomas E. Blanford (Univ. of New Hampshire, Durham, NH), and Daniel C. Brown (Graduate Program in Acoust., Penn State Univ., State College, PA)

Combining multiple passes of a volumetric synthetic aperture sonar (SAS) may open new possibilities for sonar imaging, if the passes can be precisely aligned. Previous work suggests that coherence-based micronavigation is a potential approach to achieve this alignment. However, existing coherence-based techniques require assumptions that break down in many volumetric imaging geometries. This breakdown leads to a loss of coherence that has not been widely studied, and consequently, has hindered the application of micronavigation to volumetric SAS systems. This work will explore the impact of sensing range and bistatic separation on spatial coherence in downward-looking, volumetric sonar applications where the far-field assumption no longer holds. As signals transmitted from downward-looking sonars may penetrate into the sediment floor, this analysis will consider both interface and volumetric scattering. First, using both simulated and experimental sonar data, we will quantify the loss in coherence that occurs in these near-field geometries. Then, methods to regain lost coherence will be examined in an effort to enable spatial coherence-based navigation for a volumetric SAS. Finally, the work's relevancy for ongoing efforts in acoustic navigation and repeat-pass imaging will be discussed.

2:45

5pUW8. Directional soundscapes in the Norwegian Seas observed with a coherent hydrophone array. Arpita Ghosh (Northeastern Univ., 360 Huntington Ave., Boston, MA 02115, ghosh.arp@northeastern.edu), Sai Geetha Seri (Northeastern Univ., Boston, MA), Hamed Mohebbi-Kalkhoran (Mech. Eng., Massachusetts Inst. of Technol., Cambridge, MA), Olav Rune Godø (Inst. of Marine Res., Bergen, Norway), Heidi Ahonen (Norwegian Polar Inst., Tromsø, Norway), Nicholas C. Makris (Mech. Eng., Massachusetts Inst. of Technol., Cambridge, MA), and Purnima R. Makris (Northeastern Univ., Boston, MA)

Directional soundscaping is an efficient approach for examining marine ecosystems since it allows the study of living organisms, their behavior, and temporo-spatial interactions with other natural and man-made objects underwater, useful for ecosystem monitoring during offshore energy activities, maritime surveillance, and defense. A large aperture coherent hydrophone array was employed for remote sensing in the Norwegian and Barents Seas in Spring 2014. Extensive analysis have been conducted in post-processing of recorded hydrophone array data for automatic detection, bearing estimation, and classification of signals by different sound producers, such as whale calls, ship radiated noise, and fish sounds. Directional soundscaping through passive ocean acoustic waveguide remote sensing (POAWRS) provides bearing-time trajectories of signal detections that can be applied for geographical mapping. The received marine mammal sounds include vocalizations from baleen whales such as humpback, fin, and minke, and toothed whales, such as beluga, pilot, and sperm. We provide an insight to the vocalizations and behaviorial patterns of these whales in the Lofoten and Northern Finmark regions. The relative contributions of distinct sound sources, including whales, fish, and ships, to the directional soundscapes are quantified. We also examine their temporo-spatial dependences via geographic mapping of acoustic signals from distinct sound producers.

3:00–3:15 Break

3:15

5pUW9. Deep-water ambient sound on near-bottom receivers in the New England Seamount Chain. Oleg A. Godin (Phys. Dept., Naval Postgrad. School, Monterey, CA 93943, oagodin@nps.edu), Matthew W. Walters (Phys. Dept., Naval Postgrad. School, Monterey, CA), Tsu Wei Tan (Marine Sci., ROC Naval Acad., Kaohsiung, Taiwan), and John E. Joseph (Oceanogr. Dept., Naval Postgrad. School, Monterey, CA)

Ambient sound was recorded for about two months by three synchronized, single-hydrophone receivers that were moored at depths of 2573, 2994, and 4443 m on seamount flanks. Hydrophones were located within 5 m from the seafloor. The data reveal the power spectra and intermittency of the ambient sound intensity in a 13-octave frequency band from 0.5 to 4000 Hz. Statistical distribution of pressure amplitude exhibits much heavier tails than the expected Rayleigh distribution throughout the frequency band of observations. Spatial variability of the observed ambient sound is controlled by the seafloor properties, bathymetric shadowing, and nonuniform distribution of the noise sources on the sea surface due to the Gulf Stream and its meanders. Interferometry of the ambient sound recorded by the receivers with the horizontal separations of 7.0 km shows strong variations of the acoustic travel time between the receivers along a surface-reflected path due to the evolution of the Gulf Stream position. The magnitude of the variations of the passively measured acoustic travel time is consistent with the available contact measurements of the sound speed profiles. Environmental inferences derived from the ambient, shipping, and flow noise data will be discussed. [Work supported by the ONR TFO DRI.]

3:30

5pUW10. Evaluation of the performance of technologies for reducing ships' machinery noise using a small-scale ship-like structure in a Water Basin. Marc-André Guy (Université de Sherbrooke, 4682, rue Mary-Cormier, Québec, QC G1Y3M8, Canada, guym3301@usherbrooke.ca), Kamal Kesour (Innovation Maritime, Rimouski, QC, Canada), Olivier Robin (Université de Sherbrooke, Sherbrooke, QC, Canada), Julien St-Jacques (Innovation Maritime, Rimouski, QC, Canada), and Mathis Vulliez (Université de Sherbrooke, Sherbrooke, QC, Canada)

Ocean ambient levels have increased in the last decades, especially in the low-frequency domain (under 500 Hz). This increase is partly due to underwater radiated noise (URN) from commercial ships. Excessive URN harms marine life and is, therefore, considered a pollution that needs to be reduced. At low speeds, machinery is the primary noise source on ships. Mitigation technologies exist to limit machinery's contribution to URN. While implementing these technologies is costly, a lack of quantitative data regarding their exact performances usually results in limited concrete ship applications since the cost-to-benefit ratio is imprecise. This study aims to quantify better the performance of standard noise mitigation technologies using a small-scale ship-like structure in a water basin. The basin's acoustic field is first characterized with and without the structure. The structure is then equipped with different mitigation technologies. A loudspeaker and a vibration shaker are fed with pink noise or measured signals on actual machinery. Hydrophones, microphones, accelerometers, and force sensors measure the response in the basin and on the structure. The performance of each tested technology is evaluated and ranked in terms of URN reduction. The relative contributions of airborne and structure-borne transmission paths on URN are also examined.

3:45

5pUW11. Comparative field trials of distributed acoustic sensing using helically wound fibers versus straight fibers in cables deployed on lakebed of the Xinfengjiang Reservoir. Chao Li (Southern Marine Sci. and Eng., Guangdong Lab., No. 1119, Haibin Rd., Nansha District, Guangzhou, Guangdong 511458, China, lichao@gmlab.ac.cn), Kan Gao (Shanghai Inst. of Optics and Fine Mech., Chinese Acad. of Sci., Shanghai, China), Haocai Huang (Ocean College, Zhejiang Univ., Zhoushan, Zhejiang, China), Siyuan Cang, Xiaoming Cui, and Huayong Yang (Southern Marine Sci. and Eng., Guangdong Lab., Guangzhou, China)

The application of distributed acoustic sensing (DAS) in underwater acoustics and marine geophysics has faced challenges due to axial and relatively low sensitivity issues associated with ordinary fibers. Helically wound fibers (HWFs), employing an elastic plastic mandrel, represent a primary approach to overcoming the sensitivity shortcomings of straight fibers. This study aims to qualitatively validate the theoretically predicted enhancements and deterioration offered by two types of HWF compared to straight submarine optical cables in underwater applications using one DAS host. Our findings confirm a significant enhancement in the broadband sensitivity of HWF for earthquake, ship noise, and active sources, reaching a maximum improvement of approximately 20 dB at 1000 Hz. In our field data, we observe that all types of cables captured signals ranging from several to 1700 Hz, encompassing the crucial frequency band for underwater acoustic and seismological studies. However, the use of HWF introduced higher background noise in the some frequency bands. The HWF cable, with a 2-cm diameter, is easily manufacturable for kilometer range, enhancing DAS sensitivity comparable to hydrophones. Consequently, HWF emerges as a promising and cost-effective solution for achieving high sensitivity in acoustic and seismic monitoring.

4:00

5pUW12. Efficiency of a cluster of underwater low-frequency sources. Andrey K. Morozov (Marine Syst., Teledyne, 49 Edgerton Dr., North Falmouth, MA 02556, amorozov@teledyne.com)

The efficiency of underwater low-frequency sound sources can be improved by using tunable high-quality resonators, increasing the emission aperture, or using source clusters. A very low frequency tunable resonator is hard to build. Large aperture sources are difficult to deploy. This study shows that the efficiency of a cluster of sources can be much higher than that of each source. Low frequency sources separated by distance operate independently, the sources add up the pressure, and the emitted pressure is proportional to the number of elements in the cluster. As a result, the radiated power of the cluster increases according to the quadratic law of the number of its elements, and the efficiency of the cluster increases in proportion to this number. To prove this, finite element modeling of an array of underwater sources was carried out. The simulation included the internal structure of each sound source with Helmholtz bubble resonators driven by standard audio 2 kW subwoofers. Indeed, the cluster efficiency increases when the distance between the sources exceeds several of their diameters. Simulations showed that a cluster of 32 broadband sources with a frequency of 10–100 Hz could produce sound pressure levels equal to a standard air gun.

4:15

5pUW13. The use of Navy range bottom-mounted, bi-directional transducers for long-term, deep-ocean prey mapping. Alexander Muniz (NUWC, 1178 Howell St., Newport, RI 02841, alexander.p.muniz.civ@us.navy.mil), Stephanie Watwood, Ronald Morrissey (NUWC, Newport, RI), and Kelly Benoit-Bird (Monterey Bay Aquarium Res. Inst. (MBARI), Moss Landing, CA)

Beaked whales have been shown to be particularly sensitive to mid-frequency active sonar but continue to spend considerable time on Navy training ranges, where exposure to sonar is frequent. Understanding the underlying movements of their prey could help to explain the distribution and abundance patterns seen for beaked whales. Beaked whales are among the deepest diving marine mammals and have been shown to feed near the ocean bottom, at depths up to 3000 m. Typical surface-deployed prey mapping technology is not suitable for probing these deep foraging depths. This pilot study aims to us the bi-directional nodes on Navy hydrophone ranges that were developed for underwater communications as a rudimentary prey mapping system. A 1 ms 2 kHz signal was broadcast on an 8 kHz carrier signal and resulting returns were processed to estimate depths of the reflectors. Data are presented from three different time periods on the Southern California Anti-Submarine Warfare Range.

4:30

5pUW14. An overview of the calibration process for a digital hydrophone. Nicolas Wood (Ocean Sonics, 110 Parkway Dr., Truro Heights, NS B6L1N8, Canada, nicolas.wood@oceansonics.com)

Digital hydrophones are instruments for measuring underwater sound that incorporate the entire data chain into a single unit. Calibration of analog and digital hydrophones is largely similar but differs in measurement signal type. Digital hydrophone calibration is not currently captured in the standards outlining hydrophone calibration (such as IEC 60565), so they currently cannot be calibrated to a recognized standard. The testing setup and calibration methods are the same as for an analog hydrophone but without needing external digitization or analog signal conditioning. Potential issues regarding digital hydrophone calibration: careful calculation of signal amplitudes is required at upper frequency band; the lack of analog output requires different units of measure or information on counts/volt conversion. The first point here is also a current issue for analog hydrophones where the digitization and the processing of data during the calibration are not prescribed. Ideally, both of these issues would be addressed in future standards updates, which would improve the operation of both analog and digital hydrophones. This paper will expand on the issues highlighted here involving digital hydrophone calibration and offer recommendations for best practice. It will also provide an outline of the current calibration procedure as used at Ocean Sonics.

4:45

5pUW15. The ONR three octave research array at Penn State. Chad M. Smith (Appl. Res. Lab., The Penn State Univ., State College, PA 16804, chad.smith@psu.edu), Luigi Troiano (KFM Technol. Resources Inc., La Spezia, Italy), Milan Markovic, and Marco Bernardini (Ctr. for Maritime Res. and Experimentation, La Spezia, Italy)

A new, large diameter towed research array has been built to support the experimental efforts of the Office of Naval Research (ONR) Ocean Acoustics Program. Array development was a collaborative effort between the Centre for Maritime Research and Experimentation (CMRE) in La Spezia, Italy, and the Penn State Applied Research Laboratory (PSU). PSU chose the array specifications based on discussions within the underwater acoustics community, prior measurement experience, cost efficiency, and flexibility in scientific measurements, while CMRE designed and assembled the array. The design includes three, forward-nested acoustic apertures cut for 1, 2, and 4 kHz, and hence, has been dubbed the Three Octave Research Array (THORA) in deference to its predecessor, the Five Octave Research Array (FORA). However, this system has a modular design to allow addition of acoustic apertures if scientific interest and sponsor support warrant. The array currently consists of a 50 m acoustic module and a 25 m vibration isolation module to help isolate the acoustic module from cable strum. This ship towed system has nearly 1 km of tow cable and a maximum measurement depth of 500 m. The array's design, its use in the ONR NESMA23 Pilot experiment, and data quality will be discussed.

5:00

5pUW16. Simultaneous in-air versus in-water measurements of humpback whale breathing sounds. Max Radermacher (Northeastern Univ., 360 Huntington Ave. Boston, MA 02115, radermacher.m@northeastern.edu), Matthew E. Schinault, Sai Geetha Seri, and Purnima Ratilal (Northeastern Univ., Boston, MA)

Humpback whales produce an assortment of sounds ranging from moans and songs to surface-breaching, breathing, and foreflipper flaps. During a sea trial of Northeastern University's coherent hydrophone array at the Great South Channel in US Northeastern coastal waters, the HF subaperture consisting of 32 hydrophone elements at 0.375 m spacings was deployed vertically with half of the hydrophones in air and the other half submerged underwater on September 6th, 2021. Many instances of humpback whale breathing sounds were recorded over several hours of observation. Visual sightings and video recordings were used to coregister sounds recorded on individual hydrophones of the HF subaperture. Here, we examine and compare the simultaneously recorded humpback whale breathing sound spectra, bandwidth, and duration for in-air versus underwater hydrophones. Whale distances to the vertical HF subaperture could be calculated from curved time of arrival differences due to close proximity of the array. These distances are applied to correct the received underwater sound pressure level for transmission loss and estimate underwater recorded whale breathing sound source level. A consistent broadband dip in the measured underwater sound spectra with null centered at 15 kHz is investigated by propagation modelling, considering both modal interference and attenuation from loud bubbles.

5:15

5pUW17. Optimizing hydrophone-preamplifier systems: Theoretical design, computational analysis, and experimental validation for towed array applications. Matthew E. Schinault (Northeastern Univ., 360 Huntington Ave., 409 Dana, Boston, MA 02115, schinault.m@northeastern.edu), Max Radermacher, Jesse Segel, and Purnima Ratilal (Northeastern Univ., Boston, MA)

Hydrophone response parameters greatly influence the design of pre-amplifier and analog to digital conversion (ADC) components. Here, we show a comprehensive methodology for the modeling of hydrophone response, focusing on the theoretical, computational, and experimental approach for low and high frequency SONAR applications. Theoretical calculation of hydrophone response based on electromechanical properties is compared with Finite Element Analysis (FEA) and experimental measurement. An electrical equivalent hydrophone simulator Butterworth-Van Dyke (BVD) circuit is constructed based on admittance and susceptance measurements to represent the transducer's electrical behavior for testing pre-amplifier and ADC response without the presence of environmental or hydrophone noise. This model facilitates a systematic approach to towed array front-end design with hardware in the loop to update models to improve performance prediction. Measurements of the constructed transducer system validate the theoretical and computational models, providing insights into real-world performance.

5:30

5pUW18. Great Lakes fisheries vessels and uncrewed surface vessels: A publicly available dataset. Steven Senczyszyn (Great Lakes Res. Ctr., Michigan Technol. Univ., 1400 Townsend Dr., Houghton, MI 49931, sasenczy@mtu.edu), Andrew Barnard (Acoust., Penn State, University Park, PA), Erik Kocher (Great Lakes Res. Ctr., Michigan Technol. Univ., Houghton, MI), Thomas M. Evans (Natural Resources and the Environment, Cornell Univ., Ithaca, NY), Lars G. Rudstam (Natural Resources and the Environment, Cornell Univ., Bridgeport, NY), Suresh A. Sethi (Natural Resources and the Environment, Cornell Univ., Ithaca, NY), Daniel L. Yule (Great Lakes Sci. Ctr., U.S. Geological Survey, Ashland, WI), David M. Warner, Steven A. Farha, Mark R. Dufour, Timothy P. O'Brien, Patricia M. Dieter (Great Lakes Sci. Ctr., U.S. Geological Survey, Ann Arbor, MI), James J. Roberts (Great Lakes Sci. Ctr., U.S. Geological Survey, Sandusky, OH), James M. Watkins (Natural Resources and the Environment, Cornell Univ., Ithaca, NY), and Peter Esselman (Great Lakes Sci. Ctr., U.S. Geological Survey, Ann Arbor, MI)

For decades, fish abundance surveys have been performed by large, crewed fisheries vessels using echosounders or trawling. However, the estimates obtained by these traditional methods may be biased due to the propagated noise generated by these vessels. A three-year collaborative effort has been conducted to measure the radiated acoustic signature of 17 fisheries vessels operating across all five of the Great Lakes under several operating conditions. This was done using a mobile ship noise measurement system developed specifically for this purpose. Additionally, with the recent advent of uncrewed surface vessels (USVs, "Saildrone"), an alternative "quiet" method of performing these abundance surveys is now possible. A detailed acoustic characterization was conducted for a USV to provide a comparison between traditional "loud" fisheries vessels and these "quiet" surface vessels. This talk will detail the acoustic signatures of all 18 vessels, compare the "loud" and "quiet" vessels, and provide a detailed description of the vessel measurement profiles which are now available as a public dataset. This dataset includes the time domain acoustic measurements, sound speed profiles, and GPS tracks of the measured vessels.