Session 1aAO

Acoustical Oceanography and Underwater Acoustics: Acoustics in Naturally Constrained Environments: Estuaries, Bays, Inlets, Fjords, and Rivers I

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Chair's Introduction—9:30

Invited Papers

9:35

1aAO1. Broadband acoustic wave propagation and scattering in Delaware Bay Estuary. Mohsen Badiey (Univ. of Delaware, Newark, DE 19716, badiey@udel.edu)

Physical oceanography of estuaries is complex due to many factors including fresh water river discharge, oceanic salt water intrusion, variable ebb and flood tides, and surface winds. In these confined environments, high frequency (1–25 kHz) broadband acoustic waves are shown to through large intensity fluctuations with substantial energy scattering due to their interaction with the sea surface, sea bottom, and refraction due to the stratified water column. In this paper, we present results from a series of experiments that were conducted in the Delaware Bay estuary where fixed source-receiver configurations allowed calibrated long-term acoustic and oceanographic measurements to be conducted while atmospheric conditions such as wind speed and direction were carefully measured concurrently. It is shown that intensity fading due to oceanographic features and tide induced sound speed profile refraction in stratified water column occurs regularly and periodically. Wind generated surface gravity water waves in a fetch limited environment also play an important role in modulating and fading surface bounced acoustic energy. Two dimensional Parabolic Equation (PE) model with rough surface is used to simulate the measured broadband signals. Excellent data-model comparison is shown. [Work supported by ONR321OA.]

9:55

1aAO2. Three-dimensional modeling in the Weymouth Fore River. Michael B. Porter (HLS Res., 12625 High Bluff Dr., Ste 211, San Diego, CA 92130, mikeporter@hlsresearch.com), Timothy F. Duda, Peter Traykovski, Arthur Newhall (Woods Hole Oceanographic Inst., Woods Hole, MA), and Laurel Henderson (HLS Res., La Jolla, CA)

The Weymouth Fore River in Eastern Massachusetts is an estuary that presents an interesting example of a constrained environment. The river bank, of course, channels the energy. Man-made features such as the Fore River Bridge, several piers, and the USS Salem (a heavy cruiser) also present interesting features in terms of the sound propagation. We are using this site as the first of several to validate acoustic modeling tools in such constrained spaces. A detailed bathymetric survey was done providing centimeter level resolution using the Woods Hole Jetyaks. The environmental information is the input to a three-dimensional beam tracing model. This talk will present preliminary comparisons of the measured and modeled acoustic data.

10:15

1aAO3. A propagation experiment in an Arctic glacial fjord. Grant B. Deane (Marine Physical Lab., Univ. of California, San Diego, 13003 Slack St., La Jolla, CA 92093-0238, gdeane@ucsd.edu) and Oskar Glowacki (Polar Sci., Inst. of Geophys., Warsaw, Poland)

A propagation experiment was undertaken in the meltwater-modified surface layer in the bay of a marine-terminating Arctic glacier. Broad-band, m-sequence transmissions were made with an International Transducer Corporation 1007 source deployed from a drifting boat tracked using gps, approximately 500 m in front of Hansbreen Glacier in Hornsund Fjord, Southwestern Svalbard in the summer of 2017. Signals were received with a short array of Hitech HTI-96 hydrophones tethered to an autonomous, anchored boat. The distance between the hydrophone array and source was measured every 2 minutes using a laser rangefinder. Propagation effects in this noisy and dynamic environment include waveguide propagation through the meltwater-modified surface layer and occlusion of transmissions by drifting icebergs. Details of the experiment and propagation effects noted during the transmission period will be presented and discussed. [Work funded by ONR, Grant No. N00014-17-1-2633, and by the Polish National Science Centre, Grant No. 2013/11/N/ST10/01729.]

10:35–10:50 Break

10:50

1aAO4. Spatio-temporal patterns in underwater sound within an urban estuarine river. Sarah A. Marley (Ctr. for Marine Sci. and Technol. (CMST), Curtin Univ., GPO Box U1987, Perth, WA 6845, Australia, sarah.marley86@gmail.com), Christine Erbe (Ctr. for Marine Sci. and Technol. (CMST), Curtin Univ., Bentley, WA, Australia), Chandra P. Salgado Kent, Miles Parsons, and Iain M. Parnum (Ctr. for Marine Sci. and Technol. (CMST), Curtin Univ., Perth, WA, Australia)

Underwater noise environments are increasingly being considered in marine spatial planning and habitat quality assessments, particularly with regard to acoustically specialised fauna. The Swan River in Western Australia flows through the state capital of Perth and consequently experiences a range of anthropogenic activities. However, the river is also extensively used by a resident community of bottlenose dolphins (*Tursiops aduncus*). This study aimed to describe underwater sound sources within the Swan River, examine spatial and temporal soundscape variability, and determine dolphin responses to noisy environments. Acoustic datasets collected from 2005 to 2015 indicated that the Swan River was comprised of multiple acoustic habitats, each with its own characteristic soundscape and temporal patterns in underwater noise. The anthropogenically "noisiest" site was the Fremantle Inner Harbour (mean broadband noise level: 106 dB re 1 μPa rms [10 Hz–11 kHz]); yet dolphins remained present in this area even at high vessel densities. However, fine-scale analyses indicated significant alterations to dolphin behavior at high vessel densities and to dolphin whistle characteristics in high broadband noise conditions. These results highlight the need to consider spatial and temporal patterns when assessing the composition of underwater soundscapes, and identify potential responses of coastal dolphins to busy, noisy environments.

Contributed Papers

11:10

1aAO5. Underwater sound pressure and particle motion (acceleration, velocity, and displacement) from recreational swimmers, divers, surfers, and kayakers. Christine Erbe (Ctr. for Marine Sci. & Technol., Curtin Univ., Kent St., Bentley, WA 6102, Australia, c.erbe@curtin.edu.au), Miles Parsons, Alec J. Duncan, Klaus Lucke, Alexander Gavrilov, and Kim Allen (Ctr. for Marine Sci. & Technol., Curtin Univ., Perth, WA, Australia)

Naturally constrained environments like bays and lakes are frequently used for water sports activities. While the sound of motorized vessels is rather well understood, non-motorized activities have received less investigation. Ten water sports activities (swimming backstroke, breaststroke, butterfly, and freestyle; snorkelling with fins; kicking a boogie board with fins; paddling with alternating or simultaneous arms while lying on a surfboard; scuba-diving; kayaking and jumping into the water) were recorded in a controlled yet even more constrained environment: an Olympic-sized pool. Activities that occurred at the surface involved repeatedly piercing the surface and hence creating the bubble clouds which were the strongest sound generators. Received levels were 110–131 dB re 1μPa (10–16,000 Hz) for all of the activities at the closest point of approach (1 m). All activities produced a characteristic spectro-temporal pattern in all the quantities measured (sound pressure, particle displacement, velocity, and acceleration) by which they could be identified. Applicability of the results to security monitoring of pools, performance assessments of professional or competitive swimmers, and studies of the distances at which humans may be detectable by marine animals in the sea will be discussed.

11:25

1aAO6. Compressional and shear *in situ* measurements in the Lower Laguna Madre. Justin T. Dubin, Megan S. Ballard, Kevin M. Lee, Andrew R. McNeese, Jason D. Sagers, Gabriel R. Venegas (Dept. of Mech. Eng. and Appl. Res. Labs, The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78713, justin.dubin@utexas.edu), Abdullah F. Rahman (The University of Texas Rio Grande Valley, Sch. of Earth, Env., and Marine Sci., Brownsville, TX), and Preston S. Wilson (Dept. of Mech. Eng. and Appl. Res. Labs, The Univ. of Texas at Austin, Austin, TX)

This presentation describes the apparatus, procedure, and results from a field experiment conducted in July 2017 in the seagrass beds of the Lower Laguna Madre. The Laguna Madre is a shallow hypersaline lagoon on the western coast of the Gulf of Mexico. The relatively flat bathymetry is covered by a thick meadow of *Thalassia testudinum*. Acoustic piezoelectric transducer probes measured compressional wave speed and attenuation in the seagrass canopy as well as geoacoustic properties (compressional and shear wave speed and attenuation) of the underlying sediment. Geoacoustic parameters were measured at depths between 5 cm and 20 cm in 5 cm increments. Compressional wave measurements were made over a frequency range of 20 kHz to 100 kHz, and shear wave measurements were made at 500 Hz and 1 kHz. The compressional wave data from this experiment indicates dispersion within the seagrass canopy and sediment. This dispersion is likely due to the high presence of gas bubbles in both the seagrass and supporting sediment. Shear wave velocity monotonically increases with depth at both frequencies. The measured wave speeds and attenuations are compared to results from sediment-acoustic models. Challenges in data collection and processing are explored. [Work sponsored by ARL:UT IR&D and ONR.]

11:40

1aAO7. Measurements and modeling of acoustic propagation in a seagrass meadow. Megan S. Ballard, Jason D. Sagers, Kevin M. Lee, Andrew R. McNeese (Appl. Res. Labs. at the Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78758, meganb@arlut.utexas.edu), Preston S. Wilson (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Abdullah F. Rahman (Sch. of Earth, Env., and Marine Sci., The Univ. of Texas Rio Grande Valley, Brownsville, TX), Justin T. Dubin, and Gabriel R. Venegas (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

This talk presents results from an acoustic propagation experiment conducted in the Lower Laguna Madre to characterize the acoustical properties of a seagrass meadow. At the location of the experiment, the water was one meter deep, and the seabed was covered by a dense growth of *Thalassia testudinum*, a type of seagrass that grows from a long, jointed rhizome buried 5 cm to 10 cm below the seafloor. The biological processes and physical characteristics associated with seagrass are known to affect acoustic propagation due to bubble production, which results in dispersion, absorption, and scattering of sound. During the experiment, a Combustive Sound Source was used to produce broadband signals at ranges of 20 m to 1000 m from the receiver location. Three sensors were positioned at the receiver location: two hydrophones located within and above the seagrass canopy, and a single-axis geophone. The data were analyzed for the purposes of predicting acoustic propagation in seagrass meadows and for estimating environmental parameters in very shallow, biologically active environments. Additionally, seagrasses are a vital part of the global carbon cycle and this work explores the feasibility of using of acoustic signals to estimate carbon stores. [Work sponsored by ARL:UT IR&D and ONR.]

Session 1aNS

Noise, Architectural Acoustics, Physical Acoustics, and Structural Acoustics and Vibration: Novel Materials for Sound Absorption, Insulation, and Vibration Control

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Chair's Introduction—8:30

Invited Papers

8:35

1aNS1. Exploration of acoustical and dynamical properties of organic nanoparticle assemblies for increased sound insulation. Mathew A. Whitney, Ning Xiang (Graduate Program in Architectural Acoust., School of Architecture, Rensselaer Polytechnic Inst., Troy, NY 12180, mathew.whitney@gmail.com), Sadeq Malakooti (Mech. Eng., the Univ. of Texas in Dallas, Richardson, TX), Habel G. Churu (Mech. Eng., LeTourneau Univ., Longview, TX), Nicholas Leventis (Chemistry Dept., Missouri Univ. of Sci. and Technol., Rolla, MO), and Hongbin Lu (Mech. Eng., the Univ. of Texas in Dallas, Richardson, TX)

Low-density highly porous solid monolithic macroscopic objects, consisting of hierarchical mesoporous three-dimensional assemblies of nanoparticles (aerogels), have been previously pursued mainly for their low thermal conductivity. Unlike classical aerogels based on silica being fragile materials, structural fragility issue has been addressed with materials referred to as polymer-crosslinked (or X-) aerogels. X-aerogels are low-density materials, yet their mechanical strengths have been significantly increased. This work explores ductile aerogels in potential uses as constrained damping layers integrated into classical wallboards for drastically increased sound transmission losses without significantly increasing the board thickness and weight. Their ductility and mechanical strength enables lightweight X-aerogel panels of less than 1 cm in thickness to be integrated into gypsum wallboards. This work has experimentally investigated the integrated wallboards to achieve significantly increased sound transmission loss without significantly increasing thickness and weight of integrated wallboard system. This paper discusses preliminary investigations on experimental methods to characterize broadband dynamic properties of X-aerogels for better understanding of its excellent effect in drastically increased sound transmission loss. Some preliminary test results carried out in chamber-based random-incident measurements demonstrate high potentials in building acoustics applications and beyond.

8:55

1aNS2. Large scale metasurfaces for seismic waves control. Antonio Palermo (DICAM, Univ. of Bologna, Viale Del Risorgimento 2, Bologna 40132, Italy, antonio.palermo6@unibo.it), Sebastian Krödel (Dept. of Mech. and Process Eng., ETH, Zurich, Switzerland), Kathryn Matlack (Dept. of Mech. Sci. and Eng., Univ. of Illinois at Urbana-Champaign, Urbana, IL), Rachele Zaccherini, Vasilis Dertimanis, Eleni Chatzi (Dept. of Civil, Environ. and Geomatic Eng., ETH, Zurich, Switzerland), Alessandro Marzani (DICAM, Univ. of Bologna, Bologna, Europe, Italy), and Chiara Daraio (Div. of Eng. and Appl. Sci., California Inst. of Technol., Pasadena, CA)

Elastic metamaterials are artificial composites with subwavelength resonant particles hosted in a medium able to manipulate the propagation of elastic waves. When the resonant particles are placed at the free surface of the medium to form a resonant "metasurface," the localization mechanism and the direction of surface waves can be fully controlled. In this talk, we discuss the use of resonant metasurfaces to control the propagation of vertically and horizontally polarized surface waves and their possible application for seismic waves mitigation. By combining analytical, numerical, and experimental studies, we describe the interaction of Rayleigh waves with a metasurface of vertical resonators and design large-scale resonant barriers to deviate damaging seismic Rayleigh waves into the medium bulk. Additionally, we investigate the effect of material stratification on the metasurface dynamics by analyzing the propagation of surface waves in unconsolidated granular media with depth-dependent stiffness profile. Finally, we describe the interaction of Love waves guided by a stratified medium with a metasurface of horizontal resonators and design large-scale resonant metalenses to redirect their propagation.

9:15

1aNS3. Fractal and bio-inspired labyrinthine acoustic metamaterials. Anastasiia Krushynska, Federico Bosia (Dept. of Phys., Univ. of Turin, Via P. Guiria 1, Turin 10125, Italy, akrushynska@gmail.com), and Nicola M. Pugno (Dept. of Civil, Environ. and Mech. Eng., Univ. of Trento, Trento, Trentino, Italy)

Control of low-frequency sound is a challenge, despite numerous advances in the field. Recently emerged acoustic metamaterials have already proven their efficiency for the development of innovative systems for sound control and demonstration of such fascinating phenomena as super-resolution imaging, transformation acoustics, acoustic cloaking, etc. For sound attenuation, a common attenuation mechanism relies on dispersion induced by local resonances that results in the generation of slow sound inside a metastructure at resonant frequencies. However, the selective bandwidth of metamaterials with identical resonators restricts their practical applications. To overcome this limitation, one can introduce rainbow-type resonators that leads to the increase of structural sizes and raises manufacturing costs. We present fractal and bio-inspired labyrinthine acoustic metamaterials with non-resonant elements for low-frequency sound control. Designed configurations contain sets of narrow channels extending wave propagation paths that leads to reduction of effective sound speed within a structure. We show that by folding these channels into fractal space-filling curves or bio-inspired geometries, one can induce strong artificial Mie-type resonances that allow the achievement of total broadband sound reflection at deep subwavelength scales. This unique phenomenon can be used to develop small-size metastructures with total reflectance of low-frequency sound for versatile applications.

Contributed Papers

9:35

1aNS4. Honeycomb metasurface panel for deep-subwavelength broadband sound absorption. Xiuyuan Peng (Nanjing Univ., Nanjing, China), Yuanchen Deng, and Yun Jing (North Carolina State Univ., 911 Oval Dr., EB III, Campus Box 7910, Raleigh, NC 27606, ydeng5@ncsu.edu)

Honeycomb sandwich panels have long been preferred for their good mechanical properties. Here, we demonstrate a design of honeycomb metasurface panel that can achieve 90% sound absorption from 600 Hz to 1000 Hz with a thickness of less than 30 mm (approximately $\lambda/20$) by applying minor modifications to commercialized honeycomb sandwich panels. The panel was made up of periodically and horizontally arranged honeycomb "supercells" which consist of different types of honeycomb unit cells. Each unit cell could be considered a Helmholtz resonator (HR) with hexagonal prism-shaped cavity and cylindrical neck. The absorption performance of designed honeycomb metasurface panel was studied both with surface impedance theory and the complex frequency plane theory, and was then validated by numerical simulations.

9:50

1aNS5. Ultrathin Ashoka Chakra like acoustic metastructure as a sound absorber. Sanjay Kumar (Dept. of Mech. Eng., Indian Inst. of Technol. Kanpur, Microsystems Fabrication Lab., IIT Kanpur, Kanpur, Uttar Pradesh 208016, India, sanjay21505@gmail.com), Pulak Bhushan (Dept. of Elec. and Comput. Eng., Florida Int. Univ., Miami, FL), and Shantanu Bhattacharya (Mech. Eng., Indian Inst. of Technol. Kanpur, KANPUR, India)

Absorption of undesirable sound is a fundamental issue with broad applications in a variety of situations, such as in a conference room, cinema hall, aircraft cabin noise, etc. A variety of sound absorbers, e.g., micro-slit, concrete walls, porous media, etc., are used for noise control but are required to have a thickness comparable to the working wavelength. Here, we report a novel ultrathin "Ashoka Chakra" like acoustic metastructure that can be utilized as a sound absorber. The developed acoustic metastructure demonstrates broad bandwidth and high absorption characteristics. Each unit cell contains 24 hollow spokes through which acoustic wavefront can move freely. Despite its simplicity, this approach provides tunability of the metastructure functionality such as operating frequency range by altering the structure dimensions. Finite element based simulations were carried out in order to optimize the metastructure's dimensions to achieve maximum absorption. The efficacy of the metastructure is validated through excellent agreement between simulation and experimental measurements. Realization of the proposed acoustic metastructure can show promising applications in the field of sound absorption.

10:05–10:20 Break

10:20

1aNS6. 3D printed sound absorbers using functionally-graded sonic crystals. Matthew D. Guild (Acoust. Div., Naval Res. Lab., Code 7165, 4555 Overlook Ave., SW, Washington, DC 20375, matthew.guild@nrl.navy.mil), Michael Rothko (Dept. of Mech. Eng., Boston Univ., Boston, MA), Caleb F. Sieck (NRC Postdoctoral Res. Program, Naval Res. Lab, Austin, Texas), Charles Rohde, and Gregory Orris (Acoust. Div., Naval Res. Lab., Washington, DC)

In this work, a thin functionally-graded sound absorber that achieves an absorption coefficient near unity is demonstrated. The sound absorber consists of a multilayer arrangement of an interwoven sonic crystal lattice with varying filling fractions, backed by a thin elastic coating that acts as a flexural acoustic element. The overall thickness of the sound absorber is about one tenth of the wavelength in air, and it was 3D printed with a thermoplastic polyurethane. Samples were fabricated and acoustically tested in an air-filled acoustic impedance tube, from which absorption and effective acoustic properties were obtained. A theoretical formulation for the effective acoustic properties of the sonic crystal lattice was used to guide the design process, and excellent agreement was found between measured and theoretically predicted results. A range of sonic crystal filling fractions and thicknesses were tested to verify the fabrication process and robustness of the theoretical formulation, and both were found to be in excellent agreement. Based on this testing and analysis, an optimal arrangement was found that achieved simultaneous near zero reflectance and transmittance over a given frequency band, thereby resulting in an absorption coefficient near unity. [Work supported by the Office of Naval Research.]

10:35

1aNS7. Modeling acoustic resonators: An application to resonator-enclosure coupling. Caleb B. Goates, Scott D. Sommerfeldt (Brigham Young Univ., N283 ESC, Provo, UT 84602, calebgoates@gmail.com), and David C. Copley (Caterpillar, Inc., Peoria, IL)

Acoustic resonators, such as the Helmholtz resonator, provide a stable, cost effective passive noise control solution, and have been widely used to attenuate unwanted sound in enclosures and ducts. Classical formulations predicting the input impedance of such resonators often have significant error, creating a need for repeated prototyping or tuning during fabrication to achieve the desired response. Previous work found that higher-order calculations, including impedance translation and equivalent circuit modelling, produce much more accurate predictions [Calton and Sommerfeldt, J. Acoust. Soc. Am. **139**, 2205 (2016)], allowing prototyping to be done quickly on a computer before fabrication of the resonators. This talk will continue the discussion in the work referenced above. In addition to resonator impedance predictions, resonator-enclosure coupling predictions will be discussed. It will be shown through comparison of predicted and experimental results that the impedance and coupling predictions can remove the need for repeated prototyping and tuning. We will also show the incorporation of

these predictions into a user interface allowing engineers without acoustics background the ability to design resonators for passive noise control.

10:50

1aNS8. Fibrous material microstructure design for optimal structural damping. Yutong Xue and J. S. Bolton (Ray W. Herrick Labs., School of Mech. Eng., Purdue, 177 S. Russell St., West Lafayette, IN 47906, xue46@purdue.edu)

It is shown here that properly designed fibrous media (e.g., glass or polymeric fibers) can be used to damp structural vibration as well as to absorb sound. The materials can then be multi-functional, reducing the number of elements required to achieve a specified level of noise and vibration performance. Past work demonstrated that layers of fibrous media placed on panels can damp the panel motion by removing energy from the nearfield acoustical motion generated by the panel vibration. The current study focused on designing the fibrous medium to ensure optimal vibration damping in a particular application. First, a method of calculating the response of a panel with an attached fibrous layer is recalled and updated to allow layers of limp or elastic porous media to be modeled. Example results will be presented, and then it will be shown that an optimal flow resistivity exists for a given frequency range and configuration of interest. Finally, based on a recently developed theory for the flow resistivity of fibrous media, the optimal flow resistivity identified in that way can be translated into a particular fiber size, given properties such as density of the fiber material and the desired superficial density.

11:05

1aNS9. Groundborne vibration attenuation by particulate dampers to reduce building vibration at sonic frequencies. Hasson M. Tavossi (Dept. of Eng. Technol., Savannah State Univ., Box 20089 3219 College St., Savannah, GA 31404, tavossih@savannahstate.edu)

Particulate medium can act as efficient damper for building vibrations. Mechanical properties of particulate material and their size distributions determine their vibration absorption characteristics. Unconsolidated particulate damper of uniform size is shown to have bandpass filter behavior for mechanical vibrations that are transmitted through them. The dampers absorption spectrum also shows bandgap structure where mechanical vibration in each frequency range is strongly absorbed. We use samples of uniform size spherical particulates that are subjected to mechanical vibrations in the audible frequency range to determine their absorption characteristics as a function of particle size, frequency, intensity, and direction of propagation. The effects of particle sizes and layer thickness on vibration absorption are determined. The goal is to determine the required absorption characteristics as a function of particle size distribution, that is required to mitigate building vibrations.

11:20

1aNS10. Vibration isolation of sensory deprivation tanks. Erik Miller-Klein (A3 Acoust., 241 South Lander St, Ste. 200, Seattle, WA 98134, erik@a3acoustics.com) and Matthew V. Golden (Pliteq, Washington, DC)

Sensory deprivation tanks or float tanks are advertised to be a powerful stress-relief and wellness tool. In these tanks, a person floats in a salt water bath and is enclosed to cut off all visual and auditory sensory inputs. Vibration can have a significant effect on the experience of using a one of these tanks by disrupting the sensory deprivation experience. A recent project will be discussed where vibration was causing such a problem. A solution was developed using recycled rebounded rubber crumb. The subject and object performance data will be shared.

11:35

1aNS11. Improving traffic noise transmission loss of plenum windows by using sound scatterer arrays. Shiu-Keung Tang (Dept. of Bldg. Services Eng., The Hong Kong Polytechnic Univ., Hong Kong, Hong Kong, shiu-keung.tang@polyu.edu.hk)

Plenum windows have been used recently in residential buildings erected next to busy trunk roads. These windows are able to provide an additional 8 dB traffic noise reduction than the conventional casement windows, but at the same time they can allow for natural ventilation of the residential units they protect acoustically. The window design consists of two parallel glass panes—one at the building façade (outer) and the other in the indoor side (inner). The outer and inner window openings are located on opposite window of the window span so that they, together with the cavity between the two window panes, form a passage for air movement, while the window's structure acts as an acoustic filter to attenuate noise. To improve the acoustical performance of this window design, it is proposed in this study to install sound scatterer arrays into the window cavity. It is found from finite-element simulation that such proposal, under appropriate settings, can result in about 4–5 dB improvement of traffic noise transmission loss across a plenum window without significant reduction of air movement across the window. The acoustical performance will further be improved without the increase in air flow resistance when the scatterer rows are staggered.

Session 1aPA

Physical Acoustics, Structural Acoustics and Vibration, Architectural Acoustics, and Underwater Acoustics: Novel Methods in Computational Acoustics I

D. Keith Wilson, Cochair
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Amanda Hanford, Cochair
Applied Research Lab, Pennsylvania State Univ., PO Box 30 - MS 3230D, State College, PA 16804

Chair's Introduction—8:40

Invited Papers

8:45

1aPA1. Deep reinforcement learning for cognitive sonar. Jason E. Summers (ARiA, 1222 4th St. SW, Washington, DC 20024-2302, jason.e.summers@ariacoustics.com), Jason M. Trader (ARiA, Culpeper, VA), Charles F. Gaumond (ARiA, Washington, DC), and Johnny L. Chen (ARiA, Culpeper, VA)

Current developments in cognitive sonar have leveraged explicit semantic representations such as ontologies to develop cognitive agents that enable adaptation of sonar settings to optimize behaviors such as waveform selection during active tracking. However, such cognitive systems based on explicit knowledge representations can be time-and-labor intensive to update in operation and limited in ultimate performance. In other applications, such as computer Go, breakthrough performance was achieved by going beyond learning from all prior games and actions of experts to allowing the algorithm to compete with itself to develop and learn from the outcome of new approaches. This hybrid approach of learning from experts and then learning via self-competition is vital for sonar applications such as active antisubmarine warfare (ASW) because ground-truthed performance data are sparse and may not display optimal solutions. Here, we discuss our application of reinforcement learning to active ASW in a simulated environment based on computational-acoustics models, including an assessment of the goal states and performance metrics we have used. The fidelity of the simulated environment is discussed in terms of the interaction with reinforcement learning and the impact on generalization from learning in simulated environments to application in real environments. [Work supported by ARiA IR&D.]

9:05

1aPA2. Machine learning for spatial interpolation of noise monitor levels. Edward T. Nykaza and Matthew G. Blevins (ERDC-CERL, 2902 Newmark Dr., Champaign, IL 61822, edward.t.nykaza@usace.army.mil)

Continuously recording noise monitoring stations provide feedback of the noise environment at monitor locations. While this feedback is useful, it only provides information at a few point locations and, in many cases, it is of interest to know the noise level(s) at locations between and beyond noise monitoring locations. In this study, we test several machine learning models (e.g., random forests, support vector machines, and neural networks) for their ability to predict noise levels using a small number of noise monitors. Unlike the hybrid geostatistical-acoustical method proposed by Nykaza [Nykaza, Ph.D. thesis, 2013] which requires knowledge of the source location, the methods proposed in this study can be trained to estimate noise levels without knowledge of the source location. The accuracy of each model is assessed with the root-mean-square-error, and we discuss the practical implications of implementing such approaches in real-time systems.

9:25

1aPA3. Coupled FE/BE for periodic acoustic systems. Andrew S. Wixom, Amanda Hanford, Jonathan S. Pitt, and Douglas E. Wolfe (Appl. Res. Lab., Penn State Univ., P.O Box 30, M.S. 3220B, State College, PA 16801, axw274@psu.edu)

While coupled finite element and boundary element (FE/BE) codes are used regularly in acoustics—particularly structural acoustics—and periodic boundary conditions are common, the combination of the three is rare. However, such calculations have been performed in the electricity and magnetism community for quite some time. This talk presents a fully coupled FE/BE framework for acoustics with doubly-periodic boundary conditions, accomplished by way of the Ewald transformation of the appropriate periodic Green's function. This method is likely of primary interest to the acoustic metamaterials community and it is used to analyze an example problem drawn from this field. Both the internal acoustic field within the FE mesh as well as the system's radiation characteristics are examined.

9:45

1aPA4. Verification and validation of a coupled elasto-acoustic-damage system. Jonathan S. Pitt (The Penn State Univ., PO Box 30, State College, PA 16804-0030, jsp203@psu.edu)

A novel time-domain method for simulating dynamic damage evolution in a coupled structural-acoustic system is presented. The system is derived via the theory of continuum damage mechanics, and incorporates standard damage evolution models, but is readily extendible to more complex formulations. The overall solution method is staggered, first solving for the dynamic damage evolution with an explicit step, and then using the new values of damage in the coupled computation of the structural-acoustic system. The spatial domain is discretized using a mixed finite element method, and the temporal domain is discretized with a higher-order implicit time discretization scheme. Code and solution verification of the fully coupled solution algorithm are presented, as are validation studies for cases with and without evolving damage. Applications with evolving damage are presented, and present a novel first principles study of changes in the structural acoustic response to dynamically evolving damage in the structure. Special attention is given to brittle fracture. Examples of downstream usage of the evolving structural response are discussed in the conclusion.

10:05

1aPA5. Numerical investigation of coupling schemes for structural acoustics. Scott T. Miller, Gregory Bunting (Computational Solid Mech. & Structural Dynam., Sandia National Labs., P.O. Box 5800, MS 0845, Albuquerque, NM 87185, stmille@sandia.gov), and Nicholas A. Reynolds (Code 6605, Naval Surface Warfare Ctr. Carderock Div., West Bethesda, MD)

Loosely coupled schemes for structural-acoustic coupling are examined that obtain the same order of accuracy as the monolithic scheme. The coupling algorithms are implemented in Sierra-SD, a massively parallel finite element application for structural dynamics and acoustics. By adapting the predictor-corrector scheme of Farhat *et al.* (2006), second order time accuracy is achieved with the loosely coupled approach. Node-to-face mappings allow arbitrary discretizations of the structural-acoustic interface. Convergence rates are verified with a one dimensional piston problem with known solution. Numerical results are compared to a shock induced plate experimental benchmark. Computational times for loose and strong coupling are compared for a sphere scattering problem. Sandia National Laboratories is a multimission laboratory managed and operated by National Technology and Engineering Solutions of Sandia, LLC., a wholly owned subsidiary of Honeywell International, Inc., for the U.S. Department of Energy's National Nuclear Security Administration under contract DE-NA-0003525.

10:25–10:40 Break

10:40

1aPA6. The Iterative Nonlinear Contrast Source method: Basics and extensions. Martin D. Verweij and Elango Selvam (Acoust. Wavefield Imaging, Delft Univ. of Technol., Lorentzweg 1, Delft 2628CJ, Netherlands, m.d.verweij@tudelft.nl)

There are many applications in which the properties of nonlinear waves play a role. For example, in medical diagnostic imaging, the higher harmonics in nonlinear ultrasound are used to reduce clutter from nearby artifacts and to improve resolution, and in non-destructive testing nonlinear mixing products from crossing ultrasound beams are used to assess the state of materials. Simulation of nonlinear waves is important for optimizing equipment and quantitatively assessing phenomena. Unfortunately, analytical approaches cannot deal with complex realistic situations, while traditional Finite Element and Finite Difference methods require extremely large grids to capture waves with many higher harmonics in large three-dimensional domains. The Iterative Nonlinear Contrast Source method was developed to overcome these issues. It is based on an integral equation involving the analytic Green's function of the linearized medium and a distributed contrast source representing nonlinearity. The integral equation is solved iteratively, and the computations are based on Fast Fourier Transforms that only require two grid points of the shortest wavelength. In this presentation, the basic principles behind the method will be explained and its extension to lossy and inhomogeneous media will be shown. Moreover, a novel extension will be presented that enables the computation of nonlinear elastic waves.

Contributed Papers

11:00

1aPA7. Efficient, wide-band rigid-body and elastic scattering computations using transient equivalent sources. John B. Fahnline (ARL / Penn State, P.O. Box 30, State College, PA 16804-0030, jbf103@arl.psu.edu)

Previous analyses by the author have shown that transient structural-acoustic problems can be solved using time stepping procedures with the structure and fluid modeled using finite elements and equivalent sources, respectively. Here, the analysis is extended to include scattering problems. Although scattering problems have been discussed extensively in the literature, the current formulation is unique because it approximates the acoustic coupling matrix as sparse. Also, most of the previous analyses have assumed the problem is time harmonic, and there is an advantage to performing the computations in the time domain because only a limited number of time steps are required to obtain wideband frequency resolution. This is especially true if the main emphasis is on the mid- to high-frequencies since the ringing response is typically dominated by the lowest frequency modes. Several examples are solved to validate the computations and to document the computation times and solution accuracy.

11:15

1aPA8. Radiation force on spheres in standing waves in ideal fluids: Improved simple approximation for material dependence. Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814, marston@wsu.edu)

When spheres are much smaller than the acoustic wavelength but sufficiently large that viscous dissipative properties are insignificant, the Yosioka-Kawasima approximation for the radiation force in standing waves is widely used. One consequence is that the force is taken to be proportional to the sphere's volume. One approach to finding how corrections to that approximation depend on material properties is to start with Hasegawa's full partial-wave-based series and convert it to partial-wave phase shifts using analytical relations [L. Zhang and P. L. Marston, J. Acoust. Soc. Am. **140**, EL178–EL183 (2016)]. When all the relevant phase shifts are real and sufficiently small in magnitude, a simple approximation can be found for (possibly imaginary) frequencies giving vanishing forces. That was an unanticipated application. (Material dissipation and surface tension are taken to be insignificant.) Using that frequency root, a modification of the Yosioka-

Kawasima radiation force is derived that allows for the liquid or solid sphere to be finite in size [P. L. Marston, J. Acoust. Soc. Am. **142**, 1167–1170 (2017) and at press]. The force is no longer strictly proportional to the volume. This approach is supported by numerical comparisons with Hasegawa's series when the phase shifts are small. [Research supported by ONR.]

11:30

1aPA9. Geometric approximation of radiation stress projections for canonical objects. Philip L. Marston and Timothy D. Daniel (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814, marston@wsu.edu)

Modulated ultrasonic radiation pressure facilitates the selective excitation of modes of objects through modulation of appropriate surface stress projections [P. L. Marston and R. E. Apfel, J. Acoust. Soc. Am. **67**, 27–37 (1980)]. For some large objects, it can be impractical to evaluate the needed stress projections using standard analytical or computational methods. For highly reflecting objects geometric approximations may provide useful insight into the relevant stress projections. The simplest projection gives the translational radiation force on the object. In that case, ray analysis of the far-field scattering was shown to recover the radiation force on large perfectly reflecting spheres in plane waves [P. L. Marston, J. Acoust. Soc. Am. **120**, 3518–3524 (2006)]. In the present research, the appropriate force limit is also recovered through integrating a geometrically approximated local stress projection on the object's surface. Appropriate limiting forces for other canonical objects or situations involving illumination by beams are also found to be recovered using geometrical methods. It was helpful to generalize some prior geometrical results [P. L. Marston, J. Acoust. Soc. Am. **121**, 753–758 (2007)] for scattering by perfectly reflecting spheres. [Work supported by ONR.]

11:45

1aPA10. Time domain simulation of porous surfaces for comparison with the ANSI impedance measurement. Junjian Zhang and Z. C. Zheng (Aerosp. Eng., Univ. of Kansas, 1530 15th St., Lawrence, KS 66045, JJzhang@ku.edu)

This study is to investigate sound propagation with different porous ground conditions. Sound propagations are simulated using finite difference time domain methods (FDTD). The ground materials are modeled as porous media. Two types of models are used in simulating the porous ground: (1) the original Zwikker-Kosten (ZK) model coupled with the immersed boundary method and (2) a modified ZK impedance model, as proposed by Wilson *et al.* (2006, 2007), as a time-domain boundary condition (TDBC) derived from the frequency-domain impedance boundary condition. The first method directly calculates the wave propagation inside porous media, while the second method only specifies the ground surface as a porous boundary condition. Both methods are based on the ZK-type models but have different responses in the frequency domain. Sound pressure levels in different frequency ranges are studied and discussed. Simulation results from both of the models are compared with the measurement data in ANSI S1.18 standard.

MONDAY MORNING, 7 MAY 2018 — NICOLLET D2, 8:00 A.M. TO 12:00 NOON

Session 1aPP

Psychological and Physiological Acoustics: Consequences of Asymmetrical Hearing

Matthew Goupell, Cochair
Hearing and Speech Sciences, University of Maryland, College Park, 0119E Lefrak Hall, College Park, MD 20742

Lina Reiss, Cochair
Otolaryngology, Oregon Health & Science University, 3181 SW Sam Jackson Park Road, Mailcode NRC04, Portland, OR 97239

Chair's Introduction—8:00

Invited Papers

8:05

1aPP1. Electrode length, placement, and frequency allocation distort place coding for bilateral, bimodal, and single-sided deafened cochlear implant users. David M. Landsberger (Otolaryngol., New York Univ. School of Medicine, 550 First Ave., New York, NY 10016, David.Landsberger@nyumc.org)

There is typically a mismatch between the frequency expected at a given cochlear angle and the frequency presented at that angle by a cochlear implant. The frequency mismatch may be particularly important for the population of cochlear implant users who are combining electric hearing across ears or who have electric hearing in one ear and acoustic hearing in the contralateral ear. If the place coding provided by a cochlear implant is misaligned with place coding in the contralateral ear, then the listener may be provided with inconsistent or even conflicting auditory cues. Due to plasticity, the auditory system may be able to compensate for these misalignments. However, data suggest that the ability of some listeners to fully adapt to the distorted or conflicting place maps is limited. The frequency mismatch from a cochlear implant can be adjusted by changing the electrode length, insertion depth, and frequency allocation. Additionally, changing these variables effect the relationship between a change in input frequency and the corresponding change in place pitch.

8:25

1aPP2. Measuring spectral asymmetry for cochlear-implant listeners with single-sided deafness. Joshua G. Bernstein, Olga A. Stakhovskaya, Kenneth K. Jensen (National Military Audiol. and Speech Pathol. Ctr., Walter Reed National Military Medical Ctr., 4954 N. Palmer Rd., Bethesda, MD 20889, joshua.g.bernstein.civ@mail.mil), and Matthew Goupell (Dept. of Hearing and Speech Sci., Univ. of Maryland, College Park, MD)

Cochlear implants (CI) for individuals with single-sided deafness (SSD) are generally programmed with a standard frequency-allocation table (FAT) to maximize the transmission of speech information through the CI. However, a standard FAT is likely to produce mismatch in interaural place of stimulation, which could limit the spatial-hearing benefits that SSD-CIs provide. This study compared three methods for estimating interaural mismatch for SSD-CI listeners: interaural-time-difference (ITD) discrimination, pitch discrimination, and computed-tomography (CT) scan estimates of electrode intracochlear position. For all three measures, the estimated matched acoustic frequency increased for subsequently more basal electrodes. The ITD-based and CT-based estimates were generally consistent with one another, and were on average about one octave above the standard FAT. The pitch-based estimates were considerably lower, averaging about a half-octave above the standard FAT. These results suggest that pitch-discrimination measurements may be more susceptible to plasticity than ITD-discrimination measurements, and that CT images align with binaural function. The measurements will provide a basis to determine whether reducing mismatch can enhance spatial hearing for SSD-CI listeners. [The views expressed in this abstract are those of the authors and do not reflect the official policy of the Department of Army/Navy/Air Force, Department of Defense, or U.S. Government.]

8:45

1aPP3. Results in cochlear implant recipients with substantial contralateral acoustic hearing. Jill Firszt (Otolaryngol., Washington Univ. School of Medicine, 660 South Euclid Ave., Campus Box 8115, St. Louis, MO 63110, firsztj@wustl.edu), Ruth Reeder (Otolaryngol., Washington Univ. School of Medicine, Hillsboro, Oregon), Laura Holden, and Noel Dwyer (Otolaryngol., Washington Univ. School of Medicine, St. Louis, MO)

The present study evaluated longitudinal performance outcomes in adult cochlear implant recipients with substantial hearing in the contralateral ear (mean better-ear PTA of 23.7 dB HL; range 0–40 dB HL). Of the 19 participants, 10 used a better-ear (BE) hearing aid both pre- and post-implant. Test intervals included pre-implant, 6 and 12 months post-implant. Participants' everyday listening condition was primarily a single hearing ear pre-implant and bimodal (CI + BE) post-implant. Results indicated significant improvement over time in the everyday listening condition for sentences at a soft presentation level, sentences in noise (either surrounding or toward the BE), and localization. Self-reported communication ability also improved significantly over time. Group mean CNC word scores for the implanted ear alone improved significantly pre- to post-implant. Ear conditions were compared at the 6-month post-implant test interval. The same measures that demonstrated improvement over time in the everyday listening condition demonstrated significant bimodal improvement compared to the BE alone at the 6-month test interval. Bimodal benefit for these participants varied in degree and by measure compared to adult CI recipients with greater HL in the contralateral ear. Cochlear implantation was an effective treatment for individuals relying on a single hearing ear pre-implant.

9:05

1aPP4. Bimodal neglect: When electrically- and acoustically stimulated ears do not play nice with each other. Mario Svirsky and Arlene C. Neuman (New York Univ., 550 First Ave., NBV-5E5, New York, NY 10010, mario.svirsky@nyumc.org)

This study describes speech perception in quiet, before and after implantation, in 54 adult cochlear implant (CI) patients with usable hearing in the ear contralateral to the CI. Patients had preoperative aided CNC word scores of at least 8% as well as stable unaided hearing in the unimplanted ear throughout the analysis period. Word identification scores in quiet were analyzed for the implanted ear only (CI), acoustic hearing ear only (HA), and the bimodal condition (CI + HA) to determine changes in performance over time and to characterize bimodal benefit (difference between bimodal score and best single-ear score). On average, patients experienced 6% bimodal benefit in quiet after implantation. Not surprisingly, speech scores in the implanted ear and in the binaural condition significantly improved post-implantation. However, scores in the HA decreased unexpectedly after implantation in almost one third of the patients despite unchanged PTA. Although simple explanations like hearing aid malfunction or improper fitting cannot be entirely ruled out, it is possible that these decreases in speech perception may be due to centrally driven neglect of auditory input from one ear when the contralateral ear becomes dominant due to highly asymmetric input.

9:25

1aPP5. Difficulty with understanding speech in background noise is predicted by broad binaural pitch fusion in bimodal cochlear implant users. Yonghee Oh (Otolaryngology-Head and Neck Surgery, Oregon Health & Sci. Univ., Mailcode NRC04, 3181 SW Sam Jackson Park Rd., Portland, OR 97239, oyo@ohsu.edu), Nirmal Kumar Srinivasan (Speech-Lang. Pathol. & Deaf Studies, Towson Univ., Towson, MD), Anna C. Diedesch (Commun. Sci. and Disord., Western Washington Univ., Bellingham, WA), Curtis Hartling (Otolaryngology-Head and Neck Surgery, Oregon Health & Sci. Univ., Portland, OR), Frederick J. Gallun (National Ctr. for Rehabilitative Auditory Res., VA Portland Health Care System, Portland, OR), and Lina Reiss (Otolaryngology-Head and Neck Surgery, Oregon Health & Sci. Univ., Portland, OR)

Combined use of a cochlear implant (CI) and a contralateral hearing aid has been shown to improve CI users' speech perception performance in multi-talker environments, presumably due to the use of acoustic temporal fine structure cues for separating one talker from several other talkers. However, there is large variability in this bimodal benefit. In this study, we show that differences in width of binaural pitch fusion, the fusion of dichotically presented tones that evoke different pitches across ears, explain a large part of this variability. Specifically, broad binaural pitch fusion could lead to fusion of multiple voices as one voice, and reduce the ability to use voice pitch differences to separate voices and understand speech in a multi-talker environment. Speech reception thresholds measured using male and female target talkers were compared with binaural pitch fusion results. Overall performance improved with different genders for target and maskers. A strong negative correlation was observed between voice gender benefit and breadth of binaural pitch fusion. These results suggest that sharp binaural pitch fusion is necessary for maximal speech perception in noise when acoustic hearing is available to transmit voice pitch cues. [Work supported by NIH-NIDCD Grant Nos. R01 DC01337 and F32 DC016193.]

9:45

1aPP6. Dichotic listening performance and listening effort for asymmetrical inputs. Kristina D. Milvae, Stefanie E. Kuchinsky (Hearing and Speech Sci., Univ. of Maryland College Park, LeFrak Hall, 7251 Preinkert Dr., College Park, MD 20742, kmilvae@umd.edu), Olga A. Stakhovskaya (Hearing and Speech Sci., Univ. of Maryland College Park, Bethesda, MD), and Matthew Goupell (Hearing and Speech Sci., Univ. of Maryland College Park, College Park, MD)

Binaural hearing typically improves speech understanding in difficult listening situations. Cochlear-implant (CI) recipients seek to gain binaural hearing benefits through bilateral implantation. However, the standard of care is currently sequential implantation, possibly with a long difference in the duration of deafness between the ears. Sequential implantation may cause interaural asymmetry in the quality of the signal, which could limit binaural benefits. A range of bilateral CI listeners, from those who experience binaural benefits to those who demonstrate interference, were recruited for this experiment. Listeners attended to one ear and reported digits heard in monotic and dichotic conditions. Perceptual performance and listening effort (via pupillometry) were measured. A control group of normal-hearing listeners were tested and were presented unprocessed, four-, and eight-channel vocoded digits, and the processing was independent across ears. It was hypothesized that attending to a poorer-quality signal in the presence of a better-quality signal would lead to poorer performance than in the monotic condition. Pupillometry was hypothesized to reflect the listener's ability to perceive the poorer-quality signal in the presence of a better-quality signal. Results will inform strategies to maximize binaural hearing in bilateral CI listeners.

10:05–10:20 Break

10:20

1aPP7. Early temporary asymmetrical hearing impairs behavioral and neural sensitivity to sound location. Kelsey L. Anbuhl and Daniel J. Tollin (Dept. of Physiol. & Biophys., Univ. of Colorado School of Medicine, University of Colorado Anschutz Medical Campus, 12631 E 17th Ave., Aurora, CO 80045, Daniel.Tollin@ucdenver.edu)

Children experiencing asymmetrical hearing early in life often display binaural hearing impairments that persist long after symmetric hearing is restored, suggesting abnormal central auditory development. Here, we test the hypothesis that abnormal neural coding of the interaural level difference (ILD) cue to sound location, resulting from asymmetrical hearing during development, underlies the hearing impairments. Guinea pigs were reared for 8 weeks from birth with a unilateral conductive hearing loss (CHL via earplug). After earplug removal we measured behavioral spatial acuity, the binaural interaction component (BIC) of the auditory brainstem response (ABR), and sensitivity to ILDs in auditory midbrain neurons (inferior colliculus, IC). Animals raised with asymmetric hearing displayed worse spatial acuity (~2x worse) for high-pass noise compared to age-matched controls, suggesting impaired sensitivity to ILDs. Physiologically, animals displayed abnormal BICs of the ABR indicating altered binaural processing in the auditory brainstem. Additionally, ILD discrimination thresholds for single neurons in the IC, determined using Fisher Information, were ~2 times worse than that found in controls. These results suggest that experiencing even temporary asymmetric hearing during early development persistently alters the normal development of spatial hearing, which may be attributed to impaired binaural processing at the level of the brainstem and midbrain.

10:40

1aPP8. Auditory and non-auditory effects of asymmetrical hearing loss in animal models. Amanda Lauer (Otolaryngology-HNS, Johns Hopkins Univ. School of Medicine, 515 Traylor, 720 Rutland Ave., Baltimore, MD 21205, alauer2@jhmi.edu)

Animal models offer an opportunity to investigate the consequences of asymmetrical hearing loss with a level of control over extraneous variables that is not possible with human listeners. Experimental manipulations to induce unilateral conductive or sensorineural hearing loss in animal models have demonstrated substantial and diverse changes in physiological function, anatomical reorganization of neural projections, and adverse effects on auditory behaviors. Our experiments in rodents exposed to acoustic trauma with one ear plugged result in asymmetrical hearing loss and abnormal auditory brainstem response waveforms that occur when either the unprotected or protected ear is stimulated. Susceptibility to auditory brainstem response deficits appears to differ between the right and left sides. Asymmetrical hearing loss is associated with increased startle reactivity to loud sounds, suggesting abnormal loudness perception or an increased emotional response to sound. These animals also show abnormal social interaction, and some subjects show reduced exploratory behavior in a novel environment indicative of heightened anxiety. These studies provide a framework for investigating the interplay between the auditory and non-auditory behavioral consequences of hearing loss, as well as the underlying neuropathology.

11:00

1aPP9. Talk to my better ear: Consequences of asymmetric bilateral hearing in development. Karen A. Gordon, Melissa Polonenko, and Blake Papsin (The Hospital for Sick Children, Rm. 6D08, 555 University Ave., Toronto, ON M5g 1x8, Canada, karen.gordon@utoronto.ca)

We study consequences of asymmetric hearing in children with bilateral profound deafness who used a unilateral cochlear implant before bilateral implantation and children who had access to acoustic hearing in one ear with or without a hearing aid before receiving a cochlear implant in the opposite ear (bimodal). Electrophysiological and behavioral measures post-implantation reveal benefits of bilateral implant/bimodal device use but also persistent consequences of early asymmetric auditory input. Auditory cortices maintain an aural preference for the first hearing ear despite chronic bilateral/bimodal hearing. Asymmetries are also evident in speech detection and perception; children receiving bilateral cochlear implants with long delays continue to favour their first implanted ear in both quiet and noise. Children with bimodal devices who have insufficient hearing in the non-implanted ear begin to prefer to listen with the cochlear implant. Impaired binaural hearing in groups with asymmetric function are evidenced by poor integration of input from the two devices and impaired perception and cortical processing of interaural timing differences. Together, these data suggest that children who had a period of asymmetric hearing in early life continue to have aural preference for a "better ear" which could compromise development of binaural/spatial hearing.

11:20

1aPP10. The effects of early acoustic hearing on speech perception, language, and literacy abilities of pediatric cochlear implant recipients. Lisa S. Davidson (Otolaryngology, Washington Univ. School of Medicine in St. Louis, 4523 Clayton Ave., Campus Box 8115, St. Louis, MO 63110, davidsonls@wustl.edu)

The primary aim of our current project is to determine the degree to which a period of hearing aid (HA) use facilitates language development in children following the receipt of their first cochlear implant (CI). We tested 117 pediatric CI recipients ranging in age from 5 to 8 years on tests of speech perception and standardized tests of receptive vocabulary and language. Follow up testing (ages 7–10 years) included early literacy assessments. The speech perception test battery includes word recognition in quiet and noise (segmental perception), talker and stress discrimination, and emotion identification (suprasegmental perception). A continuum of residual hearing levels and length of HA use are represented: some children have bimodal devices, while others received their second CI either simultaneously or sequentially at varying time intervals since their first CI (mean age at 1^{st} CI/s = 2.1 years). Comprehensive threshold and device histories were collected on all CI recipients. Based on our analyses, we identified a critical duration of HA use for each of three ranges of hearing loss severity (pure-tone-averages of 72, 92, and 110 dB HL) that maximizes suprasegmental perception (which operates independently from segmental perception to promote spoken language and literacy).The clinical implications of these results will be discussed.

11:40

1aPP11. Interhemispheric auditory cortical relationships in asymmetric hearing loss. Steven W. Cheung (Otolaryngology-Head and Neck Surgery, Univ. of California San Francisco, 2233 Post St., Ste. 341, San Francisco, CA 94115, Steven.Cheung@ucsf.edu)

Directional acoustic trauma, sudden hearing loss with incomplete recovery, Meniere's disease, and cerebellopontine angle tumors are common clinical conditions that present with asymmetric hearing loss (ASL). Presentation of input signals from both ears for interhemispheric processing may be necessary to achieve the best hearing outcomes. To review temporal and spatial interhemispheric functional relationships revealed by microelectrode mapping studies in an animal model of ASL and by functional imaging studies in human single-sided deafness (SSD). Animal and human cohort comparison studies by investigators at the University of California. In animals with mild-to-moderate ASL, interaural temporal difference is a strong driver of cortical plastic change. Six months after hearing loss induction, the two cortices reorganize into a state of relative local hemispheric autonomy. In humans with SSD, temporal and spatial cortical plastic change is evidenced by reduced interhemispheric mean difference of the M100 peak response latency and increased activation spread distance in the hemisphere contralateral to the only hearing ear and decreased distance in the ipsilateral hemisphere in SSD subjects to tonal stimuli, respectively. Variations in hearing outcomes for acoustical and electrical rehabilitation of the poorer ear in ASL may be related particular states of interhemispheric relationships.

Session 1aSP

Signal Processing in Acoustics, Underwater Acoustics, and Architectural Acoustics: Signal Processing for Complex Environments. I

Sandra L. Collier, Cochair
U.S. Army Research Laboratory, 2800 Powder Mill Rd., RDRL-CIE-S, Adelphi, MD 20783-1197

Kainam T. Wong, Cochair
Dept. of Electronic & Information Engineering, Hong Kong Polytechnic University, 11 Yuk Choi Road, Hung Hom KLN, Hong Kong

Max Denis, Cochair
U.S. Army Research Lab., 2800 Powder Mill Road, Adelphi, MD 20783-1197

Invited Papers

9:00

1aSP1. Outdoor acoustic tomography using retrieved Green's function from controlled-sources: An Experimental Feasibility Study. Max Denis (U.S. Army Res. Lab., 2800 Powder Mill Rd., RDRL-CIE-S, 1 University Ave., Lowell, Massachusetts 01854, max_f_denis@hotmail.com), Sandra L. Collier, John Noble, W. C. Kirkpatrick Alberts, David Ligon, Leng Sim, Christian G. Reiff, and Deryck D. James (U.S. Army Res. Lab., 2800 Powder Mill Rd., RDRL-CIE-S, Adelphi, MD)

In this work, outdoor acoustic tomography is investigated using the retrieve Green's function originating from a virtual source located inside a region of interest (ROI). Of particular interest are the location and amplitude of reflectors, as well as the extraction of atmospheric conditions. Existing methods utilize Green's function retrieval approaches, such as cross-correlation (CC) and multidimensional deconvolution (MDD), of reflection data from inhomogeneities (virtual sources) within the ROI. This allows one to retrieve the Green's function without detailed knowledge of the medium itself. The imaging resolution, artifacts, and close match with the true reflectors are investigated and discussed for both Green's function retrieval methods.

9:20

1aSP2. Propagation effects on acoustic particle velocity sensing. Sandra L. Collier, Max Denis, David Ligon, Latasha Solomon (U.S. Army Res. Lab., 2800 Powder Mill Rd., RDRL-CIE-S, Adelphi, MD 20783-1197, sandra.l.collier4.civ@mail.mil), Madeline M. Erikson (U.S. Military Acad., West Point, NY), John Noble, W. C. Kirkpatrick Alberts, Leng Sim, Christian G. Reiff, and Deryck D. James (U.S. Army Res. Lab., Adelphi, MD)

The effects of atmospheric turbulence on the acoustic particle velocity are examined for both narrow-band and wide-band sources. The statistical distributions of the acoustic particle velocity and pressure fields are analyzed for a series of field tests. Applications to signal processing with particle velocity sensors are considered.

9:40

1aSP3. Obtaining acoustic intensity from multisource statistically optimized near-field acoustical holography. Trevor A. Stout (Brigham Young Univ., N283 Eyring Sci. Ctr., Provo, UT 84602, tastout6@gmail.com), Alan T. Wall (Battlespace Acoust. Branch, Air Force Res. Lab., Wright-Patterson AFB, OH), Kent L. Gee, and Tracianne B. Neilsen (Brigham Young Univ., Provo, UT)

Multisource statistically optimized near-field acoustical holography (M-SONAH) improves the field reconstruction process by directly incorporating into the pressure propagator types of wavefunctions that correspond most closely to the source geometries of interest [A. T. Wall *et al.*, J. Acoust. Soc. Am. **137**, 963–975 (2015)]. The M-SONAH method has previously been used to localize acoustic sources in a full-scale jet engine plume above a rigid reflecting plane by adding a second set of cylindrical wavefunctions corresponding to the image source [A. T. Wall *et al.* J. Acoust. Soc. Am. **139**, 1938–1950 (2016)]. Here, M-SONAH theory is extended to obtain the vector particle velocity and, by extension, the acoustic intensity. Discussed are two examples that relate to the full-scale jet noise-with-image-plane reconstruction problem: (1) a Gaussian line source with image and (2) a jet-like wavepacket and image, with hologram geometry identical to that of the full-scale experiment. The results from both examples reveal intensity errors less than 3 dB and 10 degrees within the top 20 dB of the reconstruction region. The results also suggest that intensity reconstruction magnitudes less than those obtained at the measurement aperture edges should be discarded. [Work supported by ONR.]

10:00

1aSP4. Directional pointing error in "Spatial Matched Filter" beamforming at a tri-axial velocity-sensor with non-orthogonal axes. Chibuzo J. Nnonyelu (Electron. and Information Eng., The Hong Kong Polytechnic Univ., BC606 (Table 3), 11 Yuk Choi Rd., Hung Hom, Hung Hom, Kowloon 999903, Hong Kong, joseph.nnonyelu@connect.polyu.hk), Charles H. Lee (Dept. of Mathematics, California State Univ., Fullerton, CA), and Kainam T. Wong (Electron. and Information Eng., The Hong Kong Polytechnic Univ., Hung Hom, Hong Kong)

The "tri-axial velocity-sensor" has three axes that are nominally perpendicular, but may be non-perpendicular in practice, due to real-world imperfections in manufacturing, deployment, or maintenance. This paper investigates how such non-perpendicularity would affect the tri-axial velocity-sensor's azimuth-elevation beam-pattern in terms of the beam's pointing direction.

10:20–10:35 Break

10:35

1aSP5. Sparse array performance in channels of limited coherence. Andrew T. Pyzdek (Graduate Program in Acoust., Penn State Univ., PO Box 30, State College, PA 16804, atp5120@psu.edu) and David C. Swanson (Appl. Res. Lab., The Penn State Univ., State College, PA)

Sparse arrays reduce the number of redundant measurements for lags on an array in comparison to fully populated arrays. For ideal measurements, where spatial stationarity is maintained and measurement error is minimal, the performance of such arrays can be shown to be similar to that of a uniform array of the same length with appropriate processing techniques. However, for complex environments with limited spatial coherence or noisy measurements, array sparsity can impact performance. We construct a simple model that relates the relative positions of redundant measurements and channel coherence length to the variance of covariance function estimates, determine a set of lag measurement weightings that minimize said variance for a given set of redundant lag positions, determine a set of conditions on which a uniform weighting is optimal, and demonstrate that regularly spaced lag repetitions optimally sample a spatially varying covariance function.

10:55

1aSP6. Evaluating the effectiveness of subarrays in beamforming computations of complex jet noise source environments. David F. Van Komen, Tracianne B. Neilsen, Blaine M. Harker, Kent L. Gee, S. Hales Swift (Phys. and Astronomy, Brigham Young Univ., N283 Eyring Sci. Ctr., Provo, UT 84602, david.vankomen@gmail.com), Alan T. Wall (Air Force Res. Lab., Wright-Patterson AFB, OH), Micah Downing, and Michael M. James (Blue Ridge Res. and Consulting, LLC, Asheville, NC)

Conventional acoustic beamforming methods, while powerful for localizing acoustic sources, require an uncorrelated monopole assumption, depend on the input array location, and are unreliable in a complex source environment, such as jet engine noise. The complexities arise from the large, partially correlated, extended source region, the multiple types of noise sources, such as directional and omnidirectional, and partially correlated sources that violate the integral uncorrelated monopole assumption of beamforming. The complexity of these sources requires advanced beamforming methods. The aim of this research is to determine what advanced beamforming methods tell us about complex, full-scale jet noise sources. To learn about these sources, an array spanning the entire length of a jet from an F-35 aircraft is split into multiple subarrays. Application of the beamforming algorithm to a subarray in the maximum sound region localizes the directional sources present in jet noise; a subarray to the side of the engine characterizes more omnidirectional sources; and a subarray farther forward yields the approximate origin of broadband shock-associated noise. Using these advanced beamforming methods, combined with the subarray processing, opens new insights in this complex environment. [Work supported by the Air Force Research Laboratory; data courtesy of F-35 JPO.]

11:15

1aSP7. The blended dominant mode rejection adaptive beamformer. John R. Buck (ECE Dept., UMass Dartmouth, 285 Old Westport Rd., North Dartmouth, MA 02747, jbuck@umassd.edu), Andrew Singer (ECE Dept., Univ. of Illinois Urbana-Champaign, Champaign, IL), and Kathleen E. Wage (ECE Dept, George Mason Univ., Fairfax, VA)

Adaptive beamformers balance competing demands of enhancing desired signals, suppressing discrete interferers and attenuating background noise. This requires an accurate estimate of the spatial covariance matrix that is challenging to obtain in nonstationary environments. The dominant mode rejection (DMR) beamformer [Abraham & Owsley, Oceans, 1990] addresses this challenge by replacing the sample covariance matrix (SCM) eigenvalues of the noise subspace with the average of these eigenvalues to improve its conditioning. The DMR beamformer requires an estimate of the interferer subspace dimension, and the DMR beamformer's performance degrades when this estimated dimension is inaccurate. We propose the Blended DMR beamformer, whose array weights are formed as a mixture of fixed-dimension DMR beamformer array weights across a range of interferer dimensions. The contribution of each fixed-dimension DMR beamformer is a function of its recent performance in suppressing interferers and attenuating noise. The performance of this blended DMR beamformer approaches that of the best fixed-dimension DMR beamformer in stationary environments, and can outperform any fixed DMR beamformer in many nonstationary environments. [Work supported by ONR Code 321US.]

11:35

1aSP8. Frequency-difference beamforming with multiple realizations in an inhomogeneous environment. Alexander S. Douglass and David R. Dowling (Mech. Eng., Univ. of Michigan, 2010 AL, 1231 Beal, Ann Arbor, MI 48109, asdougl@umich.edu)

Frequency-difference beamforming is a sparse-array signal processing method that shifts the signal analysis to lower, out-of-band (difference) frequencies, provided that the difference frequency does not exceed the signal bandwidth. In an inhomogeneous environment with discrete scatterers, low-frequency fields may not be significantly affected, provided the scatterers are small relative to the

signal wavelength. However, when the scatterer sizes are comparable to the signal wavelength, scattering effects are likely to be more significant. Previously, it was shown that frequency-difference beamforming can overcome some of the performance degradation from the high frequency scattering by shifting the processing to lower, out-of-band frequencies. In this case, the frequency-difference output was comparable to the output using a lower-frequency signal in the same environment. In this talk, theoretical fields that include scattering at the in-band frequencies are considered to predict the performance of the frequency-difference method. Furthermore, simulations and experiments in a 1.07-m-diameter water tank with a sparse array are used to determine the performance and robustness of the method in the presence of discrete high-contrast scatterers. Multiple realizations of randomly placed scatterers are combined to quantify performance statistics. [Sponsored by NAVSEA through the NEEC, and by ONR.]

MONDAY AFTERNOON, 7 MAY 2018 NICOLLET C, 1:00 P.M. TO 4:15 P.M.

Session 1pAA

Architectural Acoustics and Noise: Architectural Acoustics Experimental Methods in Laboratories

Jin Yong Jeon, Cochair
Department of Architectural Engineering, Hanyang University, Seoul 133-791, South Korea

Ning Xiang, Cochair
School of Architecture, Rensselaer Polytechnic Institute, Greene Building, 110 8th Street, Troy, NY 12180

Invited Papers

1:00

1pAA1. Portable goniometers with high angular resolutions and large radii for experimental investigations of acoustic diffractions. Jonathan Kawasaki, Ning Xiang, Jolene Stoffle (School of Architecture, Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, kawasj@rpi.edu), and Peter D'Antonio (RPG Acoust. Systems, LLC, Passaic, NJ)

Surface scattering and diffuse reflections of acoustic diffusers are of relevance in room-acoustics simulations or for characterization purposes. Polar responses of diffraction objects to acoustic plane-wave excitations can be experimentally characterized using a circular microphone array: acoustical goniometer. Recently, the Graduate Program in Architectural Acoustics at Rensselaer Polytechnic Institute has begun to explore the advanced physical theory of diffraction (PTD) [Xiang and Rozynova, *J. Acoust. Soc. Am.*, **141**, pp. 3785 (2017)]. In order to validate the PTD applied to a number of canonical shapes (rigid wedges or plates) of finite sizes, experimental investigations on the diffraction objects need to be carried out with high angular resolutions and sufficiently wide angular ranges. A large radius of the acoustical goniometer is also crucial to satisfy acoustical far-field conditions. This talk will discuss the effort in implementation and processing algorithms for various radii of portable acoustical goniometers between 1 m and 4 m with angular resolutions up to 1.25 degrees, that can be easily deployed in indoor empty spaces of sufficient dimensions for an angular range across 180 degrees and beyond when diffusers or diffraction objects with one-axis symmetry need to be experimentally characterized.

1:20

1pAA2. High spatial resolution scanning for experimental room-acoustic measurements in scale models. Daniel Tay, Ning Xiang, and Aditya Alamuru (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., School of Architecture Rensselaer Polytechnic Inst. 110 8th St. - Greene Bldg., Troy, NY 12180, tayd@rpi.edu)

This paper presents an experimental study of low frequency behavior of sound propagation in acoustically coupled volumes. The experiments are carried out in an eighth-scale model of two coupled volumes with an automated high spatial-resolution scanning system. Both sound pressure and three-dimensional sound intensities are automatically measured in terms of room impulse responses at each scanning positions over densely defined two-dimensional grids covering the entire areas of the two coupled rooms that enable the analysis and visualization of sound energy and sound energy flows. Through these measurements, and with the aid of a predictive computational model, this paper will discuss low frequency modal behaviors of sound energy flows within one of two coupled volumes. As such, the distribution of potential and kinetic energies, as well as that of active and reactive sound intensities within the coupled space will be separately analyzed. The distributions of intensity vector fields are compared through computational simulation. Consequently, a discussion on possible discrepancies in modal distributions between the computational and experimental results rounds off this paper.

1:40

1pAA3. The irrelevant speech effect in multi-talker environments: Applications to open-plan offices. Manuj Yadav, Densil Cabrera (The Univ. of Sydney, Rm. 469, Wilkinson Bldg., 140, City Rd., Darlington, NSW 2008, Australia, manuj.yadav@sydney.edu.au), Lucas Brooker (Sydney, NSW, Australia), James Love, Jungsoo Kim, and Richard de Dear (The Univ. of Sydney, Sydney, NSW, Australia)

Irrelevant speech from co-workers has consistently been listed as one of the major nuisances within open-plan work environments. To explore the psychoacoustic basis of the so-called irrelevant speech effect (ISE; which causes cognitive decline and increases acoustic distraction) within a multi-talker context, a series of laboratory-based experiments were run. These experiments used room acoustical simulations of various open-plan office environments within a climate-controlled chamber to test several factors relevant to the ISE. These factors included the number of simultaneously active talkers, their spatial arrangements, sound pressure levels and the semanticity of speech. The main findings thus far indicate that multi-talker speech lead to a stronger ISE than speech from a single talker, which further interacts with the overall sound level, semanticity, etc., to exhibit a more complicated effect than what has been proposed in previous studies. The effect of the number of active talkers alone suggests reconsiderations of some assumptions in the ISO 3382-3. More generally, the perception of speech "babble" seems to be affected by multiple factors in a realistic working environment. Some of our research is directly related to characterising speech babble and its function as a beneficial sound masker in multi-talker environments.

2:00

1pAA4. Experiments with high-back chairs and retroreflective surfaces for increasing the support of one's own voice over short conversational distances. Densil Cabrera, Manuj Yadav, Dagmar Reinhardt, Jonothan Holmes, Beau Ciccarello, Hugo Caldwell, and Adam Hannouch (The Univ. of Sydney, Rm. 589, Wilkinson Bldg., 140, City Rd., NSW 2006, Australia, manuj.yadav@sydney.edu.au)

Acoustic support of one's own voice affects speech, and increased support can foster more relaxed voice projection, which reverses the Lombard effect. While hearing one's own voice in typical rooms shows subtle influences of "global" room acoustics, local treatment can yield stronger effects for talking-listeners. This paper considers two types of architectural acoustic treatment for supporting one's own voice and modifying speech propagation—high-back chairs and retroreflective ceilings. For a talking-listener, local acoustic treatment such as high-back chairs can be designed to selectively attenuate ambient noise while providing enhanced reception of sound from a particular direction, project speech towards a listener with increased gain for speech intelligibility, and also provide voice support. Acoustically retroreflective surfaces (e.g., ceilings and vertical partitions) provide increased voice support by reflecting a person's voice back to them, without such local treatment. This also has the advantage of reducing the transmission of speech beyond the conversation, providing a way to ameliorate speech distraction in multi-talker environments. This paper describes experiments in which the voice support of such treatments is quantified, via oral-binaural transfer functions. Results are supported with numerical simulation (finite-difference time-domain), and discussed in relation to practical design solutions for speech privacy and vocal comfort.

2:20

1pAA5. Diffusion: How new research and data formats can be of use in simulations. Ronald Sauro (NWAA Labs, Inc., 90 Tower Blvd., Elma, WA 98541, audio_ron@msn.com)

We will explore the new ways of measuring diffusion using free field methods and how those resulting data formats can be used in simulation programs such as EASE etc. This includes the calculation of complex balloons that reflect the total directivity of entire arrays of both diffractive and geometric styles of "diffusors." We will show actual measurements vs simulations. Included will be research done by Ron Sauro and extensions done by Jim DeGrandis that include some new graphs that extend our usage of phase data in an understanding of the psychoacoustic behavior of the human brain and our perception of how much data is of use to our designs.

2:40–2:55 Break

2:55

1pAA6. Old problems, new solutions: Architectural acoustics in flux. Ronald Sauro (NWAA Labs, Inc., 90 Tower Blvd., Elma, WA 98541, audio_ron@msn.com)

Many problems that have plagued architectural acoustics for years and have escaped real solutions because of a lack of research and measuring facilities. We will explore the types of problems that are still around and the facilities that are needed to explore those problems. This will include reverberation rooms that have the abilities to explore VLF problems in absorption and transmission loss as well as sound power problems at these irritating very low frequencies. This exploration will include looking at present day abilities including specifications of the future rooms that are needed. Extended frequency response information is needed in today's litigious society when everything offends someone else and helps consultants provide solutions to present and future problems.

Contributed Papers

3:15

1pAA7. Field testing of in-room impact noise. John LoVerde and David W. Dong (Veneklasen Assoc., 1711 16th St, Santa Monica, CA 90404, jloverde@veneklasen.com)

The sound generated by footfall within the source space, sometimes called in-room impact noise, affects the noise level due to movement within a room and also the occupants' perception of quality. Initial studies are based on the standard tapping machine as the source. While no ASTM standard yet exists for measuring the source room sound level, source room measurements with the tapping machine have been performed on several recent laboratory testing programs. The authors have measured source room power levels in the field using the same flooring products that were measured in the laboratory. The data are analyzed to evaluate the correlation between the laboratory data and the resultant noise level realized in the field. The usefulness and applicability of future laboratory and field in-room impact noise standards is discussed.

3:30

1pAA8. Impact noise of bare concrete slabs. David W. Dong, John LoVerde (Veneklasen Assoc., 1711 16th St, Santa Monica, CA 90404, jloverde@veneklasen.com), and Michel Morin (MJM Acoust. Consultants Inc, Montreal, QC, Canada)

A poured concrete slab with no finish flooring and no ceiling is one of the simplest floor assemblies possible. Most structural slabs share similar design parameters such as density, mix, and compressive strength, and might be expected to have a similarly limited range of impact noise isolation. The authors have compiled a large number of bare slab tapping tests performed in laboratories, which provides information on the uncertainty in the test method for the specific assemblies used in the laboratory tests. The authors have compiled a large number of field impact tests on bare slabs in buildings, and the data reveals a surprisingly large variation in the impact sound pressure levels for actual slabs. The effects of this variation is discussed, including the suitability of various ratings of impact sound insulation, the relationship between laboratory and field measurements, and the uncertainty of impact noise measurements.

3:45

1pAA9. Drop Tower Testing—The next step in heavy weight impact testing. Matthew V. Golden (Pliteq, 616 4th St., NE, Washington, DC 20002, mgolden@pliteq.com)

Over the last few years, research into heavy hard impacts from fitness activities has become an active field of investigation and product development. In order in facilitate our efforts, Pliteq has built a custom drop tower capable of repeatability dropping up to 150 kg from well over 2 m height. During the impact, the force, displacement, velocity, and acceleration can be obtained. This paper will give an overview of this apparatus, the type of data that it can collect, how this data can be used to make better products and how the data can be used to predict field performance.

4:00

1pAA10. Finite element modeling of Rayleigh wave propagation in a laboratory sand tank. Anthony F. Ragusa, James H. Miller, and Gopu R. Potty (Ocean Eng., Univ. of Rhode Island, 15 Receiving Rd., Narragansett, RI 02882, afragusa@my.uri.edu)

Rayleigh waves were measured in a laboratory sand tank using an array of accelerometers. This tank allows for the measurement of interface waves for various sediment and soil arrangements in a well-controlled laboratory setting. Rayleigh waves were excited by simple impulsive sources including dropped balls and hammer impacts. In this study, a finite element modeling tool developed by Sandia National Laboratories was used to simulate the sand tank during these excitations. Different modeling scenarios were investigated. Time series of the Rayleigh wave arrivals were modeled at the sensor locations using time domain analysis. A calibrated bender element system was also developed to provide direct measurements of the shear wave speed profile at selected depths. These data were used to develop the material property input parameters for the finite element analysis. The simulated time series at the sensor locations are compared to the accelerometer data. [Work supported by Army Research Laboratory and BOEM.]

MONDAY AFTERNOON, 7 MAY 2018 LAKESHORE B, 1:20 P.M. TO 3:50 P.M.

Session 1pAB

Animal Bioacoustics: Bat and Biomimetic Sonar/Marine Mammal Hearing and Sound Production

Rolf Müller, Chair
Mechanical Engineering, Virginia Tech, ICTAS Life Sciences District (Mail Code 0917), Virginia Tech, Blacksburg, VA 24061

Contributed Papers

1:20

1pAB1. Biosonar beampattern measurements in flying old world leaf-nosed bats. Yanan Zhao, Hui Ma (Shandong University - Virginia Tech Int. Lab., Shandong Univ., Shandong University Central Campus, Jinan, Shandong 2501000, China, 820123652@qq.com), and Rolf Müller (Mech. Eng., Virginia Tech, Blacksburg, VA)

Emission beampatterns, i.e., emitted ultrasound amplitude as a function of direction and frequency, could be an important characteristic of a bat's biosonar since they determine which part of an environment is illuminated with a given frequency and at a given time. Numerical results have predicted intricate geometrical features such as multiple peaks and notches in bat biosonar beampatterns. Furthermore, it has been shown that bats with nasal emission change the shapes of their emission baffles, "noseleaves," during pulse emission. The work presented here is aimed at understanding if and how these two aspects (beam geometry and emission dynamics) are integrated into the bats' biosonar behaviors on the wing. The experiments are conducted with Old World leaf-nosed bats (*Hipposideros pratti* and *Hipposideros armiger)* that have a pronounced noseleaf dynamics. The animals fly in a tunnel that is instrumented with synchronized arrays of high-speed video cameras and microphones. The setup allows for precise tracking of the bats with high spatial resolution as well as beampattern estimates as a function of time. A first pilot dataset has indicated the presence of local shape features in the beampattern that can also vary with time. However, further testing is needed to corroborate these findings.

1:35

1pAB2. Characterization of pinna motion patterns in hipposiderid bats. Peiwen Qiu and Rolf Müller (Mech. Eng., Virginia Tech, 1075 Life Sci. Cir, Blacksburg, VA 24061, peiwenq@vt.edu)

Hipposiderid bats have a set of highly differentiated pinna muscles that support different pinna motion patterns during biosonar behaviors. It has been shown that these bats' pinna motions fall into two separate categories: Rigid motions that change only pinna orientation and non-rigid motions that also change pinna shape. It remains unclear if further subdivisions of these two categories (e.g., different orientations for rigid motions and different deformation patterns for non-rigid motions) can be made. In order to investigate this question, experiments have been conducted where the positions of a dense set of landmark points (50–80 per pinna) were reconstructed based on footage from an array of four high-speed video cameras. The resulting point trajectory data (positions as a function of time) was clustered to facilitate the detection of patterns. Clustering revealed a structure of coherent surface patches for rigid as well as non-rigid motion. For the rigid motion pilot data set, these patches followed a constant pattern that reflects different motion directions. Within the non-rigid motion pilot data, there were similarities as well as pronounced differences among the analyzed motion sequences, which leads us to formulate the hypothesis that there could be more than one type of non-rigid pinna motion.

1:50

1pAB3. System integration for a biomimetic dynamic sonar head. Dean Conte (Mech. Eng., Virginia Tech, 1075 Life Sci. Cir, Blacksburg, VA 24061, dconte97@vt.edu), Lucas Mun, Shubham Dawda, Brandon Walker (Elec. and Comput. Eng., Virginia Tech, Blacksburg, VA), Joseph Sutlive (Translational Biology, Medicine, and Health Program, Virginia Tech, Roanoke, VA), and Rolf Müller (Mech. Eng., Virginia Tech, Blacksburg, VA)

Horseshoe bats (Rhinolophidae) have dynamic biosonar systems with interfaces for ultrasonic emission/reception that deform while diffracting the outgoing/incoming sound waves. This peripheral dynamics has been shown to enhance sensory information encoding. To investigate the properties and potential uses of these system features in the real world, a biomimetic sonar head is being developed with the goal of mimicking as much of the bats' peripheral dynamics as faithfully as possible. Further constraints on this development were suitability for integration with mobile platforms and support for interactive quantitative/in-depth experimentation. Flexible noseleaf and pinna shapes along with an actuation system have been designed to mimic the static geometry of the respective structures in bats as well as several of the animals' degrees of freedom in deforming them. The electrical subsystem has been designed to support high-quality ultrasound generation and recording. Generation of all source voltages and signal amplification have been consolidated into a single circuit board to reduce weight and complexity of the overall system. The frame has been altered to reduce weight and add modularity to the noseleaf and pinna elements. The software architecture is based on a back-end (Python, LabVIEW) combined with a flexible front-end (MATLAB) for experimental design and data analysis.

2:05

1pAB4. Beamwidth in bat biosonar and man-made sonar. Rolf Müller and Bryan D. Todd (Mech. Eng., Virginia Tech, ICTAS Life Sci. District (Mail Code 0917), Virginia Tech, Blacksburg, VA 24061, rolf.mueller@vt.edu)

Beamwidth measures how a sonar system distributes emitted energy or receiver sensitivity over angle. For man-made sonars, this is considered to be of critical importance, since a small beamwidth supports high-resolution imaging. For bat biosonar, the role of beamwidth is far less clear. In order to investigate how biosonar beamwidths could compare to its man-made peers, we have compiled beamwidth data for bats (based on numerical predictions) and man-made sonar systems (based on data sheets). An analysis of this data shows that bat biosonar and man-made sonars operate in very different regimes with respect to the ratio of characteristic dimension (i.e., aperture diameter for bats and array length for man-made sonars) to wavelength. Whereas the aperture diameters of bats ranged from falling below the employed wavelength to being about 10 times larger, man-made sonars were typically at least one order of magnitude larger than the employed wavelengths and ranged up to 1000 times larger. Furthermore, for any given characteristic-dimension-to-wavelength ratio, bats showed much more variability in beamwidth than man-made sonar. In fact, the variability in the bat beamwidths was found to be as large as that in a set of random horns, which indicates a role for other determining factors.

2:20

1pAB5. The problem of finding passageways in foliage with biomimetic sonar. Ruihao Wang, Xin Jin, Haosen Wang, and Rolf Müller (Mech. Eng., Virginia Tech, 1075 Life Sci. Cir, Blacksburg, VA 24061, ruihaow1@vt.edu)

Bats living in densely vegetated habitats must have the ability to find passageways in foliage. To explore the sensing problem posed by this navigation task, a sonar head that mimicked a bat's biosonar periphery and emitted a bat-like chirp was placed on a linear track to scan artificial foliages (plastic leaves resembling broad-leaf foliage) interrupted by gaps of controllable width. The first approach to detecting the gaps tested has been based on comparing echo energy levels received when facing a gap as opposed to a closed foliage. The ability to detect a gap in this way depends on the employed beamwidth and the distance to the foliage, i.e., the narrower the beam and closer the sonar head, the narrower a gap can be detected. However, narrower beams were also found to increase the variability in the energy of the returned echoes which could impact the ability to reliably distinguish the different levels of echo energy associated with closed foliage and gaps. Since bats tend to have wide beams due to their limited size relative to the employed wavelengths, it may be hypothesized that the animals have evolved alternative approaches to improve the ability of detecting narrow passageways at a distance.

2:35

1pAB6. Assessing continuous sensory information encoding by a biomimetic dynamic sonar emitter. Luhui Yang (Shandong University - Virginia Tech Int. Lab., Shandong Univ., 1075 Life Sci. Cir, Blacksburg, VA 24061, 913022794@qq.com) and Rolf Müller (Mech. Eng., Virginia Tech, Blacksburg, VA)

Bat species in the rhinolophid and hipposiderid families can deform the shapes of their noseleaves and pinnae when emitting/receiving ultrasound. Prior work has shown that these deformations enhance the encoding of sensory information. So far, quantifications of the impact on sensory information encoding have been based on small numbers of discrete shapes and clustering of head-related transfer functions into discrete alphabets. In the current work, an improved estimator for mutual information that is based on entropy estimates from k-nearest neighbor distances (Kraskov *et al*., 2004) was used to overcome these limitations. With this approach, mutual information was estimated based on individual head-related transfer functions along the entire duration of biosonar pulses that were represented by high sampling rates suitable for ultrasonic signals. Testing was carried out with a data set from a dynamic sonar emitter inspired by the noseleaf of Pratt's roundleaf bat (*Hipposideros pratti*). The results were found to be in good agreement with the cluster-based approach. The high temporal resolution employed showed that mutual information between beampatterns decreased very quickly within the pulse duration demonstrating that the noseleaf dynamics can provide the bats with the equivalent of several nearly independent views of the environment within a single pulse.

2:50–3:05 Break

3:05

1pAB7. Extensible detection and classification in Ishmael. David K. Mellinger, Curtis Lending, Sharon L. Nieukirk, Sara L. Heimlich (Coop. Inst. for Marine Resources Studies, Oregon State Univ., 2030 SE Marine Sci. Dr., Newport, OR 97365, David.Mellinger@oregonstate.edu), Julie N. Oswald, and Michael Oswald (Bio-Waves, Inc., Encinitas, CA)

Ishmael is a bioacoustic analysis software package that has offers automatic detection of animal calls via a number of user-configurable detection methods. Ishmael now offers three significant improvements. The first is that users can download pre-configured detectors from within Ishmael for several cetacean species, including several mysticetes and several odontocetes. Each such detector is characterized with performance-measurement curves (receiver operating characteristic, detection error tradeoff, precision-recall) for a test dataset. Information on the signal-to-noise ratio of test datasets is also provided, since performance information is nearly meaningless without some indication of call clarity. The second is that Ishmael is extensible via a MATLAB interface that allows users to write their own detection plug-ins. Ishmael sends sound data and associated metadata to the plug-in, which analyzes the data and sends back detections and classifications for Ishmael to further handle as it would handle detections from built-in detectors. The third is an interface to Real-time Odontocete Call Classification Algorithm (ROCCA): Ishmael detects clicks and whistles of odontocetes, measures various acoustic characteristics, and sends the resulting acoustic parameters to ROCCA. ROCCA classifies both individual vocalizations, and also groups successive vocalizations together into "encounters," which are then classified separately. [Work supported by LMR and ONR.]

3:20

1pAB8. A decade of marine mammal acoustical presence and habitat preference in the Bering Sea. Kerri Seger (School of Marine Sci. and Ocean Eng., Univ. of New Hampshire, 9331 Discovery Way, Apt. C, La Jolla, CA 92037, kseger@ucsd.edu) and Jennifer L. Miksis-Olds (School of Marine Sci. and Ocean Eng., Univ. of New Hampshire, Durham, NC)

As Arctic seas rapidly change with increased ocean temperatures and decreased sea ice extent, traditional marine mammal distributions may be altered, and non-traditional Arctic species may shift poleward. Extant and seasonal continental shelf odontocete species in the Arctic Ocean include sperm whales, killer whales, beluga whales, delphinids, harbour porpoises, and Dall's porpoises. Until recently, recording constraints limited higher sampling rates, preventing detection of many of these high frequency-producing species in the Arctic seas. Using one of the first long-term data sets to record clicks, buzzes, and whistles, multiple species have been detected and classified to explore shifting distributions in the Arctic Corridor. The data were then paired with environmental variables for generalized linear and additive modeling. Pacific white-sided dolphins were detected for a few years during minimal ice coverage. Dall's porpoises and Risso's dolphins may have been acoustically detected in areas farther north than previously documented in acoustical records. Of all the species, beluga whales are the most highly coupled to the ice edge. Adjusted habitat distributions, suggestions for newly present species to be aware of during visual surveys, and environmental model results will be reviewed in this presentation. [This work was funded by the Office of Naval Research.]

3:35

1pAB9. Binaural AEP audiograms in seven Beluga whales (*Delphinapterus leucas*) from the Okhotsk Sea population. Evgeniya Sysueva, Dmitry Nechaev, Vladimir Popov, Mikhail Tarakanov, and Alexander Supin (Inst. of Ecology and Evolution, 33 Leninskij prosp., Moscow 119071, Russia, popov.vl.vl@gmail.com)

Hearing thresholds were measured and audiograms were obtained in seven belugas, three males and four females, provisionally 2 to 7 years old. The measurements were performed using a transducer located 1 m in front of the head. The stimuli were tone pip trains of carrier frequencies ranging from 11.2 to 128 kHz with a pip rate of 1 kHz. Auditory evoked potentials (the rate following responses) were recorded from the head vertex. In majority of the subjects, audiograms were similar to the typical odontocete audiograms with the lowest thresholds (from 53.7 to 56.4 dB re 1 μPa) at mid-frequency range (from 32 to 64 kHz) and a sharp thresholds rise (up to 79.5 dB re 1 μPa) at high frequencies (90–128 kHz). One beluga (female, 6–7 years old) featured an asymmetric hearing loss within a frequency range from 22.5 to 54 kHz. The reason for the loss is the subject for not defined. The evoked potential audiograms should be included into base screening of odontocete subjects involved in any kind of hearing research. Work supported by the Russian Science Foundation (project # 17-74-20107), the Russian Foundation for Basic Research and The Russian Geographic Society.]

MONDAY AFTERNOON, 7 MAY 2018 GREENWAY A, 3:00 P.M. TO 4:40 P.M.

Session 1pAO

Acoustical Oceanography and Underwater Acoustics: Acoustics in Naturally Constrained Environments: Estuaries, Bays, Inlets, Fjords, and Rivers II

Andone C. Lavery, Cochair
Applied Ocean Physics and Engineering, Woods Hole Oceanographic Institution, 98 Water Street, MS 11, Bigelow 211, Woods Hole, MA 02536

D. Benjamin Reeder, Cochair
Oceanography, Naval Postgraduate School, 833 Dyer Rd., Bldg. 232, SP-311B, Monterey, CA 93943

Invited Papers

3:00

1pAO1. Progress on the use of autonomous underwater vehicles to study estuaries. Craig L. McNeil, Andrey Shcherbina (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, cmcneil@apl.washington.edu), Marcia J. Isakson, Autumn N. Kidwell (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), D. Benjamin Reeder (Dept. of Oceanogr., Naval Postgrad. School, Monterey, CA), Andone C. Lavery, and W Rockwell Geyer (Appl. Ocean Phys. and Eng., Woods Whole Oceanographic Inst., Woods Hole, MA)

Estuaries are a challenging environment to use acoustically navigated Autonomous Underwater Vehicles (AUVs) due to highly variable currents, relatively shallow and variable bathymetry, large buoyancy changes, suspended sediments, marine biota, and bubble plumes. The benefit to using AUVs is their ability to perform repeat automated surveys and targeted sampling of features using either remote control or on-board redirects. Our REMUS 100 AUVs are equipped with up/down looking ADCPs, CTDs, and optical backscatter sensors. Our AUVs use Long Base Line (LBL) underwater navigation, with up to four transponders, and were recently equipped to carry broadband hydrophones. We will share our experience operating these AUVs in several estuaries to characterize variability in optical and acoustical backscatter associated with estuarine features of interest (fronts, river plume, and the salt-wedge). We will discuss how these features negatively impact AUV sampling via, for example, degraded underwater communications and bottom tracking. We will also present some examples of concurrent sampling by AUVs and an advanced sonar (static and mobile) demonstrating the potential use of AUVs in estuarine research for 4D visualization of estuarine features of interest. [This work was supported by Office of Naval Research.]

3:20

1pAO2. Effects of tidal forcing in an estuarine environment on high-frequency acoustics. Marcia J. Isakson, Autumn N. Kidwell, and Mark Story (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78713, misakson@arlut.utexas.edu)

Tidal forcing in rivers and estuaries has a large impact on the use high-frequency acoustics for bathymetric mapping. Changes in the sound speed profile due to changing salinity and temperature conditions affect acoustic propagation, and bubbles, entrained in the tidal flow, cause high backscattering and attenuation. In this study, a forward-looking, high-frequency bathymetric sonar was used to observe estuarine dynamics near the mouth of the Connecticut River in June 2017. The sonar was deployed on a REMUS 600 AUV under the fresh water plume near the river mouth during flood tide, in the lower estuary during ebb tide and in the upper estuary during slack tide. Backscattering from subducted bubbles was observed which was dependent on the propagation path through the fresh or salt water. The bubble backscatter over a 200-m swath at the front of the fresh water plume allowed the structure of the plume to be observed from beneath the surface. These data are compared with other measurements on site, including X-Band radar and *in situ* salinity and temperature measurements. The data can be used to understand both acoustic propagation and estuarine dynamics in constrained environments. [Work sponsored by the Office of Naval Research.]

Contributed Papers

3:40

1pAO3. 3D underwater sound propagation in coastal environments. Ying-Tsong Lin (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Bigelow 213, MS#11, WHOI, Woods Hole, MA 02543, ytlin@whoi.edu), D. Benjamin Reeder (Naval Postgrad. School, Monterey, CA), and David K. Ralston (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

Underwater sound propagation in a coastal ocean can be influenced by a variety of physical oceanographic conditions that are unique in the coastal and estuarine environment, such as salt wedge fronts, river-plume induced nonlinear internal gravity waves, strong tidal currents, surf zone waves, etc. The confined boundaries in the area can also produce reflection, diffraction and even scattering of sound due to shoaling seafloor. The morphological dynamics in the area is complex and results in corrugated and channeled bathymetry, which may cause 3D ducting and trapping of sound. In this talk, we will present our numerical study of 3D underwater sound propagation in Long Island Sound, a tidal estuary of the Atlantic Ocean. The Unstructured Grid Finite Volume Community Ocean Model (FVCOM) will be utilized to simulate physical oceanographic processes and produce water-column environmental input to the sound propagation model. A bathymetric database from high-resolution multibeam surveys will also be used to incorporate the realistic seafloor depth in the model. The numerical model results will be analyzed to present the evidences of 3D sound propagation effects and investigate the causes.

3:55

1pAO4. Acoustic propagation in the Connecticut River at low-to-mid frequencies. D. Benjamin Reeder, John E. Joseph, and Tarry Rago (Oceanogr., Naval Postgrad. School, 833 Dyer Rd., Bldg. 232, SP-311B, Monterey, CA 93943, dbreeder@nps.edu)

An estuary is a constrained environment which often hosts a salt wedge, the stratification of which is a function of the tide's range and speed of advance, river discharge volumetric flow rate and river mouth morphology. A field experiment was carried out in the Connecticut River in June 2017, one goal of which was to investigate the low-to-mid-frequency acoustic propagation characteristics of the riverine salt wedge as well as the plume outside the river mouth. Linear frequency-modulated (LFM) acoustic signals in the 500–2000 Hz band were collected during several tidal cycles. Data-model comparisons demonstrate the degree to which this highly energetic environment impacts acoustic propagation; dominant mechanisms are sound speed stratification, boundary interaction, flow noise, and background noise.

4:10

1pAO5. High-frequency acoustic observations of a tidal cycle in the Connecticut River. Autumn N. Kidwell, Marcia J. Isakson, Mark Story, and Andrew J. Reiter (Appl. Res. Labs., Univ. of Texas, Appl. Res. Labs., University of Texas, 10000 Burnet Rd., Austin, TX 78758, akidwell@arlut.utexas.edu)

Rivers and estuaries can be highly energetic and present a challenging environment for sonar deployment and observation. To better understand this environment, a high-frequency, wide-sector forward-looking bathymetric sonar was used to observe tidal phenomena in the lower Connecticut River in June 2017. The stationary sonar system provided Eulerian measurements of the lower estuary for a complete tidal cycle, covering a large portion of the main channel. The system was able to map the observed region and differentiate between bathymetry and non-stationary phenomena in the water column. The data highlight the difference in acoustic propagation for the different phases of the tidal cycle. Doppler processing of the data shows the motion of various subsurface features in the river. These data are compared with other observations, including simultaneous *in situ* point measurements of salinity, temperature and turbidity from a REMUS 100 AUV. The data from this experiment can be used to improve the understanding of acoustic propagation in riverine and estuarine environments. [Work sponsored by the Office of Naval Research.]

4:25

1pAO6. Spectral quantification of a tidal plume from an autonomous underwater vehicle. Andone C. Lavery (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., 98 Water St., MS 11, Bigelow 211, Woods Hole, MA 02536, alavery@whoi.edu), Joseph T. Jurisa (Dept. of Civil and Environ. Eng., Portland State Univ., Portland, OR), and W. Rockwell Geyer (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

Buoyant plumes are ubiquitous in estuarine environments, yet their frontal structure and dynamics are not fully understood. Tidal plumes are dynamic over a broad range of temporal and spatial scales, and are characterized by strongly stratified turbulence and frontal instabilities. Observations from autonomous underwater vehicles (AUVs) with a suite of sensors, including temperature, salinity, and acoustic backscatter, provide a useful complement to ship-board sampling to characterize the frontal structure and dynamics at high resolution. Results are presented from an experiment in which a broadband acoustic backscattering system (160–410 kHz) was integrated onto a REMUS-100 AUV and used to sample the near-field ebb tide plume at the mouth of the Connecticut River. Scattering mechanisms are determined from the frequency-dependent acoustic spectra, and acoustic measurements of the frontal structure are compared to data collected with a towed CTD array. Data are also presented that illustrate the challenges associated with AUV operations in estuarine environments, including high currents, shallow and variable bathymetry, large buoyancy changes, and bubble entrainment at the plume front.

Session 1pBA

Biomedical Acoustics: Transcranial Focused Ultrasound for Targeted Brain Therapies

Emad S. Ebbini, Chair
Electrical and Computer Engineering, University of Minnesota, 200 Union St. SE, Minneapolis, MN 55455

Chair's Introduction—1:25

Invited Papers

1:30

1pBA1. Progress in the utilization of focused ultrasound for treatment of the brain. Kullervo Hynynen (Medical Biophys., Univ. of Toronto, Sunnybrook Health Sci. Ctr., Toronto, ON M4N 3M5, Canada, khynynen@sri.utoronto.ca)

Magnetic Resonance imaging (MRI) guided focused ultrasound (FUS) has been shown to be able to safely ablate deep tissues in the brain in clinical settings resulting in reduction in recovery time and complications. Pre-clinical work has demonstrated that FUS energy when combined with micro bubbles(MB) could be used also for image-guided drug delivery into tumours and brain. The preclinical studies have shown significant survival benefits when FUS + MBs has been used to open the blood-brain-barrier (BBB) for large molecular chemotherapy agents, antibodies, and natural killer cells. Effective and safe delivery of nanoparticles and viral vectors has also been shown. This talk will briefly review the physical and engineering aspects related to the clinical FUS + MB exposures. The first ongoing clinical trials investigating the feasibility and safety of FUS + MB enhanced BBB permeability will be reviewed. Finally, the next generation phased array technology and the ultimate clinical potential of the FUS + MBs in brain treatments will be discussed.

1:55

1pBA2. Real-time monitoring of transcranial focused ultrasound *in vivo*. Dalong Liu and Emad S. Ebbini (Elec. and Comput. Eng., Univ. of Minnesota, 200 Union St. SE, Minneapolis, MN 55455, ebbin001@umn.edu)

Transcranial focused ultrasound (tFUS) is being investigated in a variety of medical applications. Both thermal and mechanical bioeffects of tFUS are of interest, but localization of the tFUS therapeutic beam remains a major concern in the presence of beam distortion through the skull. We have developed a system for monitoring and delivery of tFUS using dual-mode ultrasound array (DMUA) transducers. The system is capable of delivering therapeutic tFUS in pulsed mode with imaging pulses interleaved at frame rates up to 500 fps utilizing the same array elements. In this way, the DMUA provides inherent registration between the imaging and therapy system thus providing feedback from the target region as well as tissue structures in the path of the beam. The beamformed DMUA echo data was used to image the thermal and mechanical tissue response to the tFUS beam with high spatial specificity. *In vivo* data from small-animal experiments have shown a temperature sensitivity of 0.2°C with high spatial specificity. Speckle tracking at up to 500 fps also allowed for tracking mechanical deformation produced by the tFUS beam in the presence of pulsations and other brain tissue motions. These results demonstrate the feasibility of characterizing both mechanical and thermal effects of subtherapeutic tFUS beams *in vivo*. The results are significant in burgeoning applications where the neuromodulatory response is induced by a combination of thermal and mechanical interactions.

Contributed Papers

2:20

1pBA3. Transcranial acoustic cavitation localization with ultrafast power cavitation imaging in non-human primates. Mark T. Burgess, Maria E. Karakatsani, Iason Apostolakis, and Elisa Konofagou (Biomedical Eng., Columbia Univ., 630 West 168th St., Physicians and Surgeons 19-418, New York City, NY 10032, mtb2153@columbia.edu)

Acoustic cavitation-guided blood-brain barrier (BBB) opening with focused ultrasound (FUS) and microbubbles is a promising technique for safe and controlled opening of the BBB. Passive cavitation imaging has the ability to monitor the spatial intensity of acoustic cavitation for targeting verification and treatment monitoring. However, isolating acoustic cavitation emissions from tissue and skull reflections is a major challenge. In this study, we perform transcranial passive cavitation imaging with a 1.5D imaging array (M5Sc-D, bandwidth: 1.5–4 MHz, GE Medical Systems) placed in the central opening of a 0.5 MHz FUS transducer (H204, Sonic Concepts) in non-human primates. Broadband FUS pulses were used along with synchronous transmit and receive sequences to perform delay and sum beamforming with absolute time delays. Image sets were acquired at ultrafast frame rates (>1000 frames per second) for calculation of mean intensity images, i.e., power cavitation images. Spatiotemporal clutter filtering of image sets was explored over a range of FUS peak negative pressures (0.05–0.75 MPa) to increase power cavitation image sensitivity. Future work will explore the use of this method for safe and controlled BBB opening based on power cavitation image feedback. [Work supported in part by NIH grants R01AG038961 and R01EB009041.]

2:35

1pBA4. Wideband transskull refocusing of ultrasound beams using dual-mode ultrasound arrays: *Ex vivo* results. Hasan Aldiabat (Elec. and Comput. Eng., Univ. of Minnesota, 200 Union St SE, Minneapolis, MN 55455, aldia001@umn.edu), Parker D. O'Brien (Elec. and Comput. Eng., Univ. of Minnesota, Richfield, MN), Dalong Liu, and Emad S. Ebbini (Elec. and Comput. Eng., Univ. of Minnesota, Minneapolis, MN)

Transcranial focused ultrasound (tFUS) is capable of providing subtherapeutic and therapeutic ablative treatments for a variety of brain disorders. A major challenge towards widespread use of tFUS-based therapies stems from the complexity of the skull that could result in severe loss of focusing gain. Using extensive hydrophone scan measurements in plain water as well as transskull, we have documented a range of tFUS beam distortions for a variety of target points and access angles. In this paper, we present quantitative measurements of tFUS distortions due to skull aberrations at different operating frequencies. In addition, refocusing results for a variety of target points at different frequencies within the transducer bandwidth are presented in terms of improvement in focusing gain. Dual-mode ultrasound array (DMUA) prototype (64 elements, concave with 40-mm radius of curvature) was used. Skull samples were extracted from animal subjects that have undergone tFUS treatments using the DMUA prototype were utilized. Experiments were performed at a set of 31 discrete frequencies in the range 2.0 MHz–5.0 MHz. A needle hydrophone was used to measure the pressure waveforms at the target locations. The element transmission efficiency varied as a function of frequency in a nonmonotonic manner with a range of 5–15 dB variation for the different target points. The array focusing gain also varied nonmonotonically suggesting the need for broadband refocusing.

2:50–3:10 Panel Discussion

3:10–3:25 Break

Invited Papers

3:25

1pBA5. Transcranial focused ultrasound with multielement arrays: Decreasing the number of transducers without compromising the focusing quality. Jean-Francois Aubry (Institut Langevin, CNRS, 17 rue Moreau, Paris 75012, France, jean-francois.aubry@espci.fr)

Transcranial ultrasonic brain therapy at frequencies higher than 500 kHz requires adaptive focusing to compensate for the aberrations induced by the skull bone. This can be achieved by using multi-element arrays driven by a dedicated electronics. A growing number of elements was used to improve the focusing: 64 elements in 2000 [1], 300 in 2003 [2], 1024 in 2013 [3], and more to come. We will present some of the salient results obtained pre-clinically and clinically with such multi-element transcranial devices. Nevertheless, we will show that comparable transcranial focusing can be achieved with a novel approach in rupture with the current trend. It consists in a single-element transducer covered with a 3D silicone acoustic lens of variable and controlled thickness. Similar lenses have been introduced in the past to perform multiple focusing patterns in homogenous media [4] but recent 3d printing capabilities make 3D lenses a feasible option for transcranial adaptive focusing. We will show that the use of a lenses increases the acoustic intensity by 97 $\pm$ 56% on average, on three human skulls. [1] G. Clement *et al.*, "A hemisphere array for non-invasive ultrasound brain therapy and surgery," Phys. Med. Biol. 2000. [2] M. Pernot *et al.*, "High power transcranial beam steering for ultrasonic brain therapy," Phys. Med. Biol. 2003. [3] N. Lipsman *et al.*, "MR-guided focused ultrasound thalamotomy for essential tremor: a proof-of-concept study," *The Lancet Neurology* 2013. [4] T. Fjield *et al.*, "Low-profile lenses for ultrasound surgery," Phys. Med. Biol. 1999.

3:50

1pBA6. Focused ultrasound for modulation of the central and peripheral nervous system. Elisa Konofagou (Columbia Univ., 1210 Amsterdam Ave., ET351, New York, NY 10027, ek2191@columbia.edu)

Noninvasive neuromodulation has been the preferred option of neurological treatment but noninvasive approaches fall short when it comes to depth penetration. Ultrasound modulation has been shown feasible in several species including humans both *in vitro* and *in vivo*. In this paper, an overview of our group's ultrasound neuromodulation in both the central (CNS) and the peripheral (PNS) nervous systems will be provided. In CNS, both motor- and sensory-related responses have been elicited in mice *in vivo* both in ipsilateral and contralateral limbs and pupils, respectively. The success are was highly correlated with the applied intensity and pressure in both the limb movement and ocular changes. The brain regions targeted were the somatosensory and visual cortex for the limb movement and the superior colliculus and locus coeruleus for the pupil dilation. In PNS, stimulation and inhibition of the sciatic nerve with FUS was elicited at different ultrasound parameters *in vivo*. Displacement of the nerve highly correlated with the elicited motor response. The success rate also correlated with higher intensities. Histology confirmed safety while temperature elevation was more correlated with inhibitory responses.

4:15

1pBA7. Intranasal administration of temozolomide combined with focused ultrasound to enhance the survival of mice with glioma (A Pilot Study). Dezhuang Ye (Mech. Eng. and Material Sci., Washington Univ. in St. Louis, 1155 Claytonia Terrace #2N, 2N, Saint Louissaint Louis, MO 63117, dezhuang.ye@wustl.edu), Yimei Yue, Lifei Zhu, Hong Chen (Biomedical Eng., Wustl, St. Louis, MO),

Intranasal route provides therapeutics direct access to the brain through the nose-to-brain pathway, bypassing the BBB and minimizing systemic exposure. However, nasal delivery is limited by its low delivery efficiency and non-localized delivery. Focused ultrasound-enhanced intranasal (FUSIN) delivery is a new technique for noninvasive and localized delivery of therapeutic agents to the brain. Previous studies have shown its feasibility in the delivery of dextrans and brain-derived neurotrophic factor to the brain. The goal of this study was to demonstrate the application of FUSIN for the delivery of temozolomide (TMZ), a first-line drug for treating glioma tumors, for the treatment of glioblastoma in an orthotopic mouse model. U87 glioblastoma cells were trancranially implanted to nude mice. The tumor mice were divided into four groups: (1) FUSIN + TMZ (n = 7); (2) IN TMZ (n = 6); (3) Oral TMZ (n = 6); and (4) IN vehicle (n = 6), and were treated once each week for 4 weeks. Mice were weighed once a week. A body weight loss reaching 20% was selected as the endpoint criteria for the study. Mice in FUSIN group showed statistically significant higher survival rate when compared with mice in all other groups. This pilot study suggested that FUSIN can potentially be an effective technique for the treatment of brain diseases.

4:30

1pBA8. Broadband transskull multi-focus wavefront synthesis. Parker D. O'Brien (Elec. and Comput. Eng., Univ. of Minnesota–Twin Cities, 7710 Penn Ave., S, Apt. B318, Richfield, MN 55423, obrie844@umn.edu), Dalong Liu, and Emad S. Ebbini (Elec. and Comput. Eng., Univ. of Minnesota–Twin Cities, Minneapolis, MN)

Multiple-focus pattern synthesis using ultrasound arrays has been previously demonstrated. Furthermore, an optimal synthesis method was shown to provide well-behaved solutions. This method creates predefined focus patterns at a single frequency for an ultrasonic array operating in continuous wave. In this paper, we extend the approach to the broadband case using orthogonal frequency-division multiplexing (OFDM). A broadband multiple-focus pattern is formed by summing the results of the synthesis algorithm at a set of discrete frequencies within the transducer bandwidth. The OFDM approach allows for achieving a predictable focusing gain at the target point(s) due to the waveform orthogonality. Outside the target region, the different frequency components can be expected to provide a larger degree of destructive interference. Hydrophone scans of multiple-focus patterns in plain water as well as transskull were performed using a phased array with a multichannel arbitrary waveform generator. The array has a 3.5 MHz center frequency with a 6-dB bandwidth in the range 2.2 MHz–4.8 MHz. The skull samples were obtained from 300 to 350 g rats that were used in other *in vivo* experiments. The results show greater focus resolution than for any single frequency pattern while reducing the interference patterns outside the target region.

4:45

1pBA9. Acoustical experimental design for ultrasonic neurostimulation on rodents. Shane W. Lani (Johns Hopkins Appl. Phys. Lab, 1454 Catherine St., Decatur, GA 30030, lani.shane@gatech.edu), Marina Congedo, Grace Hwang, and Allan Rosenberg (Johns Hopkins Appl. Phys. Lab, Laurel, MD)

While the mechanisms of ultrasonic neurostimulation are still unknown, this method has been successfully shown on rodents from a variety of research groups. While the results are compelling, there has been limited work in characterizing the acoustic fields that are generated and ultimately delivered to the rodent brain which is important in order to determine the underlying neurostimulation mechanism. This presentation details some of the acoustical experimental design processes that are being undertaken at JHU/APL as well as our current progress in the rodent ultrasonic neurostimulation. The experimental design utilizes a complementary process of both simulation and experimentation verification to determine the expected delivered acoustic wave field. The simulation encompasses the full propagation path from the single element focused transducer, through an acoustic waveguide and skull, to the treatment location. This work uses a focused transducer with a center frequency of 0.5 MHz with several different waveguides that are characterized in our calibration tank with regards to intensity over space. Each of these waveguides is modeled and compared relative to the focal spot size and intensity delivered to the focal spot. These measurements are used to select an optimal waveguide which is used in the model to determine the expected performance of the full system. The presented results compare the performance of the waveguides and their impact on ultrasonic delivery to the rodent brain.

Session 1pID

Interdisciplinary: Introductions to Technical Committees

Jonathan R. Weber, Cochair
Durham School of Architectural Engineering & Construction, University of Nebraska-Lincoln, 1110 S. 67th St., Omaha, NE 68182-0816

Tao Sun, Cochair
Radiology, Brigham and Women's Hospital; Harvard Medical School; Tufts University, 221 Longwood Avenue, EBRC 514, Focused Ultrasound Laboratory, Boston, MA 02115

Vahid Naderyan, Cochair
Physics/National Center for Physical Acoustics, University of Mississippi, NCPA, 1 Coliseum Drive, University, MS 38677

Invited Papers

1:00

1pID1. The Technical Committee on Musical Acoustics: Diverse membership with a common interest. James P. Cottingham (Phys., Coe College, 1220 First Ave. NE, Cedar Rapids, IA 52402, jcotting@coe.edu)

The Technical Committee on Musical Acoustics is concerned with the application of science and technology to the field of music. Topics of current interest include the physics of musical sound production, music perception and cognition, and analysis and synthesis of musical sounds. TCMU is an interdisciplinary committee with ties to architectural, psychological and physiological, signal processing, speech, physical, and structural acoustics. Members of the committee represent a variety of backgrounds and interests, yet are united by a common interest in understanding all aspects of the science of music.

1:10

1pID2. An introduction to the Engineering Acoustics Technical Committee. Michael V. Scanlon (Army Res. Lab., 2800 Powder Mill Rd., Adelphi, MD 20783-1197, michael.v.scanlon2.civ@mail.mil)

The Engineering Acoustics Technical Committee's scope encompasses the theory and practice of creating tools for investigating acoustical phenomena and applying knowledge of acoustics to practical utility. This includes the design and modeling of acoustical and vibrational transducers, arrays, and transduction systems in all media and frequency ranges; instrumentation, metrology, and calibration; measurement and computational techniques as they relate to acoustical phenomena and their utility; and the engineering of materials and devices. The breadth and scope of EATC's mission is an enabler to the other committees and to the interests of every member.

1:20

1pID3. Introduction to the Technical Committee on Physical Acoustics. Veerle M. Keppens (Dept. Mater. Sci. and Eng., Univ. of Tennessee, Knoxville, TN 37996, vkeppens@utk.edu)

The Technical Committee on Physical Acoustics (TCPA) of the Acoustical Society of America (ASA) is home to scientists and engineers with an interest in the underlying physics of acoustical phenomena and/or use acoustic waves to study the physical properties of matter. It is one of the broader communities within the Acoustical Society, as the research areas and methods cover the entire frequency range from infrasound to ultrasound, while studying the interaction of these sound waves in all three states of matter: liquids, solids, and gasses. TCPA currently has about 575 members with a primary interest in physical acoustics with 75–125 typically attending open TCPA meetings.

1:30

1pID4. An introduction to research topics in underwater acoustics. Jason D. Sagers (Environ. Sci. Lab., Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, sagers@arlut.utexas.edu)

The Acoustical Society of America Technical Committee on Underwater Acoustics (UW) investigates sound wave phenomena in marine environments, including oceans, lakes, and rivers. Research interests span a broad spectrum from the measurement and modeling of acoustic propagation and scattering, to the detection and characterization of underwater sound, to signal processing algorithms and statistics. This diverse technical committee also shares interests with Animal Bioacoustics (AB), Acoustical Oceanography (AO), and Signal Processing (SP). This talk will highlight a few past and present research topics in underwater acoustics, emphasizing the important role of sound as a tool in subsea research and exploration.

1:40

1pID5. An introduction to the acoustical oceanography technical committee. John A. Colosi (Dept. of Oceanogr., Naval Postgrad. School, 833 Dyer Rd., Monterey, CA 93943, jacolosi@nps.edu)

As is evidenced by the complex acoustic physiology and sophisticated auditory processing capability of marine organisms, it is clear that acoustics is a critical modality for interpreting the ocean environment. The acoustical oceanography (AO) technical committee (TC) seeks to foster a broad range of work in pure and applied acoustics with an aim towards gaining new fundamental understanding of physical, biological, geophysical, and chemical processes in the ocean and at its boundaries. With this focus, the AO TC is necessarily strongly interdisciplinary having close ties to other TCs such as animal bioacoustics, physical acoustics, signal processing, and underwater acoustics. This talk will describe many of the break-through discoveries that have been made in our field and point to inspiring present and future work.

1:50

1pID6. Overview of animal bioacoustics. Christine Erbe (Ctr. for Marine Sci. & Technol., Curtin Univ., Kent St., Bentley, WA 6102, Australia, c.erbe@curtin.edu.au)

Research in Animal Bioacoustics includes (1) animal sounds, communication, biosonar, and associated behavior; (2) sound production anatomy and neurophysiology; (3) auditory capacities and mechanisms, anatomy, and neurophysiology; (4) acoustic phylogeny, ontogeny, and cognition; (5) acoustic ecology, acoustic characterization of habitats, and effects of sound on animals; (6) passive acoustic tools and methods, hardware and software, for detection, classification, localization, tracking, density estimation, and behavior monitoring; and (7) active acoustic tools and methods, including animal-tracking sonars and echosounders, acoustic tags, pingers, deterrent devices, etc. Study species include birds, terrestrial mammals (in particular, bats), marine mammals, amphibians, reptiles, fishes, insects, and crustaceans. Researchers have diverse backgrounds ranging from acoustics, engineering, mathematics, and physics, to biology and zoology. We pool our expertise in this truly interdisciplinary field of Animal Bioacoustics.

2:00

1pID7. Biomedical acoustics: From diagnostic imaging to treating brain disorders. Subha Maruvada (U.S. Food and Drug Administration, 10903 New Hampshire Ave., Bldg. WO 62-2222, Silver Spring, MD 20993, subha.maruvada@fda.hhs.gov)

The Technical Committee on Biomedical Acoustics (BATC) is one of the most diverse groups in the Acoustical Society of America. BATC is comprised of scientists and engineers who study a wide range of biomedical applications using ultrasound from imaging fetuses to ablating fibroids and treating brain disorders. Diagnostic ultrasound has expanded to include exciting new techniques such as shear wave elastography and opto-acoustic imaging. Therapeutic ultrasound applications include physiotherapy, lithotripsy, as well as the treatment of tumors via thermal ablation, i.e., HITU, or mechanical ablation, i.e., histotripsy. The most recent developments in therapeutic ultrasound involve neuromodulation and transcranial magnetic resonance guided focused ultrasound for the treatment of various brain disorders, including essential tremor, neuropathic pain, and Parkinson's disease. The ability to temporarily open the blood-brain barrier using focused ultrasound has been demonstrated in animals and, more recently, in humans. Opening of the blood-brain barrier could allow the transmission of drugs to treat brain diseases. Diagnostic and therapeutic ultrasound applications continue to advance the field of medicine and are integral to the advancement of both diagnosis and treatment of debilitating diseases.

2:10

1pID8. Introduction to the structural acoustics and vibration technical committee. Robert M. Koch (Chief Technol. Office, Naval Undersea Warfare Ctr., Code 1176 Howell St., Bldg. 1346/4, Code 01CTO, Newport, RI 02841-1708, Robert.M.Koch@navy.mil)

The Structural Acoustics and Vibration (SAV) Technical Committee is a broad group of scientists and engineers sharing a common interest and shared stewardship in the health and advancement of the SAV technical discipline. Structural acoustics may be defined as a multidisciplinary coupled field of physics usually referring to the characterization of either (1) the sound power emitted by a vibrating structure subjected to external dynamic excitation or (2) the vibrational response of structures excited by incident sound fields or fluid excitation. While the fundamental scientific study of the underlying SAV physics is certainly important to SAV practitioners, there is also interest on the practical application of the prediction, control, and potential reduction of the vibroacoustic response of given structural acoustic systems. This paper first provides a fundamental definition of the underlying physics behind the SAV technical discipline. The many categories and subdivisions within the general SAV area are then addressed, along with illustrations of the broad array of scientific and engineering real-world applications in the field. Following that, examples of interesting career options as well as exciting new research areas in SAV are presented. Finally, the paper concludes with information regarding the makeup, functioning, and administrative philosophy of the SAVTC.

2:20

1pID9. Introduction to the Technical Committee on Noise. William J. Murphy (Hearing Loss Prevention Team, Centers for Disease Control and Prevention, National Inst. for Occupational Safety and Health, 1090 Tusculum Ave., Mailstop C-27, Cincinnati, OH 45226-1998, wjm4@cdc.gov)

Noise presents both a health hazard and a societal hazard. Health hazards due to noise have been around since the development of metal forging and forming. As modern society has developed, community noise has affected where and how we interact with the soundscapes where we live. The United States Environmental Protection Agency, Office of Noise Abatement and Control was responsible for noise regulations that rate the production and reduction of noise until it was defunded. Federal agencies such as the National Institute for Occupational Safety and Health, the Occupational Safety and Health Administration, and the Mine Safety and Health Administration have conducted research to develop regulations for occupational noise exposures. Recent community noise issues have focused on windfarms and the low-frequency noise that nearby residents might experience. This presentation will cover a wide range of noise-related topics and highlight efforts in standards for noise assessment and noise control.

2:30

1pID10. Architectural acoustics: From concert halls to classrooms. Ronald Freiheit (Wenger Corp., 555 Park Dr., Owatonna, MN 55060, ron.freiheit@wengercorp.com) and Matthew T. Neal (Graduate Program in Acoust., Penn State Univ., University Park, PA)

Architectural Acoustics (AA) finds its application in and around the built environment and, in doing so, touches and interacts with many different disciplines. Within ASA, the TCAA consistently hosts joint special sessions with the noise, musical acoustics, psychological and physiological acoustics, and signal processing technical committees. Apart from its involvement with special sessions, many unique subcommittees bring added diversity within architectural acoustics. These include subcommittees on classroom acoustics, speech privacy, green building acoustics, healthcare acoustics, building performance standards, and the concert hall research group (CHRG). The makeup of members in the AA group is quite varied. It includes those involved in education and research to those in consulting as well as those in industries that supply products to this segment. Along with published research which is typical of most technical committees, a large segment of the TCAA's accomplishments are based on the projects produced by consultants or the products provided by industries. The AA group has consistently been one of the larger groups of active participants. There is a unique balance between research, application, and industry support for this segment of acoustics, making the TCAA a great place to see new perspectives and make new connections with colleagues, friends, and mentors.

2:40

1pID11. Psychological and physiological acoustics: Responses to sounds in biological systems. Frederick J. Gallun (VA RR&D National Ctr. for Rehabilitative Auditory Res., VA Portland Health Care System, 3710 SW US Veterans Hospital Rd., Portland, OR 97239, Frederick.Gallun@va.gov)

The Technical Committee on Psychological and Physiological (P&P) acoustics is organized around the investigation and dissemination of information about responses to sound in humans and other species. Areas of interest include, but are not limited to the following topics: (1) perception and perceptual organization of simple and complex sounds, including speech; (2) anatomy and function of the auditory pathways, including all physical and biological responses to auditory stimulation; (3) hearing disorders, hearing loss, and auditory prostheses; (4) vibrotactile and vestibular sensation, and the interaction of hearing with other sensory modalities; (5) developmental, aging, learning, and plasticity effects in auditory function; and (5) theories and models of auditory processes. Examples will be given of hot topics in these areas as well as areas of overlap with other TCs in the ASA.

2:50

1pID12. It is like ESP without the E: An Introduction to the Speech Communication Technical Committee. Benjamin Munson (Speech-Language-Hearing Sci., Univ. of Minnesota, 115 Shevlin Hall, Minneapolis, MN, munso005@umn.edu)

Speech communication (SC) is a quintessentially human behavior. Scholars in the SC Technical Committee conduct research on three broad areas: the production of speech, the perception of speech, and the machine processing of speech. We study people across the globe and across the entire lifespan. Our work includes people with and without speech, language, and hearing impairments. This brief talk will highlight three recent projects by SC TC members that illustrate the breadth of our work. We will review research on how speech style affects memory for speech, how ultrasound can be used to measure tongue movement during speech, and how speech synthesis can be used to estimate articulatory movements from acoustic records of speech. We hope that this presentation will encourage students and new members from our society's many technical committees to learn more about how our work in the SC TC complements research and scholarship in many different fields of acoustics.

3:00

1pID13. Signal processing everywhere you look. Zoi-Heleni Michalopoulou (Mathematical Sci., New Jersey Inst. of Technol., 323 ML King Blvd., Newark, NJ 07102, michalop@njit.edu)

Whether designing an auditorium, interpreting a marine mammal song, analyzing a sonar signal, or extracting information from ambient noise, signal processing is there. With active research in the field, signal processing provides a multitude of tools that are critical for solutions of complex acoustic problems. With time-series analysis, time-frequency representations, higher-order statistics, model-based and Bayesian processing, compressed sensing, and many other approaches, acoustic signal processing brings together physics, mathematics, and statistics, theory and computation, for the better understanding of acoustical phenomena. The diversity of the field is evidenced by the fact that members of the Technical Committee on Signal Processing in Acoustics also have membership in a multitude of other technical committees: underwater acoustics, acoustical oceanography, animal bioacoustics, and architectural acoustics, to name a few. Additionally, many special sessions at Acoustical Society Meetings are cosponsored by the Technical Committee on Signal Processing in Acoustics, proof that signal processing is everywhere you look. It is this diverse field that we explore in this talk, providing an overview of current research in a number of areas.

Session 1pNS

Noise, Psychological and Physiological Acoustics, Signal Processing in Acoustics, Speech Communication, and Musical Acoustics: Open Source Audio Processing Platforms

Odile Clavier, Cochair
Creare, Inc., 16 Great Hollow Rd., Hanover, NH 03755

William J. Murphy, Cochair
Hearing Loss Prevention Team, Centers for Disease Control and Prevention, National Institute for Occupational Safety and Health, 1090 Tusculum Ave., Mailstop C-27, Cincinnati, OH 45226-1998

Chair's Introduction—1:00

Invited Papers

1:05

1pNS1. SignalMaster—An open hardware and software system for speech and signal processing. Patrick Davies and Rafael E. Delgado (Intelligent Hearing Systems Corp, 6860 S.W. 81st St., Miami, FL 33143, pdavies@ihsys.com)

A portable speech and signal processing system was developed using a TMS320C6748 Digital Signal Processor (DSP) with two input and output channels with 32 bit Analog-to-Digital (AD) and Digital-to-Analog (DA) converters. Lithium-ion batteries provide up to 10 hours of freestanding operation. The system may also be used while connected to a personal computer (PC). Special boot loading code and an interface protocol was developed in order to allow communications and standardize the transfer of instructions and data between the DSP hardware and PC through a Universal Serial Bus (USB) interface cable. In addition, a PC-based user application interface high-level language dynamic link library was also developed to enable development of a wide range of applications. The library provides functions for user developed applications to communicate with the hardware and be able to upload C language programs into the DSP system. This allows for different programs to be uploaded in order to reconfigure the processing algorithms being executed for any experiment at any time. DSP and PC-based software source code examples provide users with the ability to develop their own user specific applications. Examples were developed to demonstrate the implementation of hearing aid amplification, filtering, and noise cancellation in real time.

1:25

1pNS2. Audiologic evaluation of the tympan open source hearing aid. Joshua M. Alexander (Speech, Lang., and Hearing Sci., Purdue Univ., Heavilon Hall, 500 Oval Dr., West Lafayette, IN 47907, alexan14@purdue.edu), Odile Clavier, and William Audette (Creare LLC, Hanover, NH)

The "Tympan" open source hearing aid (Creare LCC) was developed in response to needs identified by the 2014 Open Speech Signal Processing Platform Workshop hosted by NIH. The Tympan Phase I prototype is a neck-worn device with external microphones and headphones implemented on a Teensy 3.6 development board and Arduino software platform. Custom-designed electronics for audio input acquisition, digital signal processing, power management, and wireless communication allows expert users to create new algorithms directly in firmware. Operating in parallel, a custom software interface allows "occasional" users with the necessary support to modify easily parameters of existing algorithms. A variety of electroacoustic analyses, including throughput delay, amplitude compression input/output curves and time constants, harmonic distortion, internal noise, frequency response, and other real-ear measurements were performed on the Phase I prototype using commercial hearing aid analyzers and compared to today's premium hearing aid products. Overall, the results indicate that the Tympan Phase I prototype performs similarly to an actual hearing aid on these metrics. Results from an early Phase II prototype will also be reported. [Work supported by NIH NIDCD Grant #1R44DC015445-01 to Creare LLC.]

1:45

1pNS3. Tools for assessing efficacy of hearing loss compensation. Krishna Chaithanya Vastare (Dept. of Elec. and Comput. Eng., Univ. of California, San Diego, Atkinson Hall, 9500 Gilman Dr., La Jolla, CA 92093, kvastare@ucsd.edu), Sergio Luna (Dept. of Mathematics, Univ. of California, San Diego, La Jolla, CA), Tamara Zubaity (Dept. of Cognit. Sci., Univ. of California, San Diego, La Jolla, CA), Ganz Chockalingam (Qualcomm Inst., Univ. of California, San Diego, La Jolla, CA), and Harinath Garudadri (Dept. of Elec. and Comput. Eng., Univ. of California, San Diego, La Jolla, CA)

We are developing a real-time, wearable, Open-source Speech-Processing platform (OSP) that can be configured by audiologists and Hearing Aid (HA) researchers for lab and field studies. This contribution describes OSP tools to (i) control the HA state, such as amplification parameters in each subband, (ii) provide stimuli to the user, and (iii) get feedback from the user. The system is based on a web

server with interfaces to the HA state, the environment state (e.g., background noise characteristics, reverberation conditions, GPS location, etc.) and the user state (as inferred from ecological momentary assessments). We describe Application Programmer's Interfaces (APIs) used in Hypertext Markup Language (HTML) and server side scripting language (e.g., PHP) to create web applications aimed at assessing efficacy of various Hearing Loss (HL) compensation approaches. HTML and PHP are versatile tools and lot easier to develop functional skills compared with software tools used in the OSP. We provide example scripts to run tests such as A/B comparison, American version of the four alternative auditory feature test (AFAAF), etc. Audiologists and hearing scientists can modify these example scripts to create studies aimed at understanding the interactions between HA state, environment and user state and discover novel HL compensation approaches.

2:05

1pNS4. An open computational platform for low-latency real-time audio signal processing using field programmable gate arrays.
Ross K. Snider, Christopher N. Casebeer (ECE Dept., Montana State Univ., 610 Cobleigh Hall, Bozeman, MT 59717, ross.snider@montana.edu), and Raymond J. Weber (Flat Earth Inc., Bozeman, MT)

Field Programmable Gate Arrays (FPGAs) provide flexible computational architectures that are ideal for digital signal processing (DSP). With support of a NIH/NIDCD SBIR grant, we are developing an open FPGA platform for the speech, hearing, and acoustics research communities. The advantage of using FPGAs in a computational platform over conventional CPU approaches is the ability to implement low-latency high-performance signal processing with deterministic latencies. The hardware portion of the platform includes an audio codec and an Intel System-on-Chip (SoC) FPGA that contains ARM CPUs alongside the computational fabric that allows custom data plane designs. Development uses Mathwork's Simulink that allows exploration and simulation of signal processing algorithms. Once a Simulink model has been developed to implement audio processing in either the time domain or frequency domain, VHDL code can be generated that implements the desired signal processing. The VHDL code is then implemented in the FPGA computational fabric where the FPGA functions as a real-time signal processor. We are currently soliciting ideas/feedback from the speech, hearing, and acoustic communities as to what features they would like to see in the next iteration of the open FPGA-based computational platform. [NIH/NIDCD R44DC015443: www.openspeechtools.com]

2:25

1pNS5. An open source noise dosimeter for evaluating exposure metrics. Christopher J. Smalt, Lawrence Thul, Shakti K. Davis, and Paul Calamia (MIT Lincoln Lab., 244 Wood St, Lexington, MA 02420, Christopher.Smalt@ll.mit.edu)

Noise dosimeters can be effective tools for measuring individual risk to sound exposure, especially in free-moving environments where sound pressure levels can vary significantly as a function of location and time. Commercial noise dosimeters typically compute noise exposure as an average A-weighted energy over an 8-hour work day (LAeq8hr), and health guidelines suggest a maximum exposure of 85 dB or 90 dB to limit noise-induced hearing injuries. However, recent studies have suggested that conventional noise exposure metrics may not be satisfactory and may not predict susceptibility to synaptic damage in the cochlea nor protect suprathreshold hearing. There is also evidence to suggest that the LAeq may under-predict health risk in complex noise environments, where there is both continuous and impulse noise, and over-predict health risk for long-duration impulse noise. This suggests a need for improved dosimeters with the ability to incorporate new noise metrics. Here we utilize a low-cost open source audio platform to implement standard and experimental noise exposure metrics in an open-source microcontroller programming language. A Bluetooth connection also enables real-time reporting of noise exposures to a smartphone. This system has a significant advantage over using persistent acoustic recordings which have privacy issues and may capture conversations. We envision that this system will be used to further noise exposure research and help evaluate new metrics of auditory health risk. [Work supported by ONR.]

Contributed Paper

2:45

1pNS6. Recent developments in the NASA auralization framework.
Brian C. Tuttle (Analytical Mech. Assoc., Inc., M.S. 463, NASA Langley Res. Ctr., Hampton, VA 23681-2199, btuttle1@gmail.com), Aric R. Aumann (Sci. Applications Int. Corp., Inc., Hampton, VA), and Stephen A. Rizzi (AeroAcoust. Branch, NASA Langley Res. Ctr., Hampton, VA)

The NASA Auralization Framework (NAF) is an open software architecture that facilitates the conversion of numerical noise data, specifically aircraft flyover noise, into an audible pressure-time history that may be further analyzed or used as stimuli in psychoacoustic studies. The framework provides a set of libraries for management of the auditory scene and propagation paths, synthesis of simple sources, and a basic environment including a uniform standard atmosphere and hard flat ground. Advanced capabilities may be added in the form of dynamic-link library plugins either developed by the user or provided by NASA as pre-compiled binaries. Recent additions to the NASA collection of advanced plugins include synthesis of periodic noise sources, enhanced ground reflection and impedance models, an atmospheric turbulence model, a post-processing module for computing sound quality metrics, and an interface with the Aircraft NOise Prediction Program 2 (ANOPP2). This presentation highlights these advanced capabilities as they relate to current NASA research into noise from existing and proposed aircraft designs.

3:00–3:15 Break

3:15

1pNS7. Open portable platform for hearing aid research. Caslav Pavlovic (BatAndCat Corp., 602 Hawthorne Ave., Palo Alto, CA 94301, chas@batandcat.com), Volker Hohmann, Hendrik Kayser (Univ. of Oldenburg and HörTech gGmbh, Oldenburg, Germany), Louis Wong (BatAndCat Corp., Cambell, CA), Tobias Herzke (Univ. of Oldenburg and HörTech gGmbh, Oldenburg, Germany), S. R. Prakash, zezhang Hou (BatAndCat Corp., Palo Alto, CA), and Paul Maanen (Univ. of Oldenburg and HörTech gGmbh, Oldenburgh, Germany)

The NIDCD has recently funded a number of projects to develop portable signal processing tools that enable real-time processing of the acoustic environment. The overarching goal is to provide a large group of researchers with the means to efficiently develop and evaluate, in collaborative multi-center environments, novel signal processing schemes, individualized fitting procedures, and technical solutions and services for hearing apparatus such as hearing aids and assistive listening devices. We report on the specific goals and results of two such projects. In one of them (R01DC015429), an open source software platform for real-time runtime environments is developed: The open Master Hearing Aid (openMHA). It provides an extendible set of algorithms for hearing aid signal processing and runs under Linux, Windows, and Mac operating systems on standard PC platforms and on small-scale ARM-based boards. An optimized version of openMHA is provided for the companion SBIR project (R44DC016247), which is a portable, rigid, versatile, and wearable platform featuring an ARM Cortex®-A8 processor. The resulting Portable Hearing Aid Community Platform consists of both hardware elements to provide the advanced desired functionality and software routines to provide for all the features that researchers may need to develop new algorithms.

3:35

1pNS8. Electroacoustic and behavioral evaluation of an open source audio processing platform. Daniel M. Rasetshwane, Judy G. Kopun, Ryan W. McCreery, Stephen T. Neely (Boys Town National Res. Hospital, 555 North 30th St, Omaha, NE 68131, daniel.rasetshwane@boystown.org), Marc A. Brennan (Univ. of Nebraska-Lincoln, Omaha, NE), William Audette, and Odile Clavier (Creare LLC, Hanover, NH)

Hearing-aid (HA) research in academia is limited by the lack of wearable, reconfigurable, and reprogrammable audio processing platforms. We are developing such a platform, called the Tympan, which includes a Teensy 3.6 processor board that leverages the Arduino development environment while providing powerful computational capabilities. Custom-designed electronics are used for audio control, power management, and wireless communication. User-friendly software can be used to test the relative benefits of amplification variants. Users can also implement new algorithms and modify parameters of algorithms. This study evaluated an eight-channel HA implemented on the Tympan, relative to a commercially available HA. Gain was prescribed using NAL-NL1. Eighteen participants with hearing loss were tested. Electroacoustic tests included ANSI 3.22-2009 standard clinical measurements, the HA speech perception index, and the HA speech quality index. Behavioral tests included recognition of CASPA words in quiet and AzBio sentences in noise. The Tympan HA performed similar to the commercially-available HA on all tasks. Efforts are ongoing to miniaturize the Tympan hardware, increase flexibility of the software, and add features such as feedback management. The collaborative development and open sharing of algorithms facilitated by the Tympan will lead to advances in HA research and in other audio signal processing fields.

3:55

1pNS9. Open-source speech-processing platforms: An application example. Arthur Boothroyd (Speech, Lang., Hearing Sci., San Diego State Univ., Campanile Dr., San Diego, CA 92182), Harinath Garudadri (Qualcomm Inst., Univ. of California, San Diego, La Jolla, CA), and Gregory Hobbs (Speech, Lang., Hearing Sci., San Diego State Univ., Campanile Dr., San Diego, CA 92182, ghobbs@sdsu.edu)

Speedy assessment of frequency-importance functions for a variety of speech materials is possible with an Open-source Speech-Processing (OSP) platform. One such platform is that developed at the University of California San Diego. It provides real-time, six-band, processing of microphone input from ear-level transducer assemblies. This platform was used to provide normative data on a new consonant-contrast test developed for use in studies of hearing-aid self-fitting. Performance of young normally hearing adults was measured under various conditions of low-pass filtering. Because the UCSD system includes two sets of filters whose center and crossover frequencies differ by half an octave, it was possible to test with cut-off frequencies at half-octave intervals from 250 through 8000 Hz. Group mean composite contrast score, after correction for chance, fell to around 50% at a (6 dB) cut-off frequency of 660 Hz. With a cut-off frequency of 1 kHz, group-mean contrast scores were in the region of 30 to 35% for place and clustering, 70% for continuance, and over 90% for voicing. Determining the frequency-importance functions of various speech materials and speech features is just one example of the potential value of OSPs in hearing-science and hearing-aid research.

4:15

1pNS10. Smartphone as a research platform for hearing study and hearing aid applications. Issa M. S. Panahi, Nasser Kehtarnavaz (Elec. Eng., Univ. of Texas at Dallas, EC33, 800 West Campbell Rd., Richardson, TX 75080, issa.panahi@utdallas.edu), and Linda Thibodeau (School of Behavioral and Brain Sci., Univ. of Texas at Dallas, Richardson, TX)

Smartphones are widely available and used by many people. Portability, processing power and useful features of smartphone, such as audio input/output and wire/wireless connectivity, offer unique opportunity to use smartphone as an open source research platform capable of implementing in real time many computationally intensive adaptive signal processing algorithms aimed at improving hearing study and hearing aid applications. In this paper, we present an overview of several adaptive algorithms developed for classifying the background noise, suppressing the noise, and enhancing the speech in different noisy environments. The proposed algorithms for hearing aid applications can run on existing smartphones such as the Android-based and iOS-based smartphones in real time. Objective and subjective test results are presented illustrating the performance of proposed algorithms running on smartphones in real time.

4:35–5:00 Panel Discussion

Session 1pPA

Physical Acoustics, Structural Acoustics and Vibration, Architectural Acoustics, and Underwater Acoustics: Novel Methods in Computational Acoustics II

D. Keith Wilson, Cochair
Cold Regions Research and Engineering Laboratory, U.S. Army Engineer Research and Development Center, 72 Lyme Rd., Hanover, NH 03768-1290

Amanda Hanford, Cochair
Pennsylvania State Univ., Applied Research Lab, PO Box 30 - MS 3230D, State College, PA 16804

Invited Papers

1:00

1pPA1. Target response in a focused, modulated sound field: Modeling and experiment. Timothy D. Daniel, Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., WSU, Pullman, WA 99164, timothy.daniel@email.wsu.edu), Ahmad T. Abawi (HLS, Res., La Jolla, CA), and Ivars P. Kirsteins (NUWC, Newport, RI)

In a previous work, we had shown that low frequency flexural modes of circular plates and cylinders could be excited using modulated radiation pressure generated by focused ultrasound [J. Acoust. Soc. Am. **139**, 2053 (2016)]. We recently conducted experiments probing how the response of these targets varied as a function of position in the focused ultrasound field. Surprisingly, the response of the circular plate was found to change sign as it was moved away from the source. Two different models were developed to analyze the target response and understand this sign change. A purely geometric model based on ray optics and a semi-physical optics model that uses the incident intensity calculated for the focused source, using a Rayleigh-Sommerfeld integral. Both geometric and semi-physical optics models predict a sign change for the plate response at approximately the correct distance from the source. The sign change is due to a factor in a mode projection that more strongly weights points farther from the center of the plate. The physical optics model was also applied to cylindrical targets. [Work supported by ONR.]

1:20

1pPA2. Potential of theoretical approach and T-matrix method for designing numerical acoustical tweezers toolbox. Zhixiong Gong (Dept. of Naval Architecture and Ocean Eng., Huazhong Univ. of Sci. and Technol., Pullman, Washington), Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA), and Wei Li (Dept. of Naval Architecture and Ocean Eng., Huazhong Univ. of Sci. and Technol., 1037 Luoyu Rd. Hongshan Dist., Wuhan, Hubei 430074, China, hustliw@hust.edu.cn)

Partial-wave series solution and T-matrix method (TMM) could be used for computations of radiation force and torque exerted on objects from an acoustic Bessel beam (ABB) with arbitrary order and location. The radiation force and torque in Bessel beams could be expressed in terms of incident and scattered coefficients which could be obtained analytically based on the multipole expansion method [Gong *et al*., J. Acoust. Soc. Am. **141**, EL574–578 (2017)].The investigation of radiation force and torque could be used to design numerical acoustical tweezers toolbox that may manipulate particles and cells as expected, showing potential in physical chemistry, biomedicine, and so on. This work claims the possibility and potential of the theoretical and numerical methods for acoustical tweezers toolbox which could trap, pull, push, and rotate targets of different shapes and components. Both the progressive and (quasi)standing Bessel beams will be considered. In addition, beams which could be expanded in terms of basis functions, for example, Gaussian beams, could also be included in this numerical toolbox. The numerical toolbox of acoustical tweezers could help to guide experimental set-ups. [Work supported by NSFC (W.L. and Z.G.), ONR (P.L.M.), and China Scholarship Council (Z.G.).]

1:40

1pPA3. A modal collocation algorithm for high-frequency propagation in discontinuous waveguides. Jerry H. Ginsberg (School of Mech. Eng., Georgia Inst. of Technol., 5661 Woodsong Dr., Dunwoody, GA 30338-2854, j.h.ginsberg@comcast.net)

A procedure for analyzing high-frequency propagation in adjoining constant width waveguides was presented previously [Ginsberg, JASA **134**, 4217 (2013)]. The parallel walls were taken to be rigid, and no results were provided. This presentation uses that development as the foundation for the analysis of waveguides having an arbitrary number of sections, each of whose walls have arbitrary local impedance. The analytical basis is description of the pressure field in terms of transverse pressure modes, each of which is associated with upstream and downstream propagation at wavenumbers that may be complex, depending on the wall properties. Pressure and particle velocity conditions at junctions of two segments are enforced exactly by collocation along the common cross section. Initiating the procedure at a passive termination leads to recursion relations for modal reflection and transmission coefficients at each junction, which are independent of the nature of the excitation. Satisfaction of the conditions at the end that is excited leads to another algorithm for all modal amplitudes. An example is used to compare the convergence properties of the collocation method to an analysis based on the orthogonality of the transverse modes.

2:00

1pPA4. A Green's function parabolic equation description of infrasound propagation in an inhomogeneous and moving atmosphere. Michael B. Muhlestein, Vladimir E. Ostashev, and D. K. Wilson (US Army ERDC-CRREL, 72 Lyme Rd., Hanover, NH 03755, Michael.B.Muhlestein@usace.army.mil)

Accurate atmospheric infrasound propagation models must account for variations in local sound speed and ambient flow. This paper describes and demonstrates a method to extend the Green's function parabolic equation (GFPE) to account for 3D high Mach number ambient flow in addition to an inhomogeneous atmosphere. Predictions of infrasonic propagation using the resulting GFPE model are then compared with predictions using other models.

2:20

1pPA5. Non-Markov character of sound propagation over relatively short ranges in the turbulent atmosphere. Vladimir E. Ostashev and D. Keith Wilson (U.S. Army Engineer Res. and Developmnet Ctr., 72 Lyme Rd., Hanover, NH 03755, vladimir.ostashev@colorado.edu)

The Markov approximation is widely used in wave propagation in random media, including sound propagation in the turbulent atmosphere. This approximation, which significantly simplifies formulations for the statistical moments of the propagating field, is valid if the propagation range is greater than the outer length scale of random inhomogeneities. For sound propagation in the turbulent atmosphere, this length scale can be as large as 400–500 m (23% of the height of the atmospheric boundary layer), indicating that the Markov approximation might not be applicable for relatively short ranges. In this paper, the statistical characteristics of a sound field propagating in the turbulent atmosphere, such as the correlation functions of the log-amplitude and phase fluctuations, the mean sound field, and the mutual coherence function are calculated, without using the Markov approximation. The new results are then compared with those obtained in the Markov approximation. The difference between the two formulations and the range of applicability of the Markov approximation are analyzed for different meteorological regimes of the atmospheric boundary layer characterized by the surface heat flux and the friction velocity.

2:40–2:55 Break

2:55

1pPA6. Exact wide-angle formulation of the Beilis-Tappert method. Kenneth E. Gilbert, Xiao Di (Physics/NCPA, Univ. of MS, P.O. box 35, 1703 Hunter Rd., University, MS 38871, kgilbert@olemiss.edu), and Carl R. Hart (U.S. Army Engineer Res. and Development Ctr., Cold Regions Res. and Eng. Lab., Hanover, NH)

The Beilis-Tappert method was originally developed for narrow-angle acoustic propagation under a rough sea surface [A. Beilis and F. D. Tappert, J. Acoust. Soc. Am. **66**, 811–826 (1979)]. The method has also been applied to narrow-angle propagation over irregular terrain for acoustic waves and radar. It is shown here that an exact wide-angle formulation of the Beilis-Tappert method can be derived simply by replacing $\partial/\partial z$ with $\partial/\partial z + ik_0 \tan\varphi$, where $k_0 = 2\pi/\lambda$, λ is the physical wavelength, φ is the slope angle, and $i = \sqrt{-1}$. The exact formulation makes clear that for large slope angles, much of the acoustic field does not propagate, but decays exponentially with range. Existing finite difference methods and all narrow-angle methods fail to properly account for the exponential decay with a range of the non-propagating components of the acoustic field. The exponential decay with this range is qualitatively explained in terms of rays and quantitatively explained in terms of waves. Properly accounting for the propagating and non-propagating components of the acoustic field is explained.

3:15

1pPA7. On the use of high-order time-domain impedance boundary conditions. Didier Dragna and Philippe Blanc-Benon (Ctr. Acoustique, Laboratoire de Mécanique des Fluides et d'Acoustique, Ecole Centrale de Lyon, 36 Ave. Guy de Collongue, Ecully 69134 Ecully Cedex, France, Philippe.blanc-benon@ec-lyon.fr)

Time-domain simulations are well-suited to study broadband sound propagation. One of the difficulties is the translation of frequency-dependent impedance boundary conditions that leads in the time domain to convolutions. Several methods have been proposed in the literature to have an efficient computation of convolutions. They are based on a multipole approximation of the impedance and on approximations of time-variations of the acoustic variables, allowing the convolution to be simply evaluated by recursive relations. Recently, Dragna *et al.* [J. Acoust. Soc. Am. **138**, 1030–1042 (2015)] have highlighted that recursive convolution methods are low-order methods and have introduced a new method, referred to as the auxiliary differential equation method, which allows one to compute convolutions by integrating in time ordinary differential equations. Its main advantage is that it preserved the order of accuracy. This approach was employed in Troian *et al.* [J. Sound Vib. **392**, 200–216 (2017)] to derive a time-domain impedance boundary condition (TDBIC). This paper aims at evaluating the accuracy and efficiency of this novel TDBIC and to compare it to those of recursive convolution approaches. The novel TDBIC is first presented. A one-dimensional test case, dealing with reflection of an acoustic pulse over an absorbing wall is then investigated. Examples of simulations in three-dimensional geometries in the context of duct acoustics or outdoor sound propagation are then shown.

3:35

1pPA8. Computation of dispersion curves of guided waves using discontinuous Galerkin finite element method: Application to functionally graded materials. Salah-eddine Hebaz, Farouk Benmeddour, Emmanuel Moulin, Jamal Assaad, Marc Duquennoy, and Mohammadi Ouaftouh (Univ. Valenciennes, CNRS, Univ. Lille, YNCREA, Centrale Lille, UMR 8520 - IEMN, DOAE, 12 Ave. de l'industrie, Appart. A213, Valenciennes 59300, France, Salah-Eddine.Hebaz@univ-valenciennes.fr)

The study of dispersive behavior of guided waves is of great interest in non-destructive testing and development of inspection systems. In this work, a semi-analytical numerical formulation for the computation of dispersion curves and mode shapes in plate-like structures is presented. The formulation is based on a discontinuous Galerkin Finite Element Method. The discretisation of the ordinary differential system in the through-thickness direction yields an eigenvalue problem. The resolution of the latter for a frequency range provides the dispersion curves. The study focuses on Lamb modes in functionally graded material plates with traction-free boundary conditions. The influence of the gradient variations is demonstrated. First, a multilayered approximation consisting of homogeneous laminate is used. Afterwards, the gradients of the parameters are added to the formulation. Numerical examples are presented and compared with those found in the literature. The results are free of spurious modes and show an excellent agreement for all cases, specially for continuously varying properties. The method is very efficient and numerically stable for small and large wave numbers. It was found that a high accuracy is obtained when high-order elements are used on a relatively coarse mesh.

3:50

1pPA9. Investigation of broadband attenuation with fractal diffusers as noise barrier top edge devices using indirect boundary element method. Prachee Priyadarshinee (Mech. Eng., National Univ. of Singapore, #21-256B, 38 College Ave. East, Singapore 138601, Singapore, a0135595@u.nus.edu)

Computation of broadband noise barrier performance in three-dimension is computationally expensive. Indirect boundary element method with double layer potentials has been developed as a numerical tool in this acoustic study. It naturally permits noise barriers to be modelled as thin structures, thus reducing the computational cost when compared to the more abundant direct boundary element tools employed for noise barriers. The present numerical model has been validated with analytical solution. Its computational capabilities are compared with a direct boundary element tool, developed in-house. The applicability of this tool has been demonstrated with the study of fractal acoustic diffusers as noise barrier top-edge devices. Schroeder diffusers have been reported to provide good noise attenuation at their design frequency, when used as noise barrier top-edge devices. This study investigates the broadband performance of fractal diffusers as noise barrier top-edge devices.

4:05

1pPA10. Assessment of learning algorithms to model perception of sound. Menachem Rafaelof (NIA-NASA Langley Res. Ctr., M.S. 463, 100 Exploration Way, Hampton, VA 23681-2199, menachem.rafaelof@nasa.gov) and Andrew Schroeder (Acoust. Branch, NASA Langley Res. Ctr., Lancaster, OH)

Predicting human response to complex sound is a nontrivial task. Besides large differences among subjects and practically infinite types of stimuli, human response to sound is typically quantified by a few parameters having nonlinear behavior. Still, such predictions are valuable for the assessment of sound quality, which is a critical step toward the development of systems that offer improved human comfort, productivity and wellness. The overall objective of this work is to learn about human perception of sound through the use of machine learning algorithms. Learning algorithms are ideal for modeling the complex behavior of subjective parameters and identifying new trends with the potential to accumulate knowledge from different experiments. This work compares the performance of four learning algorithms (linear regression, support vector machines, decision trees, and random forests) to predict annoyance due to complex sound. Construction of these models relies on the annoyance response of 38 subjects to 103 sounds described by five known predictors (loudness, roughness, sharpness, tonality, and fluctuation strength). Comparison of these algorithms in terms of prediction accuracy, model interpretability, versatility, and computation time indicates that decision trees and random forests are the best algorithms for this task.

Session 1pPPa

Psychological and Physiological Acoustics and Signal Processing in Acoustics: Future Directions for Hearing Aids: Multi-Sensor, User-Informed, and Environment-Aware

Martin McKinney, Cochair
Starkey Hearing Technologies, 6600 Washington Ave. S, Eden Prairie, MN 55344

Tao Zhang, Cochair
Signal Processing Research, Starkey Hearing Technologies, 8602 Zachman Circle, Eden Prairie, MN 55344

Invited Papers

1:00

1pPPa1. Trends that are shaping the future of hearing aid technology. Brent Edwards (National Acoust. Labs., Australian Hearing Hub, Level 4, 16 University Ave., Macquarie Univ., NSW 2109, Australia, brent.edwards@nal.gov.au)

The development of hearing aid technology has accelerated over the past decade. Hearing aids are converging with consumer electronics in the area of hearables, and a recent government law mandating the creation of an over-the-counter hearing aid category will continue to bring a more consumer electronics focus to hearing aids. Meanwhile, new advances in hearing science are redefining the criteria of who needs hearing help. This talk will review these new intersections of technology and hearing need. It will also detail the technological and psychoacoustical challenges that face the ability of these new technologies to meet the needs of current hearing aid wearers, the needs of this emerging segment of the hearing impaired, and changes to hearing health delivery.

1:20

1pPPa2. Ecological momentary assessments for evaluation of hearing-aid preference. Karolina Smeds, Florian Wolters, Josefina Larsson, Petra Herrlin, and Martin Dahlquist (ORCA Europe, Widex A/S, Maria Bangata 4, Stockholm SE-118 63, Sweden, karolina.smeds@orca-eu.info)

Ecological Momentary Assessments (EMA) is a method that involves repeated evaluations in for instance a hearing-device user's everyday life. Compared to retrospective evaluations, the EMA method minimizes memory bias, and by making several evaluations each day, patterns in the data can be studied. In a pilot study at ORCA Europe, an EMA method was used to evaluate preference for two hearing-aid programs. For two weeks, the test participants were prompted to make evaluations, using a smartphone questionnaire. They were first asked to describe and classify each listening situation into categories such as "conversation with one person" and "monitoring surroundings." Then they made paired comparisons of the two hearing-aid programs. After the field-trial period, the participants also selected their overall preferred program. Group data for all listening situations pooled showed that the two programs were preferred almost equally often. However, when the data were divided into the seven listening categories, a pattern of preference could be seen. There also seemed to be an association between the test participants' auditory ecology profiles and their overall preferred program. The study will be presented, and the potential of the EMA method for research, development, clinical evaluations, and individualization of hearing-aid settings will be discussed.

1:40

1pPPa3. Listener preferences and speech recognition outcomes using self-adjusted hearing aid amplification. Trevor T. Perry (Speech-Language-Hearing Sci., Univ. of Minnesota, 115 Shevlin Hall, 164 Pillsbury Dr SE, Minneapolis, MN 55455, trevortperry@gmail.com), Peggy B. Nelson (Ctr. for Appl. and Translational Sensory Sci., Univ. of Minnesota, Minneapolis, MN), and Danne VanTasell (Ear Machine, Minneapolis, MN)

Self-adjustment of hearing aid gain empowers users to set amplification parameters according to their personal preferences. Listeners with mild-to-moderate hearing loss demonstrated self-consistency when adjusting gain in noisy backgrounds, although variability across subjects was very high and poorly predicted by user characteristics such as age, hearing thresholds, or previous experience using hearing aids. Despite the wide range of self-fit settings, speech intelligibility in noise was similar between self-fit and audiologist-fit settings. For average-level speech in noisy backgrounds, audibility was driven primarily by the signal-to-noise ratio, not gain settings. Self-adjustment for speech in quiet backgrounds will be compared to results obtained in noise. User interactions with the self-adjustment technology as well as preferences for self-fit settings and audiologist fit settings suggest that users can successfully use self-adjustment technology to achieve more desirable amplification from their hearing aids.

2:00

1pPPa4. Research on hearing-aid self-adjustment by adults. Carol Mackersie, Arthur Boothroyd (Speech, Lang. and Hearing Sci., San Diego State Univ., 2550 Brant St., SDSU: Speech, Lang, Hearing Sci., San Diego, CA 92101, cmackers@mail.sdsu.edu), and Harinath Garudadri (Qualcomm Inst., Univ. of California, San Diego, La Jolla, CA)

The purpose of the work is to develop a protocol for user self-adjustment of hearing aids and to determine its efficacy and candidacy. An initial study involved 26 adults with hearing loss. Control of overall volume, high-frequency boost, and low-frequency cut employed prerecorded and preprocessed sentence stimuli. Participants took a speech-perception test after an initial self-adjustment and then had the opportunity to repeat the adjustment. The final self-selected outputs were not significantly different from those prescribed by a widely used threshold-based method (NAL-NL2)—regardless of prior hearing-aid experience. All but one participant attained a speech-intelligibility index of 60%. Previous users of hearing aids, however, did not meet this criterion until after taking the speech-perception test. This work is continuing with the UCSD Open-source Speech-processing Platform which provides real-time, six-band, processing of microphone input from ear-level transducer assemblies. This system provides a more realistic participant experience, finer control of level and spectrum, and places no limits on the speech materials used for self-adjustment and outcome assessment. Ongoing work investigates the need for the speech-perception test as part of the self-adjustment protocol and the importance of the level and spectrum from which users make their initial adjustments.

2:20

1pPPa5. Tracking eye and head movements in natural conversational settings: Effects of hearing loss and background noise level. Hao Lu (Psych., Univ. of Minnesota, 75 East River Rd., Minneapolis, MN 55455, luxx0489@umn.edu), Martin McKinney, Tao Zhang (Starkey Hearing Technologies, Eden Prairie, MN), and Andrew J. Oxenham (Psych., Univ. of Minnesota, Minneapolis, MN)

Although beam-forming algorithms for hearing aids can produce gains in target-to-masker, the wearer's head will not always be facing the target talker, potentially limiting the value of beam-forming in real-world environments, unless eye movements are also accounted for. The aim of this study was to determine the extent to which the head direction and eye gaze track the position of the talker in natural conversational settings. Three groups of participants were recruited: younger listeners, older listeners with clinically normal hearing, and older listeners with mild-to-moderate hearing loss. The experimental set-up included one participant at a time in conversation with two confederates approximately equally spaced around a small round table. Different levels of background noise were introduced by playing background sounds via loudspeakers that surrounded the participants in the conversation. In general, head movements tended to undershoot the position of the current talker, but head and eye movements together generally predicted the current talker position well. Preliminary data revealed no strong effects of hearing loss, or background noise level on the amount of time spent looking at the talker, although younger listeners tended to use their eyes, as opposed to head movements, more than the older listeners. [Work supported by Starkey Laboratories.]

2:40

1pPPa6. A visually guided beamformer to aid listening in complex acoustic environments. Todd R. Jennings and Gerald Kidd (Dept. of Speech, Lang. & Hearing Sci., Boston Univ., Boston University College of Health and Rehabilitation Sciences: Sargent College, 635 Commonwealth Ave., Boston, MA 02215, toddj@bu.edu)

The purpose of this talk is to provide an overview of recent work on the visually guided hearing aid (VGHA; Kidd *et al.* JASA **133**, EL202). The latest prototype VGHA consists of a lightweight, head-mounted, 18-microphone array combined with PC-based software to create a highly-tuned acoustic beamformer steered in real time by an integrated binocular eye tracker. Recent work from our group on the two main functional sub-components of the VGHA (beamforming and eye-gaze control) focused on the benefits that may be obtained by listeners with hearing loss (and in some cases by listeners with normal hearing) attempting to understand speech masked by one or more competing sounds (such as Gaussian noise or additional speakers). These benefits include improved signal-to-noise ratio under different masked conditions and faster steering (compared to fixed directional amplification steered by head turns). The VGHA may be simulated over headphones by PC-based implementations using HRTFs and an optional external eye-tracker. This has allowed testing a number of variations of beamformer properties and the evaluation of multiple types of sound processing strategies including the incorporation of natural binaural cues. Overall, the VGHA holds considerable promise for improving selective listening in a variety of complex listening conditions.

3:00–3:15 Break

3:15

1pPPa7. Robust and real-time decoding of selective auditory attention from M/EEG: A state-space modeling approach. Sina Miran (Dept. of Elec. and Comput. Eng., Univ. of Maryland, 2351 A. V. Williams Bldg., 8223 Paint Branch Dr., College Park, MD 20742), Sahar Akram (Facebook, Menlo Park, CA), Alireza Sheikhattar, Jonathan Z. Simon (Dept. of Elec. and Comput. Eng., Univ. of Maryland, College Park, MD), Tao Zhang (Starkey Hearing Technologies, Eden Prairie, MN), and Behtash Babadi (Dept. of Elec. and Comput. Eng., Univ. of Maryland, College Park, MD, behtash@umd.edu)

Humans are able to identify and track a target speaker amid a cacophony of acoustic interference, which is often referred to as the cocktail party phenomenon. Results from several decades of studying this phenomenon have culminated in recent years in various promising attempts to decode the attentional state of a listener in a competing-speaker environment from M/EEG recordings. Most existing approaches operate in an offline fashion and require the entire data duration and multiple trials to provide robust results. Therefore, they cannot be used in emerging applications such as smart hearing aids, where a single trial must be used in real-time to decode the attentional state. In this work, we close this gap by integrating various techniques from state-space modeling paradigm such as adaptive filtering, sparse estimation, and Expectation-Maximization, and devise a framework for robust and real-time decoding of the attentional state from M/EEG recordings. We validate the performance of this framework using comprehensive simulations as well as application to experimentally acquired M/EEG data. Our results reveal that the proposed real-time algorithms perform nearly as accurate as the existing state-of-the-art offline techniques, while providing a significant degree of adaptivity, statistical robustness, and computational savings.

3:35

1pPPa8. Towards cognitive control of hearing instruments using EEG measures of selective attention. Torsten Dau, Jonatan Maercher Roersted, Søren Fuglsang, and Jens Hjortkjær (Tech. Univ. of Denmark, Ørsteds Plads, Bldg. 352, Kgs. Lyngby 2800, Denmark, tdau@elektro.dtu.dk)

Single-trial EEG measures of selective auditory attention have recently suggested the perspective of decoding who a listener is focusing on in multi-talker scenarios. Here, we report results from work within the COCOHA (Cognitive Control of a Hearing Aid) project investigating the possibility of integrating EEG into neuro-steered hearing instruments. Our EEG decoding strategy relies on measuring cortical activity entrained to envelope fluctuations in the attended speech signal. Currently, a major challenge has been to obtain robust EEG measures of selective attention in older hearing-impaired (HI) listeners. We report our recent COCOHA attempts to decode selective attention from the EEG of hearing-impaired (HI) listeners. Aided HI listeners and age-matched normal-hearing controls were presented with competing talkers at 0 dB target-to-masker ratio and instructed to attend to one talker. We show that single-trial decoding accuracies similar to those reported for younger listeners can be obtained with both groups of older listeners (70–100% correct single-trial classification). Importantly, we did not find differences in decoding accuracies between the NH and the aided HI listeners. Although numerous other challenges involved in integrating EEG signals in hearing instruments are evident, our results suggest that single-trial attention decoding is possible with hearing impaired listeners.

3:55

1pPPa9. Towards transparent EEG: Ear-centered EEG acquisition using the cEEGrid. Martin G. Bleichner and Stefan Debener (Psych., Univ. of Oldenburg, Universitaet Oldenburg, Ammerlaender Heerstrasse, Oldenburg 26111, Germany, martin.bleichner@uol.de)

Ear-centered electroencephalography (EEG) allows for inconspicuous acquisition of EEG signals. Different approaches of placing electrodes in the ear canal, the concha, or around the ear have been proposed. As integral part of a hearing device, they promise to capture the neural correlates of a user's listening intent or listening effort in everyday situations. We have developed a ten electrode c-shaped array (cEEGrid); the flex-printed grid is positioned around the ear using an adhesive tape and allows for extended (>8 hours) high quality EEG recordings. We have shown that neural activity related to auditory processing originating from different neural sources can be reliably recorded. The signals recorded from different locations around the ear contain non-redundant information. The signal similarity between electrodes depends on their respective angle relative to the reference electrode. If this angle difference is small (20°) the signals recorded at these electrodes correlate highly (correlation score around 0.8). If this angle difference is large (160°) the signals recorded at these electrodes are uncorrelated. From this we conclude that the distributed electrode placement around the ear provides an interesting perspective on different neural processes. In combination with smartphone based experimentation, signal acquisition and analysis ear-centered EEG may soon enable the use of brain signals captured in daily life situations.

4:15

1pPPa10. Robust speech processing in human auditory cortex. Nima Mesgarani (Elec. Eng., Columbia Univ., 560 Riverside Dr. 17B, New York, NY 10027, nima@ee.columbia.edu)

The brain empowers humans with remarkable abilities to navigate their acoustic environment in highly degraded conditions. This seemingly trivial task for normal hearing listeners is extremely challenging for individuals with auditory pathway disorders, and has proven very difficult to model and implement algorithmically in machines. In this talk, I will present the result of an interdisciplinary research effort where invasive and non-invasive neural recordings from human auditory cortex are used to determine the representational and computational properties of robust speech processing in the human brain. These findings show that speech processing in the auditory cortex is dynamic and adaptive. These intrinsic properties allow a listener to filter out irrelevant sound sources, resulting in a reliable and robust means of communication. Furthermore, incorporating the functional properties of neural mechanisms in speech processing models greatly impact the current models of speech perception and at the same time, lead to human-like automatic speech processing technologies.

4:35

1pPPa11. An optimization model for electroencephalography-assisted binaural beamforming. Wenqiang Pu (Xidian Univ., Shenzhen, Guangdong, China), Jinjun Xiao, Tao Zhang (Starkey Hearing Technologies, Eden Prairie, MN), and Zhi-Quan Luo (Shenzhen Res. Inst. of Big Data, Longxiang Rd. 2001, Shenzhen, Guangdong, China, luoxx032@umn.edu)

In a multi-speakers noisy environment, the incoherent correlation between the electroencephalography (EEG) signal and the attended speech envelope provides a *user-informed* way to design the beamformer for preserving the attended speech and suppressing others. In this work, we exploit such incoherent correlation property in terms of Pearson correlation and propose a unified optimization model for simultaneously designing beamformer and aligning *attention preference* of the user to each speech source. To balance different design considerations, the proposed optimization formulation makes a trade-off among aligning attention preference, controlling speech distortion, and reducing noise in a weighted manner in the objective function. Specifically, the attention preference is aligned by maximizing the Pearson correlation between the envelope of beamforming output and the linearly transformed EEG signal. To control speech distortion with flexibility, the spatial response of the beamformer to each source is penalized in a min-max sense. And the noise is further reduced by minimizing its mean squares at beamforming output. Experiments on collected EEG signal in two speakers environment demonstrate the effectiveness of the proposed model.

4:55

1pPPa12. Realistic virtual audiovisual environments for evaluating hearing aids with measures related to movement behavior. Maartje M. Hendrikse, Gerard Llorach, Giso Grimm, and Volker Hohmann (Medizinische Physik and Cluster of Excellence Hearing4all, Universität Oldenburg, Carl-von-Ossietzky-Straße 9-11, Oldenburg 26129, Germany, maartje.hendrikse@uni-oldenburg.de)

With increased complexity of hearing device algorithms a strong interaction between motion behavior of the user and hearing device benefit is likely to be found. To be able to assess this interaction experimentally more realistic evaluation methods are required that mark a transition from conventional (audio-only) lab experiments to the field. In this presentation, we describe our methodology for acquiring ecologically valid behavioral data in realistic virtual audiovisual testing environments. The methods are based on tools to present interactive audiovisual environments while recording subject behavior with gaze and motion tracking systems. The results of a study that evaluated the effect of different types of visual information (e.g., video recordings vs. animated characters) on behavior and subjective user experience are presented. It was found that visual information can have a significant influence on behavior and that it is possible to systematically assess this. Furthermore, first results are presented of two studies that observed head and eye movement behavior: (1) in typical everyday listening situations that were replicated with virtual audiovisual environments in the lab (e.g., cafeteria) and (2) when visual cues were presented via a head-mounted display or projected onto a panoramic cylindrical screen in front of the subject.

5:15

1pPPa13. Multimodal signal processing and machine learning for hearing instruments. Tao Zhang and Martin McKinney (Signal Processing Research, Starkey Hearing Technologies, 8602 Zachman Circle, Eden Prairie, MN 55344, tzhang28@ieee.org)

With advances in wireless technology and sensor miniaturization, more and more non-audio sensors become available to and are being integrated into hearing instruments. These sensors help not only improve speech understanding and sound quality, enhance hearing usability and expand the hearing instruments' capabilities to health and wellness monitoring. However, the introduction of these sensors also present a new set of challenges to researchers and engineers. Compared with traditional audio sensors for hearing instruments, these new sensor inputs can come from different modalities and often have different scales and sampling frequencies. In some cases, they are not linear or synchronized to each other. In this presentation, we will review these challenges in details in the context of hearing instruments applications. Furthermore, we will demonstrate how multimodal signal processing and machine learning can be used to overcome these challenges and bring a greater degree of satisfactions to the end users. Finally, future directions in multimodal signal processing and machine learning research for hearing instruments will be discussed.

MONDAY AFTERNOON, 7 MAY 2018 — NICOLLET A, 1:00 P.M. TO 5:00 P.M.

Session 1pPPb

Psychological and Physiological Acoustics: Spectral, Pitch, and Physiological Measurements (Poster Session)

Tess K. Koerner, Chair
University of Minnesota, 3330 SE Tibbetts St., Portland, OR 97202

Posters will be on display from 1:00 p.m. to 5:00 p.m. To give contributors in this session an opportunity to view other posters, contributors of odd-numbered papers will be at their posters from 1:00 p.m. to 3:00 p.m. and contributors of even-numbered papers will be at their posters from 3:00 p.m. to 5:00 p.m.

Contributed Papers

1pPPb1. Human dissimilarity ratings of musical instrument timbre: A computational meta-analysis. Etienne Thoret (Schulich School of Music, McGill Univ., 555 Rue Sherbrooke Ouest, Montreal, QC H3A 1E3, Canada, etienne.thoret@mcgill.ca), Baptiste Caramiaux (CNRS LRI UMR 8623, Univ. Paris-Sud, Paris, France), Philippe Depalle, and Stephen McAdams (Schulich School of Music, McGill Univ., Montreal, QC, Canada)

Musical instrument timbre has been intensively investigated through dissimilarity rating tasks. It is now well known that audio descriptors such as attack time and spectral centroid, among others, account well for the dimensions of the timbre spaces underlying these dissimilarity ratings. Nevertheless, it remains very difficult to reproduce these perceptual judgments from distances computed on acoustical representations such as the waveform or the spectrogram. Interestingly, biologically inspired representations based on spectrotemporal modulation spectra such as spectrotemporal receptive fields (STRF) have been shown to be well-suited to reproduce human dissimilarity ratings (Patil *et al.*, 2012). Here, we propose a meta-analysis of seven former studies on timbre spaces in light of these recently developed representations. We implemented a computational framework that optimizes the correlation between the perceptual results and distances

obtained from a set of different acoustic representations, in particular through the STRF. We observed that distances computed from spectrotemporal modulation representations provide the best correlation with the perceptual results across the seven timbre spaces. Finally, we highlighted the parts of the representations contributing the most to the correlation suggesting new insights into the underlying perceptual metrics. [Supported by Canada Research Chair, NSERC (RGPIN-2015-05208, RGPAS-478121-15), (RGPIN-262808-2012), and EU MSCf (Project MIM, H2020-MSCA-IF-2014, GA no. 659).]

1pPPb2. Neural coding of perceptual temporal asymmetry for sounds with rising vs. falling intensity envelopes in non-attentive and attentive listening conditions. Bing Cheng (English Dept. & Inst. for Lang., Cognition and Brain Sci., Xi'an Jiaotong Univ., 28 Xianning St. West, School of Foreign Studies, Xi'an, Shaanxi 710049, China, bch@mail.xjtu.edu.cn), Yang Zhang (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN), Keita Tanaka (School of Sci. and Eng., Tokyo Denki Univ., Hatoyama, Saitama, Japan), Robert S. Schlauch (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN), and Toshiaki Imada (Inst. for Learning and Brain Sci., Univ. of Washington, Seattle, WA)

Behavioral studies have shown that ramped sounds are judged to be louder and longer than damped sounds. Here we employed magnetoencephalography (MEG) to examine cortical processing of the perceptual temporal asymmetry. The participants were 6 normal-hearing right-handed male adults. The synthesized stimuli included three kinds, pure tone, piano note, and broadband noise with time-reversed intensity envelopes for the ramped vs. damped comparison. Each stimulus was 200 ms. In the non-attentive condition, subjects were instructed to watch a silent movie and ignore the randomly presented tones. In the attentive condition, listeners were required to judge whether the first or the second of paired sounds was longer. The stimuli were presented at 50 dB SL. The behavioral results replicated previous temporal asymmetry findings for all three types of stimuli. There were significant effects of stimulus type and stimulus order in the MEG ON and OFF responses. Despite the stimulus type effect, the ramped tonal stimuli consistently elicited smaller and later N1m responses than the damped controls in both ignore and attend conditions. These data supports distinct cortical mechanisms for coding the stimulus type information and subjective duration information of the ramped vs. damped auditory stimuli.

1pPPb3. Optimal frequency filtering of auscultation sounds. Lukasz Nowak (Inst. of Fundamental Technolog. Res., Polish Acad. of Sci., ul. Pawinskiego 5B, Warszawa 02-106, Poland, lnowak@ippt.pan.pl)

Most of the auscultation sounds do not reveal any significant single-frequency components, and their acoustic energy is concentrated in the low frequency region—up to about 100 Hz, falling even below the threshold of hearing. Such character is determined not only by the vibroacoustic behavior of sources, but mostly by high damping introduced by the sound transmission path through tissues underlying the skin surface. The contained diagnostic information is very subtle, and thus it can be easily masked by internal or external noise sources. Not all of those corrupting signals can be efficiently blocked, hence frequency filtering is the most obvious solution for improving the diagnostic capabilities. Many various filtering strategies and techniques were developed and implemented in both acoustic and electronic stethoscopes, however they are based primarily on (not always correct) intuition and subjective evaluation, without implementation of any accurate and objective measurement means or optimization algorithms. The present study introduces various signal to noise ratio (SNR) measures, applicable for different examination cases. Frequency filtering optimization strategies, for maximizing the values of the introduced coefficients for different heart and lung auscultation sounds, are presented.

1pPPb4. The effects of frequency fine-tuning in hearing impaired phone recognition. Ali Abavisani and Jont Allen (ECE, Univ. of Illinois at Urbana-Champaign, 405 N Mathews Ave., Rm. 2137, Urbana, IL 61801, aliabavi@illinois.edu)

A key factor on correct phone recognition in Normal Hearing (NH) and Hearing Impaired (HI) listeners, is the intensity of primary cue. One can assess this intensity for a given speech sound, by examining it at various Signal to Noise Ratios (SNR) presented to NH listeners, and detect the threshold in which listeners recognized the token at least 90% correct (SNR_{90}). For each token, we have determined the time-frequency window corresponding to correct recognition, as well as the conflicting cues. Two sets of tokens T_1 and T_2 having same consonant-vowels but different talkers with distinct SNR_{90}s had been presented at flat gain (frequency independent) at listeners' most comfortable level (MCL). We studied the effects of frequency fine-tuning of the primary cue by presenting tokens of same consonant but different vowels with similar SNR_{90}s. Additionally, we investigated the role of changing the intensity of primary cue on HI phone recognition, by presenting tokens from both sets T_1 and T_2. This presentation discusses how the frequency an/or intensity of the primary cue changes the confusion pattern of phone recognition for HI listeners, given a flat gain condition at MCL. We will also explore the effect of these changes on conflicting cues.

1pPPb5. An oscillatory template pitch model. David A. Dahlbom and Jonas Braasch (Architecture, Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, dahlbd@rpi.edu)

Traditional approaches to explaining missing fundamental and pitch-shift phenomena have relied on either the fitting of harmonic templates to spectral information (generally assumed to correspond to a neural excitation pattern), or by performing an autocorrelation-based calculation on temporal information. In this paper, an alternative approach relying on the dynamics of phase-locking oscillators is proposed. Rather than applying some decoding procedure, such as spectral analysis or autocorrelation, a pitch estimate is given by the state (firing frequency) of an appropriately excited oscillator in a template structure operating on peripheral information. This approach is shown to reproduce many of the classical pitch shift-phenomena. In addition to a direct implementation in terms of phase-locking oscillators, a simplified model consisting of arrays of adaptive templates, operating directly on timing information, is also presented. This latter approach offers a straightforward way to tune the model in accordance with existing psychoacoustical data and suggests an approach to modeling pitch strength and multiple f0 detection. The overall goal is to suggest the advantages of a modeling approach that relies on arrays of tuned, adaptive elements which are physiologically-inspired, though not intended to be detailed representations of precise physiological components. [Work was supported by NSF BCS-1539276.]

1pPPb6. Pitch synchronous speech analysis for the assessment of subjects with Parkinson's disease. Sai Bharadwaj Appakaya and Ravi Sankar (iCONS Res. Lab, Dept. of Elec. Eng., Univ. of South Florida, 4202 E Fowler Ave. ENB 381, Tampa, FL 33620, saibharadwaj@mail.usf.edu)

Analysis of speech samples from subjects with Parkinson's Disease (PD) is a field of growing research interest. Studies in this field predominantly include pitch-based features with the intuition that PD affects the movement of musculature involved in speech production. These features are usually extracted on a segment of fixed length that traverses over the entire speech sample. This methodology, however, gives the net estimate of the feature over each segment and cannot account for the fast variations that transpire before and after a vocal fold closure. In this paper, we present a pilot study with the focus on in-depth pitch synchronous analysis that can fill-in the gap by capturing the said variations. Speech samples from 22 patients are preprocessed and used for analysis. Data that can pin point the precise indices of the vocal fold closures for each cycle are extracted in preprocessing and features that can capture the swift variations are extracted from every cycle and used for analysis. The results show a good margin in features between PD and healthy speech that reinforced the intuition.

1pPPb7. Acoustic correlates of auditory object perception: Speakers, musical timbres, and environmental sounds. Mattson Ogg and L. Robert Slevc (Neurosci. and Cognit. Sci., Univ. of Maryland, College Park, Biology/Psych. Bldg. Rm. 3150, 4094 Campus Dr., College Park, MD 20742, mogg@umd.edu)

Human listeners must identify and orient themselves to auditory objects in their environment. What acoustic features support a listener's ability to differentiate the variety of sound sources they might encounter? Typical studies of auditory object perception obtain dissimilarity ratings between pairs of objects, often within a single category of sound. However, such an approach precludes an understanding of general acoustic features that might be used to differentiate sounds across categories. The present experiment takes a broader approach to the analysis of dissimilarity ratings by leveraging the acoustic variability within and between different sound categories as characterized by a large, diverse set of 36 sound tokens (12 speech utterances from different speakers, 12 instrument timbres, and 12 everyday objects from a typical human environment). We analyze multidimensional scaling results as well as models of trial- level dissimilarity ratings as a function of different acoustic representations including spectral, temporal and noise features as well as modulation power spectra and cochlear spectrograms. In addition to previously noted differences in spectral and temporal envelopes, results indicate that listener's dissimilarity ratings are also related to spectral variability and noise, particularly in differentiating sounds between categories. Dissimilarity ratings also appear to closely parallel sound identification performance.

1pPPb8. A pitch encoding model based on the intrinsic oscillation circuit: Effects of the time constant and input stimulus types. Minoru Tsuzaki (Kyoto City Univ. of Arts, 13-6 Kutsukake-cho, Oe, Nishikyo-ku, Kyoto 610-1197, Japan, minoru.tsuzaki@kcua.ac.jp) and Katuhiro Maki (Faculty of Human Informatics, Aichi Shukutoku Univ., Aichi, Japan)

Many temporal models for pitch perception have adopted a configuration of delay-lines and coincidence detectors after the cochlear filtering. Autocorrelation functions are a usual way of its implementation. However, a series of experiments by the authors' group have revealed that the perceived pitch would shift upwards by the effect of aging. Because the autocorrelation simply represents the time intervals statistics in the physical domain, the aging cannot affect this statistics. Therefore, a further pitch encoding process where the physical (physiological) temporal intervals are mapped against any internal reference in the brain. We propose a model comprised of bank of self-oscillatory circuits and the coincidence detectors. The periods of oscillations are intrinsic characteristics of the neural circuit. The proposed model could pick up the fundamental periods of a various types of stimuli, i.e., pure tones, missing fundamentals and iterated ripple noises. To check an effect of aging, the changes in the time constant of the temporal waveform of the oscillation was investigated. Although the shortest sensitive period increased due to the longer time constant, the age-induced pitch shift could not be predicted. It is necessary to assume a systematic lengthening of the period of each oscillation to predict the pitch shift.

1pPPb9. Speaker-dependent low-level acoustic feature extraction for emotion recognition. Tejal Udhan and Shonda Bernadin (Elec. Eng., Florida State Univ., 2525 Pottsdamer St., Tallahassee, FL 32310, tu13b@my.fsu.edu)

In this paper, accuracy for emotion recognition using low- level acoustic features is investigated. The aim of any speech emotion recognition system is to extract acoustic features that are representative of the emotional state of the speaker. Frequency formants, intensity, and pitch are the low-level features proposed for characterizing four different emotions, anger, happy, sadness, and neutral, using acoustic data. Low- level features describe the acoustic, prosodic, and spectral properties of the speech signal and limit the complexity of emotion recognition systems. An algorithm is designed for characterizing each emotion using the acoustic features. It has been proven that various aspects of a speaker's physical and emotional state can be identified by speech alone. However, the accuracy of such analyses has not been optimized due to acoustic variabilities such as length and complexity of human speech utterance, gender, speaking styles and speech rate. It has also been found that speaker-dependent systems are more accurate in emotion recognition than that of speaker-independent systems. Since speech emotion recognition is relatively a newer field, the set of most powerful features which can distinguish different emotions is not defined; hence, examining the accuracy of emotion recognition using selected acoustical features is an important task.

1pPPb10. Predicting timbral and perceptual characteristics of orchestral instrument combinations. Aurelien Antoine and Eduardo Miranda (Interdisciplinary Ctr. for Comput. Music Res. (ICCMR), Univ. of Plymouth, Plymouth University, The House Bldg. - Rm. 304, Plymouth PL4 8AA, United Kingdom, aurelien.antoine@postgrad.plymouth.ac.uk)

Orchestration is a compositional practice that consists of writing for several instruments. This process often involves harnessing each instrument's sound to create sonic textures that could not be achieved with a single instrument. These sound fusions are usually sought by composers to express specific perceptual effects. However, the number of potential combinations is significant. Testing and analyzing all combinations to identify the ones matching the desired perceptual effects is logistically and computationally complex. Using supervised learning methods to create regression and classification models, it is possible to predict specific timbral and perceptual characteristics from information about a combination of different orchestral instruments. Such developments would provide methods to estimate the perception of instrument timbre fusions directly from abstract information. Similar methods could potentially be applied to other types of sources and predict specific perceptual characteristics without the need to perform an acoustical and psychoacoustical analysis on every audio sources.

1pPPb11. Informational masking of calm speech targets by emotional speech maskers. Shae D. Morgan (Commun. Sci. and Disord., Univ. of Utah, 390 South 1530 East, Ste. 1201, Salt Lake City, UT 84112, shae.morgan@utah.edu)

Recent research suggests that some types of emotional speech are more intelligible than others when presented in background noise (Dupuis & Pichora-Fuller, 2015). Yet, emotional speech can often be present as the background noise itself. Attentional mechanisms and auditory stream segregation capabilities likely impact target word recognition in these situations with emotional speech maskers. The present study examined calm target sentences in masking background combinations of four emotions (calm, sad, happy, and angry) and two masker types (2-talker babble and speech-shaped noise). The emotion categories differ from one another on perceptual dimensions of activation (low to high) and pleasantness (unpleasant to pleasant). Speech-shaped noise maskers were spectrally and temporally similar to the 2-talker babble maskers. Performance was compared between the two masker type conditions to quantify the amount of informational masking induced by different emotional speech maskers. Furthermore, the number of target-masker confusions were identified for each masker emotion and subsequently analyzed. Results showed lowest target recognition with calm speech maskers, consistent with auditory stream segregation framework; however, confusions and informational masking were high for angry and happy speech maskers, which suggests that auditory attention to high-arousal stimuli may have influenced performance.

1pPPb12. Mechanical sound similarity and categorization. Annamarie Rosen, Nirmal Kumar Srinivasan (Audiol., Speech-Lang. Pathol., and Deaf Studies, Towson Univ., 8000 York Rd., Towson, MD 21252, nsrinivasan@towson.edu), Jeremy Gaston, and Brandon Perelman (Human Res. and Eng. Directorate, U.S. Army Res. Lab., Adelphi, MD)

Recent studies suggest that perceptual similarity among sounds can be used to predict accuracy in perceptual tasks (Dickerson & Gaston, 2014; Gaston, *et al.*, 2017). We operationalize this similarity as distance in a 2D similarity space generated using multidimensional scaling (MDS) on pairwise similarity stimulus ratings. We tested listeners' ability to categorize mechanical environmental sounds, and modeled the relationship between stimulus categorization and similarity. Stimuli were 18 mechanical sounds selected hierarchically from three broad categories (vehicles, power tools, and household appliances), and three subcategories. Thirty-five normal hearing listeners completed pair-wise similarity rating and categorization tasks. Perceptual similarity ratings from all paired stimuli were used to develop an MDS solution. Stimuli in the solution generally were segregated by broad category membership, with some exceptions. Categorization performance for each broad category varied, with vehicles being the easiest and household appliances being the most difficult to categorize. MDS distances among the stimuli were used in a simple regression model to predict performance in the categorization task and explain confusions in categorization. The results of this study indicate that perceptual similarity is a strong predictor of variance in listeners' judgments of complex environmental sounds.

1pPPb13. The effects of carrier bandwidth and intensity on spectral ripple perception in listeners with hearing loss. Evelyn E. Davies-Venn, Kristi Oeding, and Andrew Haug (SLHS, Univ. of Minnesota, 164 Pillsbury Dr SE, Minneapolis, MN 55445, venn@umn.edu)

Listeners with hearing loss often need to listen to speech signals at high intensities to ensure proper audibility. Spectral processing deteriorates at high intensities, but signal acoustics such as bandwidth and filtering can be used to mitigate this deterioration. This study evaluated the effect of carrier bandwidth on broadband spectral processing at high intensities for individuals with hearing loss. Spectral modulation detection thresholds were measured using a rippled noise carrier with varying bandwidths from 1 to 4 octaves, to assess whether individuals with hearing loss were more susceptible to the deleterious effects of high-intensity compared to their counterparts with normal hearing. Results show that spectral processing degrades at high intensities. Listeners with hearing loss benefited from the increase in signal bandwidth at high intensities much more than their counterparts with normal hearing. Our findings suggest that spectral processing for broadband signals involve within and across channel resolution and both succumb to the negative effect of level-induced broadening of auditory-filter bandwidths. A better understanding of how filtering can be used to alleviate the adverse effect of high-intensity signals could prove useful for improving outcomes for those individuals whose only viable treatment option often involves listening to speech signals at high intensities.

1pPPb14. Multiple mechanisms in pitch perception revealed by individual differences. Malinda J. McPherson (Speech and Hearing BioSci. and Technol., Harvard Univ., MIT Bldg. 46-4078, 43 Vassar St., 46-4078, Cambridge, MA 02139, malindamcpherson@g.harvard.edu) and Josh McDermott (Brain and Cognit. Sci., Massachusetts Inst. of Technol., Cambridge, MA)

Pitch perception is traditionally assumed to involve the estimation of fundamental frequency (F0) from sound. Some pitch-related tasks, such as those involving musical intervals, are more difficult with inharmonic sounds (lacking a clear F0). Other tasks, such as up-down pitch discrimination, are unaffected by inharmonicity, suggesting that listeners track the spectrum, not the F0. To test whether these effects reflect distinct mechanisms, we measured individual differences among large sets of participants on a battery of tasks. Performance on tasks using a common mechanism should be correlated across participants. Two tone up-down discrimination for wideband harmonic tones was highly correlated with performance for wideband inharmonic tones. To isolate F0-based pitch, we also measured discrimination for pairs of harmonic tones with non-overlapping sets of harmonics. Harmonic and inharmonic discrimination did not strongly correlate with this latter task. These results suggest that the same mechanism subserves up-down change detection for harmonic and inharmonic tones, and that this mechanism is distinct from F0-based pitch estimation. We also found weak correlations between spectral centroid discrimination and all other pitch tasks, suggesting separate mechanisms for fine and coarse spectral comparisons. The results suggest an additional pitch mechanism for tracking shifts in the spectrum rather than F0.

1pPPb15. The function of F0-based pitch. Malinda J. McPherson (Speech and Hearing BioSci. and Technol., Harvard Univ., MIT Bldg. 46-4078, 43 Vassar St., 46-4078, Cambridge, MA 02139, malindamcpherson@g.harvard.edu) and Josh McDermott (Brain and Cognit. Sci., Massachusetts Inst. of Technol., Cambridge, MA)

Pitch is traditionally described as the perceptual correlate of a sound's fundamental frequency (F0). The importance of perceiving F0 is often attributed to the need to compare sounds with different timbres, for which spectral comparisons might be error-prone. Alternatively, because the F0 is a summary measure of a set of harmonic frequencies, it could provide a compressed representation of sound for memory storage. To clarify the function of F0-based pitch, we tested its importance for discriminating sounds with different spectra, and sounds separated by delays, using both harmonic and inharmonic stimuli. If a task relies on F0-based pitch, performance should be impaired for inharmonic stimuli. We observed an advantage for harmonic stimuli for discrimination of synthetic tones with extreme (and unnatural) spectral variation. However, no such advantage was observed for instrument or speech fragments (with naturally occurring spectral variation). By contrast, discrimination was better for harmonic tones separated by brief (5–10 s) delays, suggesting that F0-based pitch aided storage across the delay. The results indicate that the spectral variation found in natural pitched sounds does not itself make listeners reliant on F0 estimation. The results raise the possibility that F0 primarily serves to facilitate compression of sounds for memory storage.

1pPPb16. Enhanced sine wave speech with an added cue for pitch improves tone perception in Cantonese. Amy Wu (Linguist, Brooklyn College, CUNY, Brooklyn College Linguist Program, 2308 Boylan Hall, 2900 Bedford Ave., Brooklyn, NY 11210, di5tortion@yahoo.com) and Jon Nissenbaum (Linguist, Brooklyn College and the Graduate Ctr., CUNY, Brooklyn, NY)

Cantonese has six lexical tones (four level and two rising) that distinguish otherwise identical syllables. Although it is known that other acoustic cues besides f0 (e.g., voice quality) enter into tone perception, it is unknown whether f0 alone provides a sufficient cue. To test this, modified Sinewave Speech (SWS) replicas of Cantonese sentences were presented to listeners. SWS replaces vocal tract formants with frequency-modulated sinusoids, and has been shown to support perception of phonemic content of speech. However, because formant trajectories alone lack information about f0 and voice quality, standard SWS is unsuitable for studying tone perception (Feng *et al.*, 2012, JASA131/EL133; Rosen & Hui, 2015, JASA138). To create the stimuli, the lowest sinusoid of the SWS replica (representing F1) was replaced with a complex tone constructed using a bandpass with the same center frequency as F1, just wide enough at any timepoint for two harmonics of an independently specified f0 contour. This two-component tone implies a missing fundamental, allowing simultaneous impression of harmonic direction and F1 direction. Modified and unmodified SWS replicas of Cantonese words were presented to native Cantonese speakers to ascertain whether implied f0 improves tone perception; preliminary results support an effect of modified SWS.

1pPPb17. Modified two-component Shepard tones and their application to sine wave speech. Jon Nissenbaum (Linguist, Brooklyn College and the Graduate Ctr., CUNY, Dept. of English, 2308 Boylan Hall, Brooklyn College, 2900 Bedford Ave., Brooklyn, NY 11210, jnissenbaum@brooklyn.cuny.edu)

Sinewave Speech (SWS), consisting of several frequency-modulated sinusoids representing vocal tract formants, elicits perception of words/sentences, making it useful for studying the perceptual basis of speech. However, SWS contains no information relevant for pitch perception, making it a poor tool for investigating tone languages or prosody (Remez & Rubin, 1993, JASA94; Rosen & Hui, 2015, JASA138). The objective of this study was to develop a method for creating modified SWS replicas adding a minimal cue for pitch. The lowest sinusoid was replaced with a "modified Shepard tone:" a tone formed from a bandpass whose center frequency tracks F1, wide enough at any timepoint for exactly two harmonics of an independently specified f0 contour. It has been shown (Ritsma 1962, JASA34, Smoorenburg 1970, JASA48, Houtgast 1976, JASA60) that tones consisting of 1–3 harmonics elicit perception of a missing fundamental. An initial experiment confirmed that a bandpass whose center frequency fell from 700 to 400 Hz over 400 ms, intersected with harmonics of a rising f0 (85–185 Hz), elicited a strong perception of a rising contour, and vice versa. A second experiment replaced F1 in a set of SWS replicas with these two-component tones and elicited robust perception of contrasting pitch contours.

1pPPb18. Short- and long-term memory for pitch and non-pitch contours in congenital amusia. Jackson Graves (Psych., Univ. of Minnesota, 75 E River Parkway, Minneapolis, MN 55455, grave276@umn.edu), Lesly Fornoni, Agathe Pralus (Lyon Neurosci. Res. Ctr. (CRNL), Université Lyon 1, Lyon, France), Andrew J. Oxenham (Psych., Univ. of Minnesota, Minneapolis, MN), Anne Caclin, and Barbara Tillmann (Lyon Neurosci. Res. Ctr. (CRNL), Université Lyon 1, Lyon, France)

Congenital amusia is a disorder characterized by deficits in music and pitch perception, but the degree to which the deficit is specific to pitch remains unclear. Amusia often results in difficulties discriminating and remembering melodies and melodic contours. Non-amusic listeners can perceive contours in dimensions other than pitch, such as loudness and brightness, but it is unclear whether amusic pitch contour deficits also extend to these other auditory dimensions. This question was addressed by testing the identification of ten familiar French melodies and the discrimination of changes in the contour of novel four-note melodies. Melodic contours were defined by pitch, brightness, or loudness. Amusic participants were impaired relative to matched controls in all dimensions, but showed some ability to extract contours in all three dimensions. In the novel contour discrimination task, amusic participants exhibited less impairment for loudness-based melodies than for pitch- or brightness-based melodies, suggesting some specificity for pitch (including spectral pitch). Impairment in loudness contours for familiar melodies may have reflected the fact that the long-term memory representation of the familiar melody was defined in pitch. [Work supported by Erasmus Mundus Auditory Cognitive Neuroscience Network, NIH grant R01DC005216, LabEx CeLyA, and Cortex (ANR-11-IDEX-0007).]

1pPPb19. Frequency difference limens as a function of fundamental frequency and harmonic number. Anahita H. Mehta and Andrew J. Oxenham (Univ. of Minnesota, N640, Elliott Hall, 75 East River Parkway, Minneapolis, MN 55455, mehta@umn.edu)

Several studies have investigated the relation between the lowest harmonic present in a complex tone and fundamental frequency (F0) difference limens (F0DLs). It is generally assumed that F0DLs are smaller when lower harmonics are present and that the ability to discriminate small changes in F0 worsen as harmonic number increases. This worsening of performance has been attributed to a lack of peripherally resolved harmonics. This assumption was tested by measuring F0DLs for harmonic complexes where the lowest harmonic present in a twelve-harmonic complex tone varied from the 3rd to the 15th harmonic, with F0s varying from 30 Hz to 2000 Hz. The harmonics were presented in either sine or random phase and were embedded in threshold-equalizing noise. Aside from F0s between 100 and 400 Hz, performance did not follow the expected pattern of good performance with low-numbered (resolved) harmonics and poorer performance with high-numbered (unresolved) harmonics. At lower F0s, performance was relatively constant across the harmonic range; at higher F0s, performance generally degraded more than would be predicted based solely on harmonic resolvability. No clear effects of phase were observed. Overall, the results do not match the patterns expected from current models of complex pitch. [Work supported by NIH grant R01DC005216.]

1pPPb20. Gliding ripple speed thresholds for rippled spectra. Alexander Supin, Olga Milekhina, and Dmitry Nechaev (Institute of Ecology and Evolution, 33 Leninsky Prospect, Moscow 119071, Russian Federation, alex_supin@mail.ru)

Rippled noise is a productive model of natural signals with complex spectrum patterns. It was used as a test signal to measure spectrum-pattern resolution both in normal-hearing listeners and in hearing-impaired listeners and users of cochlear implants. However, a variety of natural auditory signals feature combined spectro-temporal patterns. These signals may be modeled by rippled noise with gliding ripples. In the present study, ripple gliding speed thresholds were measured in normal-hearing listeners. The highest gliding speed (expressed in oct/s or ripples/s) at which the gliding ripple pattern could be distinguished from a non-rippled noise was determined. The ripple gliding speed threshold expressed in ripple/s was inversely-proportional to the ripple density (ripples per octave, RPO); it decreased from approximately 400–500 ripple/s at RPO of 1/oct to approximately 50 ripple/s at RPO of 10/oct. Expressed in oct/s, the threshold decreased from 400 to 500 oct/s at RPO of 1/oct to 5 oct/s at RPO of 10/oct. Thus, the ripples fused into non-rippled spectrum at gliding speed more than 400 to 500 Hz/oct. At this speed, 400 to 500 ripples per second glide through an octave frequency band. [Work supported by The Russian Science Foundation, Grant 16-15-10046.]

1pPPb21. Masking of synthesized vowel sequences: The potential roles of perceptual segregation. Yi Shen and Dylan Pearson (Speech and Hearing Sci., Indiana Univ. Bloomington, 200 S Jordan Ave., Bloomington, IN 47405, shen2@indiana.edu)

Modulation masking is known to impact speech intelligibility, but it is not clear how this phenomenon interacts with the perceptual segregation between the target speech and background noise. Using an experimental task in which listeners identified synthesized-vowel sequences in amplitude-modulated noises, a previous study from our laboratory found few signs of modulation masking, contrary to the findings of speech-on-speech masking experiments. It is hypothesized that modulation masking at a low rate (<6 Hz) may reflect failures of segregating the target speech from the noise. The current study systematically removed segregation cues and evaluated the amount of modulation masking. These segregation cues included the periodicity in the temporal fine structure of the target vowel sequences (in Exp. I) and the temporal regularity of the vowel sequences (in Exp. II). Results will be discussed in terms of the potential interactions between modulation masking and perceptual segregation.

1pPPb22. Primacy and recency effects in serial recall of synthesized vowel sequences. Yi Shen (Speech and Hearing Sci., Indiana Univ. Bloomington, 200 S Jordan Ave, Bloomington, IN 47405, shen2@indiana.edu)

Recognizing running speech involves not only audibility of speech signals but also cognitive processes that support the encoding and retrieving of speech information to and from memory. The current study aims to separate the effects of audibility and cognitive load on speech recognition performance. Synthesized vowel sequences were presented in noise, and the listeners were instructed to recall all vowels in the sequences in the correct order. In separate conditions, the sequence lengths were 2, 5, and 8 vowels, the presentation rates were 1, 2, 4, and 6 Hz, and the vowel durations were between 12.5 and 100 ms. The signal-to-noise ratios used in the experiment were adjusted for each listener and each of the vowel durations so that the expected recognition performance for isolated vowels was always 80%. A primacy effect was observed and it was more prominent at higher presentation rates and longer sequence lengths. A recency effect was also observed in most listeners, but no significant effect of presentation rate was found.

1pPPb23. Late negative response revealed language-specific processing of word-level prosodic phonology in the adult brain. Luodi Yu, Yang Zhang (Dept. of Speech-Language-Hearing, Univ. of Minnesota, 164 Pillsbury Dr SE, Minneapolis, MN 55455, zhang470@umn.edu), Suiping Wang, and Jiajing Zeng (School of Psych., South China Normal Univ., Guangzhou, China)

Native speakers develop specialized neural architectures in the processing of auditory patterns found in their language. However, little is known about the neural basis of language-specific processing of phonological patterns beyond the syllable level. The current event-related potential (ERP) study investigated cortical processing of native vs. nonnative prosodic phonology in a disyllabic context. The participants were 18 normal native Chinese-speaking adults who had at least six years of English-as-a-second-language course work in school. The speech stimuli included disyllabic nonsense words spoken in Chinese and in English, which were matched in syllabic structure and digitally edited for matching in overall duration and intensity. The control nonspeech stimuli were hummed versions of the speech stimuli, which preserved the acoustic prosodic features but not the phonetic content. The ERP data were obtained with a passive listening paradigm, showing an enhanced late negative response (LNR) for the native vs. nonnative comparison in the speech condition but not in the nonspeech condition. These results provided the initial evidence that the LNR component can reflect language-specific differences in prosodic phonology beyond auditory processing of the critical acoustic cues at the supra-syllabic level.

1pPPb24. Analysis of the effect of different patologies on the sound transmission through the middle ear by means of a finite-element model. Lucas C. Lobato (Lab. of Acoust. and Vibrations, Federal Univ. of Santa Catarina, Av. Roraima n° 1000, Santa Maria, Rio Grande do Sul 97105-900, Brazil, lucascostalobato@gmail.com), Stephan Paul (Lab. of Acoust. and Vibrations, Federal Univ. of Santa Catarina, Joinville, Brazil), Julio A. Cordioli (Lab. of Acoust. and Vibrations, Federal Univ. of Santa Catarina, Florianópolis, SC, Brazil), and Evandro Maccarini (Lab. of Acoust. and Vibrations, Federal Univ. of Santa Catarina, Florianopolis, Brazil)

The middle ear is part of the human auditory system and its function is to transmit acoustical energy to the inner ear by means of mechanical vibrations. Finite-Element (FE) models of the human middle ear have been studied since 90s, making it possible to evaluate its dynamic behavior in different ways. In this work, an accurate FE model of the human middle ear is described and used for assessing the relationship between the reduction of sound transmission through the ossicular chain and hearing loss due to three different pathologies: tympanic membrane perforation, stapedial tendon ossification and stapes fracture. For these three pathologies, case studies from the literature with experimental hearing loss data are reviewed and clinical results are compared to the FE model predictions. In all cases a good relationship between sound transmission reduction predicted by the FE model and the hearing loss measured through audiometric tests were found. Results show that an accurate FE model of the human middle ear can be a powerful tool for clinical applications and research in otology, allowing to predict the effect of middle ear pathologies and to evaluate the results of corrective surgeries on hearing.

1pPPb25. Human inference of force from impact sounds: Perceptual evidence for inverse physics. James Traer and Josh McDermott (Brain and Cognit. Sci., MIT, 77 Massachusetts Ave., Cambridge, MA 02139, jtraer@mit.edu)

An impact sound is determined both by material properties of the objects involved (e.g., mass, density, shape, and rigidity) and by the force of the collision. Human listeners can typically estimate the force of an impact as well as the material which has been struck. To investigate the underlying auditory mechanisms we played listeners audio recordings of two boards being struck and measured their ability to identify the board struck with more force. Listeners significantly outperformed models based on simple acoustic features (e.g., signal power or spectral centroid). We repeated the experiment with synthetic sounds generated from simulated object resonant modes and simulated contact forces derived from a spring model. Listeners could not distinguish synthetic from real recordings and successfully estimated simulated impact force. When the synthetic modes were altered (e.g., to simulate a harder material) listeners altered their judgments of both material and impact force, consistent with the physical implications of the alteration. The results suggest that humans use resonant modes to infer object material, and use this knowledge to estimate the impact force, explaining away material contributions to the sound.

1pPPb26. A framework for evaluating perceptual interactions of various dimensions of sound for data sonifications. Samuel Chabot and Jonas Braasch (School of Architecture, Rensselaer Polytechnic Inst., 40 3rd St, Troy, NY 12180, chabos2@rpi.edu)

Due to its multidimensionality, sound is a powerful medium for presenting data. Various parameters, including pitch, loudness, timbre, rhythm, and spatial location, may be utilized to convey information. Interactions between dimensions can affect perceptions of this information. Experiments evaluating these interactions in the context of sonifications are important in designing more accurate auditory displays. Oftentimes, these experiments focus on a single pair of parameters. This project offers a framework for utilizing different combinations of parameters to streamline evaluations. A user interface allows for the direct manipulation, and subsequent juxtaposition, of these combinations. [Research funded by the Cognitive and Immersive Systems Laboratory at Rensselaer.]

1pPPb27. A study on the reducing stress of music listening by measuring heart rate. Bong Young Kim and myungjin bae (Sori Sound Eng. Lab, Soongsil Univ., 21-1, Garak-ro 23-gil, Songpa-gu, #203, Seoul 05669, South Korea, bykim8@ssu.ac.kr)

Humans are subject to many types of stress. Stress affects various nervous system transit hormone systems and weakens the immune system and weakens the autonomic nervous system, which slows the activities of various organs. It is already well known that those who overcome stress well have a strong immune system and overcome disease. It also achieves many successful results. This fact indicates how important it is to reduce stress. Music listening is one of the great ways to reduce stress. Music listening is known to stimulate the autonomic nervous system and the motor center to reduce complaints, edgy, and tension. Music listening also has an effect on mental treatment such as post-traumatic stress. In this paper, we tried to find out whether music listening helps to reduce stress through HRV (Heart Rate Variability) SDNN (Standard Deviation N-N Interval) obtained from heart rate measurement. Five subjects measured the heart rate by listening to four experiment song. The SDNN value increased when four of five people listened to all the experiment song. In the remaining one, the SDNN value was lowered for only one experiment song, and for the remaining three songs, the SDNN value was increased. However, the degree of reducing stress was different according to the musical preference of each individual. The results of this experiment showed that listening to music according to taste helps to reduce stress.

1pPPb28. Perceptual consequences of cochlear synaptopathy in middle age. Brooke Flesher, Alexandra Mai, Kelsey Dougherty, Jennifer M. Simpson, Michael G. Heinz, and Hari M. Bharadwaj (Purdue Univ., 715 Clinic Dr., West Lafayette, IN 47906, bflesher@purdue.edu)

It is known that as the body ages, listening in noisy situations becomes increasingly more difficult. Middle-aged people have more difficulty than younger people in complex listening environments, even with similar audiometric thresholds. One hypothesis for why this might be is age-related cochlear synaptopathy. Cochlear synaptopathy has been robustly observed to occur with normal aging in animals, and in post-mortem studies of human temporal bones harvested at different ages. However, there are currently no clinical assessments to measure cochlear synaptopathy or the perceptual consequences of this phenomenon. Currently, we are evaluating a battery of clinical and laboratory measures by comparing young and middle-aged listeners with matched audiometric thresholds within the normal range. The battery includes both objective physiological measures that correlate with synaptopathy in animal models, and perceptual measures that hypothetically may be sensitive to synaptopathy-mediated loss of redundancy in afferent coding. Here, we discuss our initial findings within these parameters.

1pPPb29. Testing for evidence of cochlear synaptopathy in normal-hearing young adults with varying noise exposure history. Brandon T. Paul (Evaluative Clinical Sci., Sunnybrook Res. Inst., 2075 Bayview Ave., Toronto, ON M4N 3M5, Canada, brandon.paul@sunnybrook.ca), Natalie Tran, Sajal Waheed, Larry E. Roberts (Psych., Neurosci., & Behaviour, McMaster Univ., Hamilton, ON, Canada), and Ian C. Bruce (Elec. & Comput. Eng., McMaster Univ., Hamilton, ON, Canada)

Loss of high-threshold cochlear synapses following mild noise exposure is suggested to degrade amplitude modulation (AM) encoding in cases where hearing thresholds are normal. However, the relationship of AM encoding to noise exposure history has not been consistent in previous studies. We investigated this relationship in young adults with normal audiograms in two studies using different methods. Study 1 (N = 25) measured

the ~80 Hz electrophysiological envelope following response (EFR) and behavioural AM detection thresholds. Both measures were taken in quiet and in a narrowband background noise designed to attenuate low-threshold synapse contributions. When subjects were divided into two groups based on their noise exposure history, subjects with more noise exposure had smaller EFRs ($p = 0.0198$). AM detection was also poorer in these subjects, but this difference fell short of significance ($p = 0.067$). Study 2 (ongoing) also measures the EFR but employs an additional wider band of background noise to attenuate possible off-frequency contributions of low threshold fibers to AM coding. In addition, AM discrimination is tested instead of AM detection. The question is whether these modifications will reveal a more robust effect of noise exposure history on AM coding than did Study 1. [Work supported by NSERC of Canada.]

1pPPb30. Brain potentials to speech perception in noise for patients with schizophrenia. Lin Mi (Dept. of Psychiatry, The Affiliated Brain Hospital of Guangzhou Medical University (Guangzhou Huiai Hospital), 36 Mingxin Rd., Liwan District, Guangzhou, China, linmi83@hotmail.com), Le Wang, Shenglin She, Haijing Li (The Affiliated Brain Hospital of Guangzhou Medical Univ. (Guangzhou Huiai Hospital), Guangzhou, China), Chang Liu (Dept. of Commun. Sci. and Disord., the Univ. of Texas at Austin, Austin, TX), and Yingjun Zheng (The Affiliated Brain Hospital of Guangzhou Medical Univ. (Guangzhou Huiai Hospital), Guangzhou, China)

Schizophrenia is characterized with a series of cognitive impairment, including basic information processing such as auditory and speech perception. Especially, patients with schizophrenia experienced additional challenge for speech perception in noisy environment. In this study, brain activity in response to speech sounds was measured in quiet and 6-talker babble conditions from 25 schizophrenia patients and 20 healthy subjects. A double-oddball paradigm was employed, in which /ba/ was played as the standard stimulus with /da/ and /bu/ as the deviant stimuli. Results showed that the schizophrenia group had lower amplitude and longer latency of mismatched negativity (MMN) compared with control group in both the quiet and babble conditions. In addition, there was a significant interaction effect between listening condition and listener group, i.e., the 6-talker babble exerted greater disturbance for schizophrenia. Moreover, the analysis of theta band oscillation in the time window of MMN also supported the above results. These findings indicated that the deficits in the pre-attentive automatic auditory processing in the patients with schizophrenia may underpin their speech perception difficulty in the daily noise environment.

1pPPb31. Pupillometry as a measure of auditory cognitive processes and listening effort. Damian Almaraz (Psych., St. Olaf College, Northfield, MN), Brock Carlson (Neurosci., St. Olaf College, Northfield, MN), Elaine Grafelman, Jacob Ingalls, Yadi Quintanilla (Psych., St. Olaf College, 1520 St. Olaf Ave., Northfield, MN 55057, grafel1@stolaf.edu), Hieu-Trung Vu (Neurosci., St. Olaf College, Northfield, MN), and Jeremy Loebach (Psych., St. Olaf College, Northfield, MN)

Although many people with hearing impairments can accurately recognize words and sounds with proper treatments (hearing aids or cochlear implants), they often report fatigue and stress from doing so. These individuals need to exert more auditory cognitive processing effort to hear well, especially in adverse listening environments or when engaging in competing tasks. We hypothesize that the allocation of these auditory cognitive resources, or auditory cognitive "X-Factors," can explain variability in outcome across cochlear implant users. To find the "X-Factors," we use pupillometry as a physiological measure of cognitive load across auditory cognitive domains including short term memory (forward digit span) working memory (backward digit span), cognitive load (Paced Serial Addition Task), listening effort (speech in noise tests) and personality traits (motivation, self-efficacy, and adaptability). A custom-built eye tracking system using Pupil Labs software provides a cost-effective, dynamic, and fully mobile system to collect data in real time. Initial data from aging adults indicate significant correlations between cognitive abilities, pupil dilation, response strategy, and auditory performance. Understanding the "X-Factors" will help inform training and rehabilitation for hearing impaired individuals and cochlear implant users.

1pPPb32. Importance of time-frequency units for sentence recognition as a function of signal-to-noise ratio. Frederic Apoux (Otolaryngol. - Head & Neck Surgery, The Ohio State Univ. - Wexner Medical Ctr., 915 Olentangy River Rd., Columbus, OH 43212, fred.apoux@gmail.com), Brittney Carter (Speech & Hearing Sci., The Ohio State Univ., Columbus, OH), Karl P. Velik (Otolaryngol. - Head & Neck Surgery, The Ohio State Univ. - Wexner Medical Ctr., Columbus, OH), and Eric Healy (Speech & Hearing Sci., The Ohio State Univ., Columbus, OH)

Speech recognition in noise may be viewed as a classification problem, in which the auditory system selects a subset of time-frequency (T-F) units to build a representation of the signal. The present study introduces local signal-to-noise ratio (SNR) importance functions, an approach to determine the contribution to speech intelligibility of T-F units as a function of their SNR. Consistent with previous work from this group on auditory-channel independence, it was hypothesized that T-F units with a relatively equal mixture of signal and noise (around 0 dB SNR) may be the most disruptive for intelligibility because they should be the most difficult to classify. This hypothesis was assessed by measuring the intelligibility of sentences presented in a competing-speech background while discarding a fixed proportion of units all with the same local SNR. The stimuli were either vocoded or left unprocessed to evaluate the influence of fine-structure cues on classification and also help better understand why cochlear implant (CI) users experience greater difficulties in noise. Results were consistent with the ambiguity hypothesis. More importantly, they suggest that discarding the noisiest units may not be the most advantageous way to improve speech intelligibility in noise by CI users.

1pPPb33. The effect of masker type and masker timing on the serial recall of speech. Elin Roverud, Virginia Best, Christopher Conroy, and Gerald Kidd (Dept of Speech, Lang. & Hearing Sci., Bosto Univ., 635 Commonwealth Ave., Boston, MA 02215, emroverud@gmail.com)

Understanding speech in noise involves a complex set of processes including segregation of sources, selection of the target talker, and recognition and comprehension of the ongoing message. Previous studies often have focused on the segregation, selection and recognition processes by measuring the intelligibility of short strings of words presented concurrently with different types of maskers. However, these studies may not fully capture the effects of masking on all aspects of ongoing speech processing. For instance, masking can cause poorer recall for *sequences* of words even if a noise/competitor leads to near-perfect recognition of the individual items [e.g., Cousins *et al.*, 2014, Mem Cogn (42)]. Furthermore, in contrast to the typical design of speech recognition tasks, background noise usually does not cease during response to the message. In the present study, we examined listeners' serial recall of target speech with different maskers presented at different points during the trial. Speech-shaped noise or confusable speech maskers were presented colocated with or spatially separated from the target. The maskers were presented during target presentation, during the response interval, or continuously throughout the trial. Results will compare the influence of different types of masking during encoding and retrieval/response stages of processing.

1pPPb34. Estimating the duration of a refractory period in perceptual learning. Alex E. Clain (Commun. Sci. and Disord., Northwestern Univ., 5445 N Sheridan Rd., Apt. 2207, Chicago, IL 60640, alexanderclain2022@u.northwestern.edu) and Beverly Wright (Commun. Sci. And Disord., Northwestern Univ., Evanston, IL)

Auditory skills improve with practice. Producing across-day learning requires a sufficient number of trials per session; additional trials in the same session produce no additional improvement. Thus, trials appear to integrate within a session to reach a learning threshold, after which there is a refractory period during which trials do not contribute to learning. The refractory period duration must be $<$24 hours because training on consecutive days yields learning each day, but its exact duration is unknown. To narrow the range of possible durations, we trained young-adult, normal-hearing listeners on a basic auditory task (interaural-level-difference discrimination) either for two sessions of sufficient training separated by 30 minutes ($n = 10$) or 10 hours ($n = 10$), or for just one session ($n = 15$), and assessed their learning on the next day. The 30-minute group learned no more than the single-session group, indicating the second session did not

aid learning, whereas the 10-hour group learned more than the other two groups, indicating the second session provided additional benefit. These results suggest that the refractory period lasts >30 minutes, but <10 hours, and does not require sleep to reset. Knowledge of the refractory period duration will improve the efficiency of perceptual training for typical and clinical populations.

MONDAY AFTERNOON, 7 MAY 2018 GREENWAY C, 1:00 P.M. TO 3:00 P.M.

Session 1pSA

Structural Acoustics and Vibration: General Topics in Structural Acoustics and Vibration

Benjamin Shafer, Chair
Technical Services, PABCO Gypsum, 3905 N 10th St, Tacoma, WA 98406

Contributed Papers

1:00

1pSA1. Characterizing stress corrosion cracking in stainless steel using nonlinear resonant ultrasound spectroscopy. Stephen M. Hogg and Brian E. Anderson (BYU Dept. of Phys. & Astron., Brigham Young Univ., N283 ESC, Provo, UT 84602, jr.hoggcube@gmail.com)

During disposal, spent nuclear fuel rods are placed into stainless steel containers and may be stored this way for decades. In order to ensure there is as little radiation leakage as possible, the structure of the stainless steel needs to be evaluated and possible damage detected. In particular, one type of damage these containers are particularly susceptible to is stress corrosion cracking (SCC). Traditional (linear) evaluation methods can detect open cracks but often a closed portion of the crack extends further, compromising the air tight seal sooner. To this end, it has been proposed to use Nonlinear Resonant Ultrasound Spectroscopy (NRUS) to quantify the degree of cracking in a structure. In order to prove this concept, cylindrical 304L stainless steel rods were immersed in a heated 42% MgCl2 solution to induce SCC in an accelerated manner to simulate the damage that is likely to occur on the actual containers. The rods were removed after different lengths of exposure. NRUS measurements were then conducted for longitudinal modes in the rods.

1:15

1pSA2. Acoustic reflection and transmission by sub-critical and super-critical elastic partitions with structural discontinuities forced by multiple-angle oblique broadband sound waves. Mauricio Villa, Donald B. Bliss, and Linda P. Franzoni (Dept. of Mech. Eng. and Mater. Sci., Duke Univ., Box 90300 Hudson Hall, Durham, NC 27708, mauricio.villa@duke.edu)

An analysis is presented for acoustic reflection and transmission from an infinite fluid-loaded flexible barrier with spatially periodic discontinuities. The plate, with similar or dissimilar acoustic fluids on both sides, is excited by an oblique wave incident acoustic field. The fully-coupled structural/acoustic problem is treated by Analytical-Numerical Matching (ANM) which improves the numerical accuracy and convergence rate. Periodic spatial discontinuities, modelled by various boundary conditions, create deviations from specular directivity. For super-critical structural wave speeds, the scattering is characterized by a distribution of reflection and transmission angles centered around the Mach Angles. For sub-critical structural waves, energy redistribution is most pronounced at the structural resonances. These results are compared to a baffled finite barrier analyzed by approximate means. For the subcritical finite barrier, a fluid loading correction is introduced to the structural wavenumber to approximate the effects of the fluid coupling and radiation damping. For subsonic flexural waves, the radiation is analyzed in terms of the wavenumber transform and interpreted as a series of edge singularities dependent on the boundary conditions. For the super-critical configuration, the effects of the fluid loading are introduced by approximating the structural wavenumber from the dispersion relation of the free response of the unbounded structure.

1:30

1pSA3. Remote acoustic detection of localized delamination in vibrating composite layered plates. Tyler J. Flynn and David R. Dowling (Mech. Eng., Univ. of Michigan, Ann Arbor, 1231 Beal Ave., Ann Arbor, MI 48109, tjayflyn@umich.edu)

Adhesively bonded, layered composite materials are ubiquitous in naval, aerospace, and automotive applications. One common challenge with such materials is the presence of delamination discontinuities at the interlaminar bonds which can have severe adverse effects to structural health and system performance. Worse yet, delamination is often visually indiscernible as it occurs below visually-obstructing surface layers, making non-visual defect detection techniques attractive in comparison. In this presentation, remote acoustic detection of localized delamination in a 0.3-m-square composite laminate plate subject to 100-6000 Hz noisy broadband forcing will be investigated. Remote measurements of the radiated acoustic field from composite plates with and without known localized delamination are collected with a 15-element microphone array. Here, statistical comparisons of spectral amplitudes between plates with and without damage are then used to evaluate remote acoustic detection performance. Additional signal processing steps are applied to the raw array data to improve performance robustness in the reverberant laboratory test setting. Considerations for detection performance include minimum detectable size of delamination, robustness to receiving-array location, and sensitivity to ambient noise. [Sponsored by NAVSEA through the NEEC and by the US DoD through an NDSEG Fellowship.]

1:45

1pSA4. Acoustic methods for regionalizing an impact force acting on a structure. W. S. Shepard and Jacob Davis (Mech. Eng., The Univ. of Alabama, Box 870276, Tuscaloosa, AL 35487, jadavis15@crimson.ua.edu)

It is often desired to know the location and magnitude of a force acting on a structure. Unfortunately, it is not always possible to install a sensor at the force location, particularly when that location is unknown. A structure subjected to an impact has many different vibrational modes that are excited to different levels based on the excitation location. These vibrations decay with time depending on their different rates of modal damping and the associated acoustic radiation characteristics. In this work, acoustic signatures for

a number of structure impact locations were measured, normalized relative to the force magnitude, and processed to examine the ability to correlate the acoustic signal to the force impact location. Various processing techniques, such as the Short Time Fourier Transform, were considered. A primary interest focused on the ability of single-number metrics to aid in identifying the force location. For the experiments, a football helmet structure was used and multiple impact locations were tested. The ability of these acoustic signatures, including those processed into single-number metrics, to aid in identifying the impact location was assessed. The location resolution for which these methods were effective was also evaluated.

2:00

1pSA5. A study on the oxygen generation in water molecules using sound wave. Yi Eun Young, Uk-Jin Song, and myungjin bae (Information and TeleCommun., Soongsil Univ., Soongsil Univ., Sangdo 1-dong, Dongjak-gu, Seoul 156-743, South Korea, go6051@naver.com)

Life is living in a different place. There are mammals and birds living in the air through oxygen. There is a life extension using oxygen present in the water. There are fish and mammals in the water that use oxygen in the water. The whales, belonging to the mammals in the sea, breathe out of the water at regular intervals and then back into the water. Fish do not hinder the extension of life without these actions. This study uses speakers as means for generating oxygen in water molecules. It is a study to decompose oxygen in water by using the dense wave of the speaker sound wave. It is a study that removes oxygen by affecting the bonding structure of water molecules through the vibration of sound waves. Unlike other chemical methods, this characteristic can generate oxygen without generating hydrogen.

2:15

1pSA6. Inter-digital transducers activated acoustic streaming in viscous liquid. Chuyi Chen, Peiran Zhang (Mech. Eng. and Mater. Sci., Duke Univ., 144 Hudson Hall, Box 90300, Durham, NC 27708, cc507@duke.edu), Zhangming Mao (Dept. of Eng. Sci. and Mech., The Penn State Univ., Durham, North Carolina), Yuyang Gu (Mech. Eng. and Mater. Sci., Duke Univ., Durham, NC), Francesco Costanzo (Dept. of Eng. Sci. and Mech., The Penn State Univ., University Park, PA), and Tony Jun Huang (Mech. Eng. and Mater. Sci., Duke Univ., Durham, NC)

Acoustic streaming induced by surface acoustic waves (SAWs) has been widely utilized in various medical and biology applications. For most of the applications, the microfluidic channel is attached to a piezoelectric substrate where the inter-digital transducers (IDTs) are fabricated on it to activate acoustic waves. The acoustic waves would propagate along the substrate surface and generate the "leaky wave" in the channel which causes the acoustic streaming. However, few studies are conducted to analyze the vibration mode of the substrate in the IDTs area and the following phenomenon (e.g., acoustic streaming generated by substrate vibration). In this study, we demonstrate a novel method which includes both simulation and experiment results to identify the substrate vibration and corresponding streaming while the liquid is loading upon the IDTs directly. With this method, we show that the acoustic streaming on top of the IDTs area has a "stagnation point" with a velocity potential to trap and manipulate the objects floating on the surface of the liquid, which is promising in many contactless biomedical applications.

2:30

1pSA7. Nonlinear resonant ultrasound spectroscopy using electromagnetic excitation. Joshua F. Gregg and Brian E. Anderson (Dept. of Phys. & Astron., Brigham Young Univ., BYU Dept. of Phys. and Astron., N283 ESC, Provo, UT 84602, joshygregg1@gmail.com)

Nonlinear resonant ultrasound spectroscopy (NRUS) is a method that can be used for detecting microcracks in structures. NRUS detects shifts in resonance frequencies versus excitation amplitude. Excitation of the sample is typically done with a piezoelectric transducer. We have been applying an electromagnetic excitation developed for resonant ultrasound spectroscopy [S. Garrett, J. Acoust. Soc. Am. **88**(1), 210–221 (1990)] as an alternative for NRUS excitation. It involves gluing a coil of wire onto the end of a rod-shaped sample and placing it in a magnetic field. By controlling which part of the coil is in the strongest part of the magnetic field, we can control the principle direction of the driven oscillations in the rod. This method allows us to selectively excite longitudinal, torsional, and bending vibrations and measure the nonlinear properties of the sample for each type of vibration. We have applied this electromagnetic method to measuring the nonlinear properties of Berea sandstone, and we plan to use it to detect stress corrosion cracking in stainless steel samples. The purpose of this presentation is to illustrate the usefulness of the electromagnetic method.

2:45

1pSA8. Thermoacoustic instability in solid media. Haitian Hao, Carlo Scalo (Mech. Eng., Purdue Univ., Herrick Labs, 177 S. Russell St, Rm. 1007, West Lafayette, IN 47906, haoh@purdue.edu), Mihir Sen (Aerosp. and Mech. Eng., Univ. of Notre Dame, Notre Dame, IN), and Fabio Semperlotti (Mech. Eng., Purdue Univ., West Lafayette, IN)

The exploration of thermoacoustic oscillations can be traced back to the 19th century when Sondhauss (1850) performed an experimental investigation that marked the birth of modern thermoacoustic research. Since then, several practical applications of the basic thermoacoustic effect were explored and ultimately led to the design of engineering devices such as heat engines or refrigerators. This fascinating mechanism was believed to occur only in fluids while our study shows theoretical and numerical evidence of the existence of thermoacoustic oscillations also in solid media. Although this mechanism shares common aspects with the more familiar thermoacoustics of fluids, our analysis shows that the underlying physical mechanism presents some important and non-trivial differences with the traditional theory. The discovery and validation of this mechanism may lead to new ideas enabling the development of the next generation of ultra-compact and robust solid-state thermoacoustic devices for direct thermal-to-mechanical energy conversion, and vice versa.

Session 1pSCa

Speech Communication: South Asian Languages I

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Linguistics, Macalester College, 1600 Grand Ave, St. Paul, MN 55105

Indranil Dutta, Cochair
Department of Computational Linguistics, English and Foreign Languages University, Tarnaka, Osmania University Campus, Hyderabad 500605, India

Chair's Introduction—1:00

Contributed Papers

1:05

1pSCa1. Acoustic characteristics and perception of voiced and voiceless nasals in Mizo and Angami. Pamir Gogoi (Dept. of Linguist, Univ. of Florida, Gainesville, FL 32611, pgogoi@ufl.edu)

This study compares and contrasts acoustic characteristics (e.g., nasal murmur's formant and anti-formant frequencies, vowel formant frequencies, bandwidths, spectral tilts, etc.) of voiced and voiceless nasals in Mizo and Angami, Tibeto-Burman languages spoken in North-East India. Mizo contrasts voiced and voiceless nasals at the bilabial, alveolar, and velar places of articulation, whereas Angami contrasts them at four different places of articulation: bilabial, alveolar, velar, and palatal. Similar to voiceless nasals previously described for Burmese (Danstuji,1984; Bhaskararao & Ladefoged,1991), in Mizo, voiceless nasal murmur is followed by a period of voiced nasal murmur as it transitions into the following vowel, while it remains voiceless throughout in Angami. Results of a perception experiment also suggests that the voiced nasal murmur portion in Mizo voiceless nasals aided the identification of its place of articulation, but a heavier perceptual load is placed on the following vowel for place identification of voiceless nasals in Angami. **References:** Dantsuji, M. (1984). "A study on voiceless nasals in Burmese." Bhaskararao, P., & Ladefoged, P. (1991). "Two types of voiceless nasals," *J. Int. Phonet. Assoc.*, ***21***(2), 80–88.

1:20

1pSCa2. Stop laryngeal contrasts of endangered languages of Northern Pakistan. Qandeel Hussain and Jeff Mielke (Dept. of English (Linguist Program), North Carolina State Univ., 221 Tompkins Hall, Campus Box 8105, Raleigh, NC 27695-0001, qhussai@ncsu.edu)

Iranian, Indo-Aryan (Dardic), and Tibeto-Burman languages of Northern Pakistan are characterized by 2-, 3-, and 4-way laryngeal contrasts at four or five places of articulation (e.g., Balti, Kalasha, Wakhi, etc.). However, the number of stop laryngeal contrasts, their phonetic realization, and the exact place of articulation of retroflex stops (apical or sub-apical) are still controversial in most of the endangered languages of Northern Pakistan. The current study provides the first detailed acoustic and articulatory (Ultrasound for coronals and velars and aerodynamic for all stops) description of the stop laryngeal contrasts of six languages of Northern Pakistan: Balti (Tibeto-Burman), Burushaski (Isolate), Kalasha, Khowar, Shina (Indo-Aryan, Dardic), and Wakhi (Iranian, Pamir). Given that there are languages belonging to different families, we may expect differences in the phonetic realization of stop laryngeal contrasts and the articulation of retroflexes across languages. The acoustic, articulatory, and aerodynamic data were collected in Chitral and Gilgit, Pakistan in 2017. A total of 33 participants were recorded (Balti: 5; Burushaski: 4; Kalasha: 9; Khowar: 5; Shina: 6; and Wakhi: 4). The results of this study will help understand the diverse nature of stop consonants in the endangered languages of Northern Pakistan.

1:35

1pSCa3. Evidence for active cavity expansion through advanced lingual place of constriction in voiced geminates. Ushasi Banerjee (The English and Foreign Lang. Univ., Hyderabad, Telangana 500007, India, banerjeeushashi@gmail.com), Indranil Dutta, and Meghavarshini Krishnaswamy (Computational Linguist, The English and Foreign Lang. Univ., Hyderabad, Telangana, India)

Geminate voicing requires both active and passive cavity expansion in order to delay the equalization of subglottal and supraglottal pressure (Ohala 1983, Ham 2001). Studies report differences in stop closure duration (Lahiri & Hankamer 1988) and pre-geminate vowel duration as cues to the laryngeal contrast. Following Schroeder (1967), Mermelstein (1967), and Iskarous (2010), we present results from formant transitions in the VC:V context and the anti-symmetric Fourier component coefficient, $a_n = -2\ (F_n - f_n)/f_n$, to partially recover the area function of the vocal tract behind the geminate constriction. Voiced geminates exhibit significantly higher and positive coefficient values than their voiceless counterparts which implies that measured F_n values for voiced geminates are lower than their neutral tract formant values. This finding indicates that voiced geminates have a more anterior constriction compared to their voiceless counterparts. Without ruling out the role of passive cavity expansion through the cheek muscles, we conclude that in order to delay the pressure equalization the lingual constriction in voiced geminates is maintained at an anterior location compared to voiceless geminates in Bangla. Results from the slopes of first order F_2 locus-equations are also presented to measure the differences in coarticulatory resistance between the voiced and voiceless geminates.

1:50

1pSCa4. A contrastive place of articulation between palatals and velars: An ultrasound study of Malayalam palatal-velar nasals. Dominique A. Bouavichith, Savithry Namboodiripad (Linguist, Univ. of Michigan, 611 Tappan St, #455C, Ann Arbor, MI 48109, dbouavichith@gmail.com), and Marc Garellek (Linguist, Univ. of California, San Diego, La Jolla, CA)

Malayalam (Dravidian) has been described as having seven contrastive nasal places of articulation: labial, dental, alveolar, retroflex, palatal, velar, and a "seventh nasal," sometimes called palatal-velar /ŋj/ (Asher & Kumari 1997; Namboodiripad & Garellek 2017). Nasal inventories of this size are cross-linguistically rare, and the palatal-velar nasal has only been proposed in a handful of languages worldwide (e.g., Irish, Yanyuwa, and Awngi). Indeed, its status in Malayalam could be deemed peripheral; it is infrequent and occurs in limited phonological contexts. Namboodiripad & Garellek (2017) claim that, though several consonants undergo palatalization in Malayalam, this palatal-velar nasal is dynamically stable and not post-palatalized. Although it is distinct in format transitions from the non-palatalized velar nasal, the articulatory basis for the distinction between this palatal-velar nasal and a velar-glide cluster is not clear. The present study uses ultrasound imaging to determine if this seventh nasal is in fact dynamically stable. Ultrasound data were collected from one native speaker of Malayalam; the placement and timing of lingual contact with the palate were examined. Discussion will focus on how the palatal-velar differs in terms of place and timing of articulations from both palatal and velar nasals, as well as from other palatalized consonants.

MONDAY AFTERNOON, 7 MAY 2018 NICOLLET A, 2:15 P.M. TO 5:15 P.M.

Session 1pSCb

Speech Communication: South Asian Languages II (Poster Session)

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Indranil Dutta, Cochair
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All posters will be on display from 2:15 p.m. to 5:15 p.m. To give contributors in this session an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 2:15 p.m. to 3:30 p.m. and authors of even-numbered papers will be at their posters from 3:30 p.m. to 5:15 p.m.

Contributed Papers

1pSCb1. Indic and Dravidian retroflexes: An ultrasound investigation. Amanda Bohnert, Samantha Myers, James Smith, and Kelly Berkson (Lingustics, Indiana Univ., 1021 E. Third St., Dept. of Linguist - Mem 322E, Bloomington, IN 47405, kberkson@indiana.edu)

The languages of South Asia somewhat famously contain what might be deemed canonical retroflexes. These are often described as involving an extreme articulation with sub-apical post-alveolar or palatal contact, and imaging studies of varying types (electropalatography, MRI, ultrasound, etc.) suggest that this is an accurate characterization for Dravidian retroflexes (Kochetov *et al.* 2014, Krull and Lindblom 1996, Narayanan *et al.* 1999, Narayanan *et al.* 2004, Scobbie *et al.* 2013). Indic retroflexes reportedly involve less extreme articulations (Ladefoged and Bhaskararao 1983), but imaging data for Indic languages remains surprisingly rare. This raises an interesting question: how do Indic and Dravidian retroflexes compare articulatorily? We explore this question by presenting ultrasound imaging of retroflexes in several Indic (Marathi, Hindi) and several Dravidian (Kannada, Telugu) languages. Results indicate that all languages contain at least some of the extreme articulations often associated with South Asian retroflexes, though the within-speaker variability noted in previous work (e.g., Arsenault 2017) is also on display.

2:15

1pSCb2. Articulatory and acoustic investigation of coronals in Hakha Chin. James Smith, Stefon Flego, and Kelly Berkson (Lingustics, Indiana Univ., 1021 E. Third St., Dept. of Linguist - Mem 322E, Bloomington, IN 47405, kberkson@indiana.edu)

Coronal obstruents are profoundly common typologically, occurring in most or all of the world's known languages. Very few languages, however, contrast coronal obstruents at the dental and alveolar places of articulation. Hakha Chin—also known as Lai—is a Tibeto-Burman language spoken in western Myanmar that is reported to do so. Very little phonetic research on Chin exists, but Maddieson and Van Bik (2004) provide acoustic and static articulatory data (palatography and linguography) suggesting that the contrast is between a lamino-dental series and an apico-alveolar series. We follow up on that research using dynamic, volumetric articulatory imaging in the form of 3D/4D ultrasound. Articulatory and acoustic data from two native speakers of Chin are presented in order to contribute to a more thorough understanding of this typologically uncommon contrast.

1pSCb3. Acoustic correlates of the four-way laryngeal contrast in Marathi stops. Olga Dmitrieva (Purdue Univ., 640 Oval Dr., Stanley Coulter 166, West Lafayette, IN 47907, odmitrie@purdue.edu) and Indranil Dutta (The English and Foreign Lang. Univ., Hyderabad, Telangana, India)

As many of its geographical and typological neighbors, Marathi demonstrates a four-way laryngeal contrast in stop consonants, which includes plain voiced and voiceless stops as well as aspirated voiceless stops and typologically rare aspirated (breathy) voiced stops. The present study investigates the acoustic correlates of the laryngeal contrast in word-initial velar stops in Marathi. Thirty three native speakers of Marathi were recorded in Mumbai, Maharashtra, India. Positive and negative Voice Onset Time (VOT), fundamental frequency at the onset of voicing (onset f0), and spectral intensity as a measure of breathiness during the following vowel were investigated as correlates of laryngeal distinctions. Preliminary results show that all four stop categories are well-distinguished via VOT and onset f0 measures. Initial spectral intensity measurements suggest that breathiness in Marathi stops is largely contained in the release and does not extend far into the following vowel, unlike what has been found in Hindi (Dutta, 2007). The results will be discussed with respect to the typology of voicing contrast across languages, specifically the relevance of VOT distinctions and the use of onset f0, and compared to relevant findings for typologically similar languages.

1pSCb4. VOT in Tibetan is conditioned by tone. Christopher A. Geissler (Linguist, Yale Univ., 370 Temple St, New Haven, CT 06511, christopher.geissler@yale.edu)

Several South Asian languages exhibit interactions among tone and consonant phonation, including Tamang (Mazoudon 2014) and Gurung (Ronkos 2015). In Central Tibetan, oral stops present a four-way contrast between two voicing/aspiration categories split across two lexical tones, high and low/rising. Across tones, previous analyses propose either a two-way contrast between unaspirated and aspirated stops (e.g., Dawson 1980), or a three-way contrast between voiced, unaspirated, and aspirated series (e.g., Denwood 1999), though acoustic measurements are lacking. This study recorded 19 native speakers of Tibetan in Kathmandu, Nepal, producing words exemplifying the phonological contrasts in stops and high/low tones. VOT measurements showed similar values for the unaspirated stops in both tonal environments, but a longer VOT in high-tone aspirated stops (46 ms) than low-tone aspirated stops (30 ms), or three different positive VOT values. That this three-way surface contrast cross-cuts tonal categories suggests an underlying two-way aspiration contrast. The apparent contrast between middle-VOT and long-VOT series results from the presence of a phonologically-specified high tone, which causes a lengthened VOT. This provides evidence for the conditioning of apparently-segmental characteristics by tone.

1pSCb5. Within-category variation in production of Hindi and English stop consonants. Ivy Hauser (Linguist, Univ. of Massachusetts Amherst, 650 North Pleasant St, Amherst, MA 01060, ihauser@linguist.umass.edu)

This paper addresses the broad question of how phonological structure affects phonetic production using within-category variation of multiple phonetic cues in Hindi and English stop consonants. The intuitive hypothesis, formalized in Dispersion Theory (Lindblom, 1986) is that languages that have more phonological contrasts along a single phonetic dimension should have relatively less within-category variation along that dimension. Hindi (four stop contrasts per place of articulation) speakers are therefore predicted to exhibit less within-category variation in production relative to English (two contrasts per place) speakers. This was observed in the results of Hauser (2016) who examined VOT variation of voiceless aspirated stops in these languages. The present study builds on those results and examines F0 as a second cue to the voicing contrast. The F0 results do not conform to the prediction that Hindi speakers should exhibit less variation. There was no difference in F0 variation between the Hindi and English speakers, yet there were significant differences in the amount of within-category F0 variation among the individual Hindi speakers (Levene's Test $p < 0.001$). It is argued that this is because F0 acts as a secondary cue to the voicing contrast. Speakers can exhibit greater and individualized levels of variability in F0 without adding perceptual confusion because the primary cue (VOT) remains distinct.

1pSCb6. Phonation type in vowels of heritage and native Gujarati speakers. Kiranpreet Nara (Linguist, Univ. of Toronto, 100 St. George St., Toronto, ON M5S 3G3, Canada, kiranpreet.nara@mail.utoronto.ca)

Gujarati, an Indo-Aryan language, contrasts phonation type in vowels: [baɾ] 'twelve' and [ba̤ɾ] 'outside' (Pandit, 1957) and the current study looked at Gujarati vowels using acoustic and electroglottographic (EGG) analyses. The participants were native and heritage Gujarati speakers. Heritage speakers were born in Canada or arrived in Canada before seven years of age and learned Gujarati as their first language. Due to limited access to their first language, such as listening more than speaking the language and using Gujarati exclusively at home, it was expected that there might not be a significant difference between the heritage speakers' breathy and modal vowel productions. This study determined if the acoustic and EGG parameters that differentiate breathy from modal vowels were the same or different for both speaker groups. Some of the parameters used to distinguish phonation type were H1-H2, H1-A1 ('A1' refers to amplitude of the first formant), harmonic-to-noise ratio, and contact quotient. Measurements were made using VoiceSauce (Shue *et al*., 2009) and EGGWorks (Tehrani, 2009). ANOVA analyses conducted on vowels /a e o/ indicated that fewer parameters distinguish phonation type for heritage than native speakers and the difference between breathy and modal vowels for heritage speakers is smaller in magnitude. The results thus far indicate that heritage speakers acquire a reduced phonation type contrast.

1pSCb7. Pre-coda vowel duration in Nepali. Martha Schwarz (Linguist, UC Berkeley, 6449 Colby St, Oakland, CA 94618, martha_schwarz@berkeley.edu)

Many previous studies have found that vowels are longer before voiced consonants than voiceless (Chen 1970). Fewer studies have examined vowel duration preceding aspirated vs. unaspirated stops, and results are inconsistent (Maddieson & Grandour 1976, Ohala & Ohala 1992). Durvasala and Luo (2012) find that both voicing and aspiration have a lengthening effect on the preceding vowel in Hindi, and that the voiced aspirated class has the longest preceding vowel duration. In the present study, I measure the duration of vowels preceding word final codas in Nepali, an Indo-Aryan language with the same four-way contrast as Hindi. A phonetic study of final stop realization has not yet been reported in the literature, though Nepali is described as variably neutralizing voiced aspirated to the plain voiced. I present an acoustic analysis of cues in word-final position including voicing, burst, and preceding vowel duration. I find evidence of neutralization of voiced and voiced aspirated stops in terms of aspiration duration, but also find results consistent with Durvasala and Luo (2012), that the voiced aspirated stops have longer preceding vowel than the voiced. The difference in vowel duration between the voiced and voiced aspirated classes suggests incomplete neutralization of voiced aspirated to voiced.

1pSCb8. Do Hindi speakers of English realize the acoustic cues for American English /v/ and /w/ contrast? Vikas Grover (Commun. Disord. and Deafness, Kean Univ., New Jersey, NJ 07083, vgrover@kean.edu), Valerie Shafer (Speech-Language-Hearing Sci., The Graduate Ctr., CUNY, New York, NY), D. H. Whalen (Speech-Language-Hearing Sci., CUNY Graduate Ctr., Haskins Labs. and Yale Univ., New York, NY), Erika Levy (Commun. Sci. and Disord., Teachers College, Columbia Univ., New York, NY), and Stephanie Kakadelis (Linguist Program, The Graduate Ctr., CUNY, NY, NY)

The Hindi phonological inventory does not include a [v-w] contrast. In a perception study, Hindi speakers identified /v/ and /w/ in American English (AE) words with approximately 50% accuracy (Grover *et al*., 2016). The present study compares AE /v/ and /w/ productions in bilingual Hindi-English speakers' with those of AE speakers' in nonsense words. Two AE listeners annotated recordings of word-initial and word-medial tokens to label F2 onset and presence of frication. F2 onset frequency and F2 slope were calculated. Hindi speakers were divided into two groups, those with length

of residence (LOR) in the US of more than five years, and those who reside in India. Results revealed a significant difference between the AE group and both Hindi groups for F2 onset frequency and F2 slope. There was no significant difference between Hindi groups. The AE group had frication for /v/ tokens, but not for /w/. Both Hindi groups produced frication at about the same rate for /v/ and /w/. The finding of no significant difference between the Hindi groups indicates that exposure to the /v/-/w/ contrast in the US did not improve Hindi speakers' production. This further indicates a need to design targeted training for this contrast.

1pSCb9. Consonant-f_0 interaction in Malayalam stops: Automatic or controlled? Indranil Dutta and Molly Varghese (Dept. of Computational Linguist, The English and Foreign Lang. Univ., Hyderabad, Telangana 500007, India, indranil@efluniversity.ac.in)

Voicing is known to perturb f_0 in the following vowel (House and Fairbanks, 1953). This lowering has been attributed to biomechanical constraints (Svantesson and House, 2006). On the basis of evidence from languages where the [voice] feature does not manifest itself phonetically, yet is accompanied by f_0 lowering, it has been argued that f_0 lowering serves to enhance the phonological [voice] contrast. We present results from an acoustic study of f_0 perturbation in Malayalam, which exhibits predictable voicing in lexical items from its Dravidian stratum. Underlying voiceless stops only appear initially, and are voiced intervocalically. Malayalam also contains a large Sanskrit stratum lexicon, where phonetic voicing is tied to its stratal affiliation. We obtained a significant effect of laryngeal setting on z-score normalized f_0 at vowel onset. These results suggest voicing is significant in determining the onset f_0 of the vowel, regardless of the phonological status of voicing in a language We conclude that f_0 lowering has physiological and biomechanical bases. In that sense, f_0 lowering is an automatic consequence of the biomechanics of voicing. At the representational level, such lowering could indeed be enlisted for enhancement, however, that is not the only, or the primary purpose of f_0 perturbation.

1pSCb10. Relation of musical experience to perceiving and producing aspirated Hindi consonants. Anmol Gupta and Terry L. Gottfried (Psych., Lawrence Univ., 711 E. Boldt Way, Appleton, WI 54911, gottfrit@lawrence.edu)

Past research shows that those with musical experience often perceive and produce unfamiliar speech contrasts better than non-musicians (e.g., Gottfried, 2007; Lee & Hung, 2008; Marie, Delogu, Lampis, Belardinelli, & Besson, 2011; Slevc & Miyake, 2006). The present research seeks to clarify the mechanisms which underlie this relationship through a "test-teach-test" paradigm (see Vlahou, Protopapas, & Seitz, 2012). In the first session, native speakers of American English with and without conservatory training were tested on their discrimination of Hindi sentences that contained either an aspirated (e.g., [b^h]) or unaspirated (e.g., [b]) consonant ("test"). In a second session, participants completed training on producing and perceptually discriminating the aspirated and unaspirated Hindi consonants ("teach"). In the third session, participants were tested on their ability to perceive and to produce these unfamiliar consonants in the same and different phonetic contexts ("test"). Preliminary results suggest that all participants benefited from the training, but Improvement was markedly greater for musicians than for non-musicians. Current studies are further testing additional musicians and non-musicians to evaluate the extent which musical experience influences perception and production of these unfamiliar Hindi phonemes.

1pSCb11. Effect of contrastive focus on vowel duration in Bangla. Bitapi Ghosh (Dept. of Linguist and Contemporary English, The English and Foreign Lang. Univ., Amrita Pritam Hostel, Hyderabad, Telangana 500007, India, ghoshbitapi92@gmail.com)

The present study looks at Bangla, spoken near Kolkata, West Bengal, which has got the status of standard Bangla in present era, belongs to the Indo-Aryan branch of the Indo-European language family. Vowel length distinction is not phonemic in Bangla. However, the impact of various prosodic positions and information structure is quite obvious on the phonetic duration of the vowel. This paper presents an acoustic study of the effect of Contrastive focus on the duration of the high vowels /i/ and /u/ in Bangla. Six monosyllabic and six disyllabic words with the target vowels, altogether 24 words have been recorded in both Contrastive focus and Neutral focus contexts. 24 sentences as response to yes-no questions and for Neutral focus, 24 simple declarative sentences were recorded from six speakers of Standard Kolkata Bangla between age group of 20-25. In Bangla, Focus is realized by low rising pitch pattern where the f_0 movement have low f_0 valley followed by a rise (Hayes and Lahiri 1991). The present study throws light on the other acoustic cue. The statistical analysis of the results from the study show significant difference in vowel duration of the focussed segment.

1pSCb12. Quantifying the manipulation of F0 as an acoustic correlate of stress in Rebkong Amdo Tibetan. Nancy J. Caplow (English, Oklahoma State Univ., 205 Morrill Hall, Oklahoma State University, Stillwater, OK 74078, nancy.caplow@okstate.edu)

Quantifying the manipulation of F0 as an acoustic correlate of stress in Rebkong Amdo Tibetan Rebkong Amdo is a non-tonal variety of Tibetan spoken in Rebkong County, Qinghai Province, PRC. Speakers draw on varied acoustic resources to convey stress in disyllabic words, which are highly frequent in the language. Non-verbs (nouns, adjectives, and numerals) are stressed on the second syllable (S2); the primary acoustic correlate of stress is F0. Verbs are stressed on the first syllable (S1); stress is conveyed by both intensity and F0. For non-verbs, speakers manipulate F0 in different ways to convey S2 prominence. The pitch contour may (a) be flat in both syllables but higher in S2; (b) be flat or slope gently in S1 but fall sharply in S2; or (c) be generally flat in S1 but define a prominent arcing curve in S2. These three patterns, which occur in free variation, require different analytical approaches in order to quantify relative acoustic prominence across syllables: (a) mean F0 (Hz) is measured for each syllable; (b) pitch slope (in Hz/100 ms) is measured for each syllable; (c) F0 curves are characterized using a principal components analysis, and by determining the tonal center of gravity. In each approach, measurements are paired to determine the magnitude of the F0 contrast across syllables in individual words; these are then aggregated, demonstrating that speakers' use of F0 to convey S2 prominence is statistically as well as perceptually significant.

1pSCb13. Glottal phonation in Punjabi: Tone or aspiration. Paroma Sanyal and Manujata Midha (Dept. of Humanities and Social Sci., Indian Inst. of Technol. Delhi, New Delhi, Delhi 110016, India, psanyal@iitd.ac.in)

This study investigates a puzzle in the glottal phonation of the South Asian Language Punjabi. Though phonetically absent from the spoken language, voiced aspirated plosives (VAP) are still perceived as a distinct phoneme by native speakers. This phonological identification is achieved by the introduction of rising and falling tones in complementary phonological environments. Specifically, the VAP lose both voicing and aspiration in word-initial positions and yet speakers are able to discern the expected phoneme due to the epenthesis of a falling tone. Elsewhere, the VAP is phonetically indistinct from the corresponding voiced plosives, but for the rising tone on the same syllable. The vowel bearing the inserted tone in Punjabi must also be stressed. Consequently, the underlying VAP and the corresponding tones can be heterosyllabic. However in such scenarios, the tones associated with the position of VAP is reversed. We propose that an attempted glottal friction is linked to the introduction of the rising and falling pitch contour. In turn, this tone is also correlated with stress, which phonetically corresponds to low pitch in Punjabi.

1pSCb14. Predictions of Laryngeal Realism for breathy-voiced stops: Counter-evidence from Bangla. Jahurul Islam (Linguist, Georgetown Univ., Poulton Hall, 1421 37th St. NW, Washington, DC 20057, mi302@georgetown.edu)

The Laryngeal Realism (LR) framework (Honeybone 2005, Beckman *et al.* 2013) argues that the breathy-voiced stops in the four-way stop systems are *specified* for multiple features: [voice] and [spread glottis] ([sg]). As a diagnostic of the *specified* features, Beckman *et al.* (2013) proposed that the acoustic cues for a specified feature be longer in slower speech. While predictions of the LR framework have been tested on two- and three-way stop systems using this technique, there has not been much on the four-way stop systems. This study presents evidence from Bangla and argues that the predictions of the LR framework apply variably for the breathy-voiced stops; [sg] is found to be the stronger candidate as the specified feature than [voice]. In slower speech, the cue for the breathy voicing (the lag-time) lengthened in the breathy-voiced stops. Contrarily, the acoustic cue for the [voice] feature (the prevoicing) was not affected by speech rate. Mixed-effects models confirmed that the effect of speech rate on the duration of breathy-voicing cues was statistically significant, but the effect on prevoicing was not. These results imply that the prediction of the LR framework may not be the best representation of the phonological features of breathy-voiced stops.

1pSCb15. The length of copied consonants in geminates, total assimilation, and ambisyllables. Mary Burke and Melissa A. Robinson (Linguist, Univ. of North Texas, 3940 N. Elm St, Ste. B201, Denton, TX 76207, burkemk412@gmail.com)

Lamkang is an under-documented endangered Kuki-Chin language of the Northeast India spoken in 40 villages spread across Manipur's Chandel district with under 10,000 Lamkang-Naga speakers (Ethnologue 2014). In Lamkang, doubled consonants have three different sources: (1) gemination when stems are followed by enclitics of the shape V; (2) total assimilation of enclitics to stem codas; (3) ambisyllables in prefixes of the shape CV (where V is an epenthetic vowel). We examine the acoustic properties of the doubled consonants in these three morphological environments to show that the length of the closures reflect the morphological derivational history. The paper investigates a method for measuring doubled consonant length: We measure from peak intensity of vowels on either side of the consonant closure, rather than the length of the closure. Orthographic practice reflects these differences as doubled consonants in the prefixes are shorter than the phonologically derived consonants with enclitics. Copied consonants are written for enclitics but not in prefixes.

MONDAY AFTERNOON, 7 MAY 2018 GREENWAY H/I, 1:00 P.M. TO 5:15 P.M.

Session 1pSP

Signal Processing in Acoustics, Underwater Acoustics, and Architectural Acoustics: Signal Processing for Complex Environments II

Sandra L. Collier, Cochair
U.S. Army Research Laboratory, 2800 Powder Mill Rd, RDRL-CIE-S, Adelphi, MD 20783-1197

Kainam T. Wong, Cochair
Dept. of Electronic & Information Engineering, Hong Kong Polytechnic University, 11 Yuk Choi Road, Hung Hom KLN, Hong Kong

Max Denis, Cochair
U.S. Army Research Lab., 2800 Powder Mill Road, Adelphi, MD 20783-1197

Invited Papers

1:00

1pSP1. Performance of WUW and general ASR speech recognition systems in different acoustic environments. Gamal Bohouta and Veton Këpuska (Elec. & Comput. Eng. Dept., Florida Inst. of Technol., 220 E University Blvd. Apt. 701, Melbourne, FL 32901, gidris2015@my.fit.edu)

The Automatic Speech Recognition (ARS) Systems have become popular in the different areas of applications, such as education assistance, medical assistance, personal assistant, robotics and vehicles, telecommunications Industry, disabilities systems, home automation, and security access control. All these applications in life will increase the number of speech commands, which are Wake-up-words. The WUW speech recognition is similar to Key-Word spotting. However, in this paper, we propose an approach that will be used to detect the Wake-up-word (WUW) as a command with 100% accuracy and to provide a whole speech recognition system that can work in both WUW and General ASR systems with high accuracy. Also, this new ASR system will be used to solve one of the biggest problems in speech recognition, which is how to discriminate between the uses of a word or phrase in an alerting and a referential context. For example, in the use of the word "Computer'' in an alerting context is "Computer, show me the chart? '' and in a referential context it is "Every computer should have a speaker. Moreover, we tested the new system with different acoustic environments, such as different background noise levels, different speaker distances to the microphone, and different speakers.

1:20

1pSP2. Automatic target recognition with uncertain scattered signal distributions. D. K. Wilson, Carl R. Hart (Cold Regions Res. and Eng. Lab., U.S. Army Engineer Res. and Development Ctr., 72 Lyme Rd, Hanover, NH 03755-1290, D.Keith.Wilson@usace.army.mil), Chris L. Pettit (U.S. Naval Acad., Annapolis, MD), Daniel J. Breton (Cold Regions Res. and Eng. Lab., U.S. Army Engineer Res. and Development Ctr., Hanover, NH), Edward T. Nykaza (Construction Eng. Res. Lab., U.S. Army Engineer Res. and Development Ctr., Champaign, IL), and Vladimir E. Ostashev (Cold Regions Res. and Eng. Lab., U.S. Army Engineer Res. and Development Ctr., Hanover, NH)

One of the primary challenges for performing robust automated target recognition (ATR) is how to compensate the signatures for environmental propagation effects. ATR algorithms tend to function well only in the specific terrain and atmospheric conditions for which they were trained. This is particularly true for acoustic signals, which undergo strong frequency-dependent scattering and refraction in both outdoor and underwater environments. To address this problem, we formulate a Bayesian sequential updating method, which accounts for realistic signal and noise distributions, and uncertainties in the parameters of these distributions. The formulation utilizes physics-based scattering models for signal fading and the cross coherence between frequencies and transmission paths. We discuss how, in a Bayesian context, the scattering models correspond to likelihood functions, which are conveniently paired with their conjugate priors to efficiently update the uncertain signal parameters (hyperparameters). The original (prior) distributions, as predicted using an initial forecast based on available weather and terrain data, are nudged towards the observed signal properties as samples are collected in real time. This leads to an optimal posterior distribution, as appropriate for use in a Bayesian classifier. The technique is demonstrated using simulations in which signal transmissions are collected along single and multiple transmission paths.

1:40

1pSP3. Active acoustic intensity errors due to contaminating noise. Mylan R. Cook, Kent L. Gee, Tracianne B. Neilsen, and Scott D. Sommerfeldt (Phys. and Astronomy, Brigham Young Univ., N201 ESC, Provo, UT 84602, mylan.cook@gmail.com)

Bias errors for both one and two-dimensional intensity probes have been previously calculated for both the traditional cross-spectral and the Phase and Amplitude Gradient Estimator (PAGE) methods [E. B. Whiting *et al*., J. Acoust. Soc. Am.**142**, 2208–2218 (2017); E. B. Whiting, M. S. Thesis (2016)]. Here, these calculations are expanded to include errors due to the source angle relative to the probe and due to contaminating noise. The noise can either be uncorrelated at each microphone or self-correlated, which is modeled as a plane wave with a varying angle of incidence. The errors in both intensity magnitude and direction are dependent on the signal-to-noise ratio (SNR), frequency, source properties, incidence angles, probe configuration, and processing method. The PAGE method is generally found to give more accurate results, though uncorrelated noise with a low SNR can yield large errors. For broadband signals, the PAGE method can be used beyond the spatial Nyquist frequency. [Work supported by NSF.]

2:00

1pSP4. Noise adaptive real-time estimation of infrasound using a compact array. William G. Frazier (Hyperion Technol. Group, Inc., 3248 West Jackson St., Tupelo, MS 38801, gfrazier@hyperiontg.com)

Previous investigations [Frazier, ASA Pittsburg (2015); ASA San Francisco (2013); ASA Kansas City (2012)] have demonstrated that exploiting the correlated statistical properties of wind noise on short spatiotemporal scales with a compact array permits enhanced detection and reduced estimation error of infrasound transients resulting from explosions. The approach was shown to be effective in real-time using non-parametric wind noise models. More recently [Frazier, ASA Boston (2017); ASA Salt Lake City (2016)], the basic ideas of the approach were extended to estimation of quasi-stationary infrasound signals such as microbaroms using parametric wind noise models and was demonstrated to be effective, but the computational requirements were too demanding for real-time implementation. This presentation will highlight modifications that enable real-time implementation for quasi-stationary signal estimation.

2:20

1pSP5. A maximum likelihood estimator for spectral models of acoustic noise processes. Gregory W. Lyons (Construction Eng. Res. Lab., U.S. Army Engineer Res. and Development Ctr., Engineer Res. and Development Ctr., 2902 Newmark Dr., Champaign, IL 61822, gregory.w.lyons@erdc.dren.mil) and Carl R. Hart (Cold Regions Res. and Eng. Lab., U.S. Army Engineer Res. and Development Ctr., Hanover, NH)

The power spectral density of time series from many acoustic phenomena can vary rapidly through several decades of magnitude, particularly for noise processes. This property can complicate evaluation of spectral density models with respect to a non-parametric estimate of the spectral density, also known as the periodogram. Typical measures, such as the sum of squares of the residuals, can suffer from bias errors and oversensitivity to large values. A log-likelihood function is here presented for a periodogram derived from a sequence of independent, circularly-symmetric complex normal Fourier components. The bias-corrected Akaike's information criterion of an expected-value spectral model is shown to be a useful measure for multi-model selection. A maximum-likelihood estimator is formed for spectral model parameters with respect to a known periodogram, along with approximate confidence intervals for the parameter estimates. Results from the maximum-likelihood estimator are compared with weighted and unweighted least-squares methods by estimating model parameters for periodograms from both synthetic and measured noise time series. Practical considerations for general periodogram fitting and numerical methods are discussed.

2:40

1pSP6. Explorations of *in-situ* passive source localization in the Philippine Sea using frequency-difference matched field processing. David J. Geroski (Appl. Phys., Univ. of Michigan – Ann Arbor, Randall Lab., 450 Church St., Ann Arbor, MI 48109, geroskdj@umich.edu), Matthew Dzieciuch (SIO/UCSD, La Jolla, CA), and David R. Dowling (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

Matched Field Processing (MFP) is a well-known technique for source localization in challenging acoustic environments. Accurate source localization using this method requires a correlation between array-recorded and replica acoustic fields calculated using knowledge of the acoustic environment. For a well-understood acoustic environment, this method is successful. However, recorded signals may be sensitive to details of the environment which are unknown and not included in the calculation of the replica field. The severity of this mismatch increases with frequency, as phase coherence is lost more easily at higher frequencies. In this case, a mismatch between the recorded and calculated fields will occur which can cause MFP to fail at relevant frequencies and ranges in the deep ocean. A proposed remedy to this problem may be analyzing the frequency-difference autoproduct of the measured acoustic field instead of analyzing the field itself. This presentation explores using this method to localize moored sources using data from the North Pacific Acoustic Library. Localization statistics are presented based on sources which are moored hundreds of kilometers from the receiving array, broadcasting between 200 and 350 Hz, and are matched using out-of-band frequencies between 10 and 50 Hz. [Sponsored by ONR.]

3:00–3:15 Break

3:15

1pSP7. Signal and information processing in complex ocean environments. David P. Knobles (KSA, LLC, PO Box 27200, Austin, TX 78755, dpknobles@yahoo.com), Preston S. Wilson (Mech. Eng. Dept. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Mohsen Badiey (Elec. Eng., Univ. of Delaware, Newark, DE)

Even a cursory examination of the information theory relationship between model capacity/complexity, model error, and prediction error should suffice to convince one that to treat the ocean as deterministic by increasing the dimensionality of the number of physical mechanisms, parameters, etc., may prove ineffective in achieving desired results in complex waveguides. Instead of a deterministic approach that results from solving a high-dimensional inverse problem which can lead to poor predictive capability, the approach advocated here predicts the statistics of the Detection, Classification, and Localization (DCL) problem. Various dimensional reduction methods are utilized such as those that can be applied to images in the spectral acoustical domain. Bayesian Information Criteria (BIC) and trans-dimensional methods coupled with machine learning algorithms can then be used to construct a meaningful signal processing approach which optimizes predictive capability. Acoustic and environmental survey data collected in the New England Seabed Characterization experimental area that has a high horizontal variability are used to demonstrate the approach. [Work supported by ONR.]

3:35

1pSP8. Temporal-spatial and source level distributions for a repertoire of fin whale vocalizations in the Norwegian Sea observed with a coherent hydrophone array. Heriberto A. Garcia (Elec. and Comp Eng., Northeastern Univ., 151 Mystic St., Apt 35, Arlington, MA 02474, garcia.he@husky.neu.edu), Delin Wang, Chenyang Zhu (Elec. and Comp Eng., Northeastern Univ., Boston, MA), WEI HUANG (Elec. and Comp Eng., Northeastern Univ., Malden, MA), Anna Kaplan, and Purnima Ratilal (Elec. and Comp Eng., Northeastern Univ., Boston, MA)

The vocalizations of the fin whale are detected, characterized and localized over instantaneous wide areas of the Norwegian and Barents Seas using a large-aperture densely-sampled coherent hydrohone array via the passive ocean acoustic waveguide remote sensing (POAWRS) technique from observations in late winter to early spring 2014. The fin whale vocalizations are comprised of their characteristic 20 Hz pulses, interspersed by 130 Hz upsweeps, less abundant 30-100 Hz downsweep chirps and 18-19 Hz centered backbeats. The time-frequency characteristics of these vocalization types and their diel occurrence rate time-series are quantified in three distinct regions of the Norwegian Sea, off the coasts of Alesund, Lofoten, and the Northern Finnmark. Their vocalization rates are a factor of roughly 5 times and 17 times larger respectively for the 20 Hz and 130 Hz centered vocalizations off Northern Finnmark than the other regions. The vocalization rate spatial density distributions, source level distributions and probability of detection (PoD) regions are estimated for the 20 Hz and 130 Hz vocalizations. Highly intense 20 Hz pulses from closeby fin whales are received with their first-order harmonics centered at 40 Hz with twice the bandwidth but 30 dB lower source levels, indicating a nonlinear mechanism for their production.

3:55

1pSP9. Design of a signal normalizer for high-clutter active-sonar detection. Frank W. Bentrem, Jonathan Botts (Appl. Res. in Acoust., 209 N. Commerce St, Ste 300, Culpeper, VA 22701-2780, frank.w.bentrem@ariacoustics.com), and Jason E. Summers (Appl. Res. in Acoust., Washington, District of Columbia)

We present normalizer design considerations for detection of targets in a high-clutter environment. Several classes of constant-false-alarm-rate (CFAR) normalizers have conventionally been used to bring out a target signal from the background noise and reverberation in active sonar signal processing. The special challenges of detection in high-clutter environments are outlined, and the normalizer classes are compared for their suitability of use in these environments. We consider statistical and multiple-look normalizers and give emphasis to the class of split-window mean-power normalizers and implementations that preserve target signal-to-noise ratio (SNR) and highlight structure while mitigating artifacts that are sometimes produced near clutter edges. Statistical comparisons and specific examples are presented, and we propose an efficient CFAR normalizer for high performance in regions of high clutter.

4:15

1pSP10. Dimension-reduced adaptive pulse compression with model error compensation. Jonathan Botts (Appl. Res. in Acoust., 209 N. Commerce St, Ste 300, Culpeper, VA 22701, jonathan.botts@ariacoustics.com), Jason E. Summers (Appl. Res. in Acoust., Washington, District of Columbia), Frank W. Bentrem (Appl. Res. in Acoust., Culpeper, VA), and Charles F. Gaumond (Appl. Res. in Acoust., Washington, District of Columbia)

In active acoustic imaging, range sidelobes from competing clutter and interferers in acoustically complex environments may mask signals of interest. In order to unmask these signals, increase range resolution, and decorrelate the time-series we introduce dimension-reduced adaptive pulse compression (APC) with model-error compensation that operates after a conventional matched filter. Structured covariance estimates minimize the required sample support while dimension reduction occurs naturally and a priori after the matched filter. We compensate for predictable signal distortions such as Doppler through both covariance matrix tapers and truncation of the covariance matrix. Model-based APC is computationally intensive, scaling with the third or fourth power with replica length, so algorithmic modifications are introduced to reduce computation by orders of magnitude. We evaluate the algorithm's robustness and computational feasibility on simulated and archived data. [Portions of this material are based upon work supported by the Naval Sea Systems Command.]

4:35

1pSP11. Signal compression technique in decomposition of arrivals having different speed of sound propagation through human lungs in the frequency range of 10–19 kHz. Vladimir Korenbaum, Anton Shiryaev, Anatoly Kostiv, and Maria Safronova (Pacific Oceanologic Inst., 43, Baltiiskaya Str., Vladivostok 690041, Russian Federation, v-kor@poi.dvo.ru)

The unexpected phenomenon of sound transmission through human lungs at frequencies above 10 kHz with a speed of about 1000 m/s was revealed by Rueter *et al*., 2010. The objective is a study of characteristics of sound transmission in human lungs in the frequency range of 10–19 kHz using signal compression technique. The 14-channel receiving apparatus was used. Chirp signals 10–19 kHz (6 min) were emitted into human thorax by small shaker. Sound propagation in human lungs was studied in paths with opposite chest positions of shaker and sensors in 4 volunteers. An existence of low-speed arrivals with propagation velocities of 150–50 m/s, which amplitude and/or velocity is inversely dependent on the air-filling of lungs (inspiration/exhalation) has been revealed. These arrivals may be treated as a result of sound propagation mainly through the lung parenchyma. On the contrary, the amplitudes of high-speed arrivals with velocities of 150–1000 m/s are enhanced with a decrease in air-filling of lungs. They may be connected to the sound propagation mainly through high-density tissues of thorax. The results are promising for medical acoustic visualization of local reduction in air-filling/ventilation of lung parenchyma. [This study was supported by the RFBR grant 16-08-00075.]

4:55

1pSP12. Processing of emitted respiratory noises to monitor health condition and spatial displacement of a scuba diver under water. Vladimir Korenbaum (Pacific Oceanologic Inst., 43, Baltiiskaya Str., Vladivostok 690041, Russian Federation, v-kor@poi.dvo.ru), Sergey Gorovoy (Far Eastern Federal Univ., Vladivostok, Russian Federation), Anatoly Kostiv, Anton Shiryaev, Aleksey Borodin, Veniamin Dorozhko (Pacific Oceanologic Inst., Vladivostok, Russian Federation), and Andrey Fershalov (Far Eastern Federal Univ., Vladivostok, Russian Federation)

Monitoring health condition of a scuba diver and his displacement within the exploited water area is necessary to ensure the safety of diving. Respiratory noises, emitted by the diver into water, may be used for this purpose. In the hydroacoustic basin, powerful signals, having repetition frequency of about 0.12 Hz are found, which correspond to diver's respiratory rate. The main broadband signal of exhalation is concentrated in the frequency band of 150–1150 Hz. The high-frequency signal in the frequency band of 3.5–4.7 kHz is associated with inspiration. Thus, acoustic estimating the respiratory rate and the ratio of the inspiration/expiration durations are possible, which are important physiological parameters to assess a health condition of the diver. When registering in sea respiratory noises of scuba diver against the background noise, it is possible to trace the acoustic signs of respiratory maneuvers associated with the noise of floating bubbles in the spectrograms at distances up to 100–200 m. The same acoustic signs provide monitoring the displacement of a scuba diver by determining the delays of the maxima of the cross-correlation function at 2 hydrophones. Trails of the delays in correlograms are traced at distances up to 300 m.

Session 1pUW

Underwater Acoustics: Instrumentation for Underwater Acoustics

Aaron Darnton, Chair
Naval Undersea Warfare Center Division Keyport, 610 Dowel St, Keyport, WA 98345

Contributed Papers

1:45

1pUW1. Design considerations for a compact correlation velocity log. Thomas Blanford (Graduate Program in Acoust., The Penn State Univ., 1623 S. Ashwicken Ct, State College, PA 16801, teb217@psu.edu), Daniel Brown, and Richard Meyer (The Appl. Res. Lab., The Penn State Univ., State College, PA)

A Correlation Velocity Log (CVL) has some advantages (lower source level and operating frequency) over a Doppler Velocity Log (DVL) as a navigational aid for an unmanned underwater vehicle (UUV). A CVL provides a bottom referenced velocity estimate by estimating displacement using the incoherently scattered field from an acoustic projector and an array of hydrophones. A small low cost UUV generally operates in shallow water and has limited space and power available for a navigational aid, creating added constraints for the design of a CVL. The important design considerations (such as size, array geometry, operating frequency, and bandwidth) will be discussed as they relate to accuracy of the velocity estimate and operating range. [The authors acknowledge the financial support for this work by Lockheed Martin Undersea Systems.]

2:00

1pUW2. Underwater fiber optic transducer. Aaron Darnton (Naval Undersea Warfare Ctr. Div. Keyport, 610 Dowel St, Keyport, WA 98345, aaron.darnton@navy.mil)

Fiber optic sensors have been used in various configurations including interferometric, Bragg grating, and Rayleigh scattering approaches. In this work, the Rayleigh approach is considered for use as a hydrophone. This approach transmits light along a single fiber and utilizes the backscattered light to infer small strains in the fiber. Typically in-water use of this type of optical acoustic receiver has been limited due to the gross motion of the fiber in the water. The gross motion tends to wash out the acoustically driven strains of interest. Additionally, sensitivity can be relatively low in this approach. This work considers sensing on a section of fiber wrapped around a mandrel for the purpose of increasing sensitivity as well as reducing the effects of gross motion.

2:15

1pUW3. Experiments with the Underwater Bubble Low Frequency Sound Resonator. Andrey K. Morozov and Douglas C. Webb (Teledyne Marine Systems, 49 Edgerton Dr., North Falmouth, MA 02556, amorozov@teledyne.com)

An infra-sound source with a resonator in the form of an underwater bubble made from an elastic material is a simple engineering solution. The elastic membrane supports high volume displacement and a large radiation aperture. The Rayleigh-Plesset equation defines the dynamics of a spherical bubble including surface tension and viscous effects. The shape of a large bubble deforms due to gravitational effects. Its internal pressure oscillations are comparable with the difference of gravity pressure and these effects are part of its dynamics. The real Q-factor of a practical bubble is smaller than theoretical. The underwater bubble resonator was studied experimentally in the pool and at the Woods Hole Oceanographic Institution's dock. The research includes the 3D finite-element analysis and its comparison with the experimental data. The study gave the real values for the Q-factor, resonance frequencies and sound pressure levels. The experimental resonator has good performance with a maximum SPL 190 dB at the resonance frequency 6-12 Hz. It was shown that a cylindrical resonator can be towed with a speed of 8 knots. The research proved that this is a practical approach for a coherent low frequency sound source, which can find applications in a marine seismic survey.

2:30

1pUW4. Videos of ultrasonic wave propagation through transparent acrylic objects in water for introductory physics courses produced using refracto-vibrometry. Matthew R. Mehrkens, Benjamin A. Rorem, and Thomas M. Huber (Phys., Gustavus Adolphus College, 800 W College Ave, Saint Peter, MN 56082, huber@gac.edu)

This project used refracto-vibrometry to produce full-field videos of ultrasonic wave propagation in water; these videos could be incorporated into high school and college physics courses to illustrate wave behaviors. Refracto-vibrometry (RV) is an interferometric method for optically measuring sound waves. The measurement beam from a Polytec PSV-400 scanning laser Doppler vibrometer was directed through a water tank towards a stationary retroreflective surface. Acoustic wave fronts (density variations) which pass through the measurement laser cause variations in the integrated optical path length. By superimposing tens of thousands of scan points, videos enable visualization of the ultrasound propagation. In one experiment, ultrasonic waves in water (1484 m/s) are incident on a submerged acrylic block at a non-zero angle. The waves are refracted as they travel at about 2700 m/s through the block. Students can determine this speed of sound by measuring the angle of refraction and lateral deflection of the wave fronts in the video. In another experiment, ultrasonic waves passed through an acrylic cylinder, producing a Mach cone in water. Measurement of the Mach cone angle enables determination of the acrylic speed of sound. Links for videos and sample worksheets will be provided.

Payment of an additional fee is required to attend this session.

MONDAY AFTERNOON, 7 MAY 2018 NICOLLET D2, 7:00 P.M. TO 9:00 P.M.

Session 1eID

Interdisciplinary: Tutorial Lecture on Hearing Loss and the Future of Auditory Implants

Christina J. Naify, Chair
Jet Propulsion Lab, 4800 Oak Grove Dr, MS 157-316, Pasadena, CA 91101

Chair's Introduction—7:00

Invited Paper

7:05

1eID1. Hearing loss and the future of auditory implants. Andrew J. Oxenham (Univ. of Minnesota, 75 E River Parkway, Minneapolis, MN 55455, oxenham@umn.edu)

Hearing loss is a major and growing health concern worldwide. According to the National Institute on Deafness and Communication Disorders (NIDCD), 17% of the adult population in the US (around 36 million people) report some form of hearing loss, with the proportion of affected individuals rising to nearly 50% among those aged 75 or older. Loss of hearing has been associated with increased social isolation, more rapid cognitive decline, and other more general health issues, although causal relationships have yet to be established. This tutorial will review the physiology of hearing loss, along with its perceptual consequences, as measured in the laboratory and experienced in everyday life. The focus of the tutorial will be on implantable technologies that have been used to alleviate severe hearing loss and deafness, with particular emphasis on cochlear implants. Although cochlear implants have enjoyed remarkable success over the past few decades, they do not restore normal hearing, and may be approaching their technological limits in terms of the benefits that patients can gain from them. The tutorial will end by exploring future directions of implantable and other technologies in the quest to restore and maintain hearing throughout the lifespan.

MONDAY EVENING, 7 MAY 2018 SKYWAY B, 5:00 P.M. TO 6:00 P.M.

Meeting of Accredited Standards Committee (ASC) S2 Mechanical Vibration and Shock

J. T. Nelson, Chair ASC S2
Wilson Ihrig & Associates, Inc., 6001 Shellmound St., Suite 400, Emeryville, CA 94608

R. J. Peppin, Vice Chair ASC S2
5012 Macon Road, Rockville, MD 20852

Accredited Standards Committee S2 on mechanical vibration and shock. Working group chairs will report on the status of various shock and vibration standards currently under development. Consideration will be given to new standards that might be needed over the next few years. Open discussion of committee reports is encouraged.

People interested in attending the meeting of the TAG for ISO/TC 108, Mechanical vibration, shock and condition monitoring, and four of its subcommittees, take note that meeting will be held in conjunction with the Standards Plenary meeting at 9:15 a.m. on Tuesday, 8 May 2018.

Scope of S2: Standards, specification, methods of measurement and test, and terminology in the field of mechanical vibration and shock, and condition monitoring and diagnostics of machines, including the effects of exposure to mechanical vibration and shock on humans, including those aspects which pertain to biological safety, tolerance, and comfort.